Whitepaper

Voice Over LTE VoLTE

TABLE OF CONTENTS

Motivations for Enabling Voice over LTE

The motivation for the deployment of 3GPP Long-term Evolution (LTE) mobile broadband technology is simple: All things considered, LTE delivers to carriers the lowest cost-per-transported bit. That said, the adage that "voice pays the bills" still applies: though in decline, carriers continue to derive the bulk of their revenues from voice and integrated messaging services.

In the context of LTE, this presents a dilemma. A fundamental aspect of legacy technologies such as GSM, UMTS, and cdma2000 is that they possess *integrated* services[1]: voice, voice supplementary services (e.g., call forwarding), short messaging, etc. In contrast, LTE makes no such provisions: it is subscriber service-agnostic. Further, LTE is a pure packet technology, with no inherent conception of a circuit-switched (CS) bearer, on which legacy voice services depend.

Because of the realities of the cellular revenue model – and, indeed, because cellular subscribers expect service continuity – the question arises: How can we best deliver voice and other legacy services via LTE? As with any engineering exercise, this requires articulation of the requirements.

Requirements for Voice over LTE

The requirements for voice[2] over LTE (VoLTE) solutions fall into one – or both – of two categories: the requirements of cellular subscribers, and the requirements of cellular carriers. The most important of these requirements are as follows.

Subscriber Requirements

Telephony – Subscribers, first and foremost, expect replication of legacy cellular telephony services. The expectation of support for voice and messaging services is intuitive. But supplementary services – and the management thereof – are also required. While this might seem like a trivial requirement, the implementation of supplementary services in the context of VoLTE isn't necessarily obvious. Further, subscribers – not to mention regulators – have clear expectations on the availability and performance of emergency calling services. Additionally, video telephony is fast becoming a basic expectation for subscribers.

Quality – Subscribers also require call quality that's not noticeably different from that of existing telephony services. This presents a technical challenge in that LTE is a purely packet-switched (PS) system, and the maintenance of quality-of-service (QoS) in PS systems is notoriously challenging – especially over the parts of the channel outside the cellular carrier's purview.

Ubiquity – Subscribers's expectations in terms of service ubiquity go only one way over time: up. This means that both local and wide-area mobility must be supported – transparently – by any VoLTE solution. In technical terms, this means seamless inter-RAT (IRAT) operations as well as domestic and international roaming.

Battery – In order to be viable, a VoLTE solution cannot result in significantly higher battery consumption than is experienced with legacy voice services.

Carrier Requirements

Cost – For cellular carriers, a minimized total cost of ownership is the fundamental requirement. But costs can be manifested in a multitude of ways. For example:

Efficiency – The more efficient a technology is, the more traffic can be handled per node and per megahertz of transmission capacity. This is particularly important over the air interface, since radio spectrum is one of a carrier's most precious assets: Spectrum procurement – unlike equipment procurement – is costly, regulated, and time-consuming.

Complexity – Reduced network complexity was a fundamental principle employed during the design of LTE. As complexity increases, the required hardware and the software development effort – particularly with terminal devices –

[1] Indeed, GSM and UMTS terminals are, strictly speaking, Integrated Services Digital Network (ISDN) terminals.

[2] For the remainder of this document, "voice over LTE" and "VoLTE" will be used generically to denote voice, voice supplementary services, short messaging services, and all other legacy subscriber services.

increases. So an excessively complex voice solution is not desirable.

Reusability – Solutions which permit the reuse of existing infrastructure – or are designed to have a long lifespan – are desirable.

It is with these requirements in mind that we evaluate the various options to deliver VoLTE.

LTE Voice Options

There is a diversity of options when it comes to solving the LTE voice problem. These range from multi-radio solutions, to solutions which tunnel legacy signaling through LTE, to “pure” SIP-based voice-over-IP (VoIP) solutions, to not using LTE for voice at all, instead relying on legacy networks to provide voice facilities. These are, of course, in addition to VoIP solutions which originated in the fixed broadband world – like Skype – which now have mobile incarnations. (These are, in some contexts, referred to as “over-the-top” solutions.) The major solutions to deliver VoLTE are as follows.

VoIP Adopted from Fixed Broadband

Voice-over-IP solutions like Skype, Google Talk, and Windows Live Messenger are used increasingly on mobile devices, and they are attractive in a number of ways, not least of which being that they exist today and are well-proven. These solutions have large user bases and are relatively cheap, which could not unreasonably be said to be two sides of the same coin.

However, having not evolved in the cellular world, fixed broadband voice solutions have shortcomings which show through when used in a cellular context. First, there is little bearer integration, which means that quality of service (QoS) cannot be guaranteed, especially during mobility to the oldest network technologies. While this can be improved to a degree in the fixed broadband world by “throwing bandwidth” at the problem, it is a non-trivial challenge in the mobile world where bandwidth is precious. Further, the number of fixed broadband VoIP solutions is large, with many utilizing proprietary technology and thus, mostly unable to interoperate. Most importantly, these services are delivered via third parties and are thus out of the carrier’s control.

Simultaneous Voice and LTE

Some of the earliest LTE deployments addressed the LTE voice issue by utilizing a dual-radio solution in which the mobile device is simultaneously connected to both the LTE and a legacy network, the latter of which being used to supply voice and messaging services. The advantages of such “simultaneous voice and LTE” (SVLTE) solutions are that, as the name implies, voice and LTE data services are supported simultaneously. Moreover, no network-to-network (NNI) interface is required; the two networks operate completely independently of each other and require no upgrades to support SVLTE.

Unfortunately, SVLTE solutions add significant complexity in terms of mobile device hardware, since two completely independent baseband and RF chains are required, with obvious cost implications. Further, SVLTE solutions have proven to be power-hungry in practice, with an unsatisfying user experience in terms of battery life. And since the LTE and legacy cells don’t necessarily have identical coverage footprints, the user experience during mobility can be less than desired.

Circuit-switched Fallback

Circuit-switched fallback (CSFB) is another solution to the LTE voice problem. With CSFB, a mobile device which is camped on LTE temporarily switches to a legacy radio access technology (RAT) in order to originate or receive a voice call. After the voice call completes, the mobile device can return to LTE. CSFB is an attractive LTE voice solution since it leverages existing infrastructure and accounts for differing states of network evolution amongst carriers – a key consideration in the context of roaming.

CSFB-capable mobile devices need not switch to a legacy RAT in order to send or receive SMS, since the processing of those messages can be handled by legacy systems. Short messages can be tunneled to or from the LTE network via the SGs network-to-network interface (NNI). Another option is SMS-over-IMS, which also permits the exchange of SMS without leaving LTE which has the added advantage of not requiring an NNI.

However, CSFB necessarily does not permit simultaneous voice and LTE data access. It also increases call setup delays as the mobile device must perform cell reselection and synchronization before proceeding with call setup. Further, field experience has shown that after a CSFB call completes, mobile devices can “stick” to the legacy network (e.g., due to packet data transfer), temporarily preventing the user from experiencing the full benefits of LTE.

Voice over LTE via Generic Access
Voice over LTE via generic access (VoLGA) is a true voice-over-LTE solution which enables LTE mobile devices to access legacy systems and services without having to leave the LTE domain. VoLGA leverages an emulation principle, with the LTE network appearing to be a legacy BSC/RNC from the MSC's perspective, and with the legacy MSC appearing to be an application function from the LTE mobile device's perspective. Legacy call control messaging is tunneled directly to and from the LTE mobile device, so the legacy call model is unchanged.

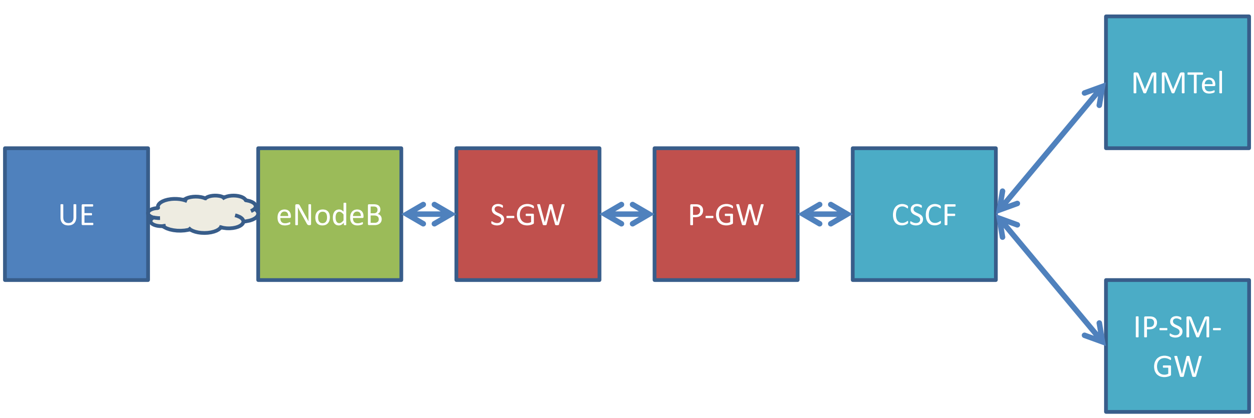
VoLGA is attractive because it relies on principles and technologies that already have been proven in the field. Unlicensed mobile access (UMA) – in which compatible devices received voice call processing services from cellular MSCs via Wi-Fi access – has been in the field for years. VoLGA also offers the added advantage of not requiring any changes to legacy systems, as it was designed specifically to hide the nature of the access network from the core network.

Even given its appeal on paper, VoLGA was not able to achieve a critical mass of industry support, with some terming it backward-looking in the sense that it would require LTE to continue to rely on legacy calling paradigms and systems rather than embracing newer calling paradigms like Session Initiation Protocol (SIP).

GSMA IR.92
Given the limitations of the VoLTE solutions articulated above, a group of cellular carriers and technology vendors organized under the banner "One Voice". While not strictly part of its mandate, the One Voice initiative focused on a "pure" VoIP solution for VoLTE. The initial work of One Voice specified a minimum set of IP Multimedia Subsystem (IMS) features to enable VoIP-based VoLTE. The GSM Association (GSMA) refined and completed this effort, which is codified in the GSMA permanent reference document (PRD) IR.92.
IR.92 is appealing because it is VoIP-based and thus forward-looking, and because it leverages the investments in IMS made by the cellular industry. But since IR.92 is a pure VoIP solution, there is substantial complexity introduced, in particular, to enable interworking with legacy systems.

A summary of the key aspects of IR.92 is as follows.

Overview of GSMA PRD IR.92
The GSMA's PRD IR.92 – *IMS Profile for Voice and SMS* – is, as its name indicates, a *profile* as opposed to a *technical specification*. That is, rather than specifying new technologies, this document defines how the frameworks and facilities already defined by 3GPP should be configured and utilized in a manner that permits, in particular, interoperability between mobile devices and serving networks, as wells as amongst networks.
The reference architecture implied by IR.92 is as follows:



Thus, in addition to the mandatory LTE nodes, IR.92 VoLTE relies on the IMS (CSCF) and the Multimedia Telephony (MMTel) server. IR.92 captures the way in which these nodes are to be configured, connected, and utilized in order to deliver voice services.

It is interesting to note that, for VoLTE calls, the VoIP "control plane" and "user plane" both are multiplexed on to the LTE user plane. Thus, VoLTE call control is performed outside the purview of the LTE network. This is a good illustration of how, unlike legacy cellular technologies, subscriber services in LTE are *non*-integrated.

Given that LTE subscriber services are non-integrated, and pursuant to the aforementioned subscriber requirements, IR.92 addresses voice call control including supplementary services, voice media flows, supplementary service management, and short messaging. And with the carrier requirements in mind, IR.92 also addresses a number of "under-the-hood" aspects of the LTE radio access network and core network. A summary of IR.92's requirements in these areas is as follows.

Call Control

SIP Registration
Intuitively, IR.92 requires the mobile device to indicate its support for MMTel-based supplementary services during SIP registration. And if the device supports SMS-over-IP, it is required to indicate so as well. Devices are further required to supply their IMEI during SIP registration, which permits carriers to implement management policies in the P-CSCF which are device or device class specific.

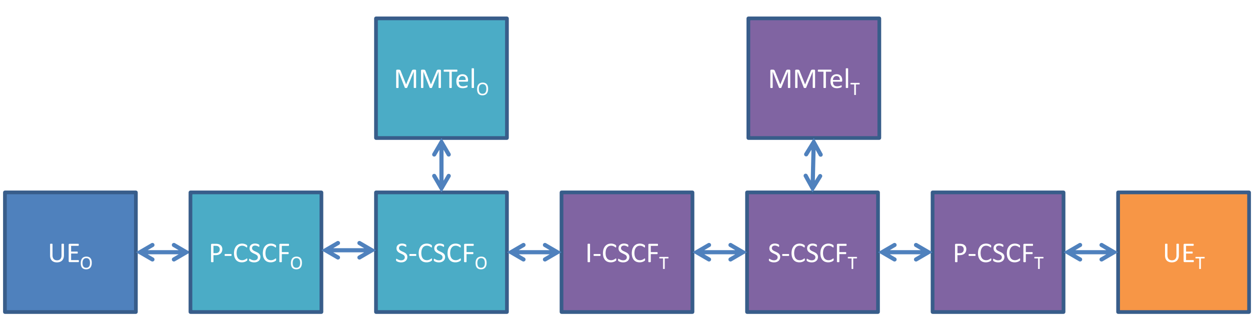
IR.92-compliant devices are required to support network-initiated de-registration, so that the network can forcibly detach a mobile device, for example, during an IMS maintenance action.

SIP Authentication
IR.92 leverages the existing mechanisms for IMS authentication, i.e., IMS Authentication and Key Agreement (IMS-AKA). VoLTE are expected to have installed a Universal Integrated Circuit Card (UICC) equipped with the IMS Services Identity Module (ISIM) application. The ISIM application contains the credentials critical to authentication such as the subscriber's IP Multimedia Private Identity (IMPI), the home carrier's domain name, and a shared secret key (which is also used in the encryption process).

If the ISIM application is absent from the UICC, IMS-AKA procedures can still be performed using the credentials contained in the Universal Services Identity Module (USIM) application. The legacy GSM SIM and CDMA CSIM applications are not supported.

SIP Addressing
IR.92 requires compatible mobile devices and networks to support specific addressing or "dialing" capabilities. Support for both SIP-native B-party addresses as well as telephony-based addresses is mandatory. The former enables calling between devices known to be IMS-capable, while the latter permits interoperability with legacy fixed and mobile networks and devices. IR.92 specifically requires support for home-local numbers, and allows for support of geo-local numbers.

SIP Call Processing
An originating mobile device, according to IR.92, must specify that it wants the call to be to be handled by an MMTel server. This is accomplished by setting the IMS Communication Service Identifier (ICSI) in the communication invitation to point to the MMTel service. This results in calls being looped – by the Serving CSCF (S-CSCF) – through the MMTel server in each subscriber's home network, as shown in the following simplified call control path:



It is the MMTel server that implements – among other things – the telephony-equivalent supplementary services such as call barring and call forwarding (see below).

With SIP, on which IMS is based, call control and bearer (transport) control are essentially independent. This is by design. But in the context of IR.92 there are two situations in which it is desirable to have some communication between the two domains. The first is during call session establishment. It is possible that a terminating device could accept a call invitation (and alert the terminating subscriber), but subsequently turn out that there are insufficient bearer resources to support the call. This results in a so-called “ghost call”. This is not normally a problem in legacy circuit-switched systems because call and bearer control are relatively more coupled. In order to minimize the possibility of this occurrence, IR.92 requires the use of the SIP preconditions framework. This ensures that the terminating subscriber is not notified of the incoming call unless it is certain that there are sufficient resources to support the call (at the required QoS).

The second situation in which call and bearer control require some communication is in the context of call drops and call establishment failures. If, for example, communication between the LTE/SAE network and the internet is lost, or if the radio link fails, IR.92 requires the IMS call session to be reestablished or cleared gracefully on both the mobile device side and on the network side.

In order to ensure at least some level of communication between parties, IR.92 requires terminating devices to accept a communication session even if they cannot support some parts of it. For example, if a certain terminating device can support the audio codec but not the video codec requested by the originator, the device is required to accept the audio portion of the call. IR.92 also requires support for call “upgrades”, such as voice-only to voice plus video.

VoLTE emergency calls must be supported natively by both the mobile device and the network according to IR.92. However, it is acknowledged that IMS rollout will take some time, and that local regulations could necessitate that emergency calls be placed over a legacy network, so redirection to a legacy system for emergency call processing is permitted.

P-CSCF Discovery

IMS calls are processed by the subscriber’s S-CSCF in the home network. The connection to the S-CSCF is, however, indirect, via a local or “proxy” CSCF (P-CSCF). Since the P-CSCF varies with time and the mobile device’s serving network, an important step in the enablement of voice calling capabilities is the determination – or “discovery” – of the P-CSCF. IR.92 requires the mobile device to discover the P-CSCF address from the core network during the IMS PDN connectivity establishment process. This ensures that the serving network can point the mobile device to the optimal P-CSCF at runtime.

Note that the use of a P-CSCF facilitates lawful intercept activities in the visited network, as call control signaling is de-tunneled at the P-CSCF.

Supplementary Services

The IR.92 standard names a specific set of legacy supplementary services that are to be replicated for VoLTE. They are (going by the legacy names):

- Calling Line Identification Presentation/Restriction (CLIP/CLIR)
- Connected Line Identification Presentation/Restriction (COLP/COLR)
- Call Forwarding – Unconditional/Busy/Not Reachable/No Reply (CFU/CFB/CFNRc/CFNRy)
- Bar All Outgoing/Incoming Calls (BAOC/BAIC)
- Bar Outgoing International Calls/Ex Home Country (BOIC/ BOICexHC))
- Bar Incoming Calls When Roaming (BICRO)
- Call Waiting (CW)
- Call Hold (HOLD)
- Message Waiting Indication (MWI)
- Multi-party (MPTY)

IR.92 also requires support for a supplementary service that has no analog in the legacy telephony world, namely, Call Forwarding – Not Logged In, which applies when the terminating subscriber has not registered with any P-CSCF. Again, all of these supplementary services are realized on the MMTel server.

In legacy systems, telephony features are managed – activated, deactivated, or queried – via dedicated signaling messages, or via star codes (e.g., *86). Since call control in VoLTE uses completely different signaling – and since star codes are tightly tied to the legacy telephony paradigm – supplementary services and features in VoLTE are managed differently. IR.92 requires the use of XML Configuration Access Protocol (XCAP). With XCAP, the mobile device – following established web paradigms – connects to the MMTel server via HTTP and utilizes the standard PUT, GET, and DELETE methods manipulate XML documents (which describe the supplementary services and their state) stored thereon. IR.92-compatible devices are required to support the XML document defined by 3GPP in TS 24.623 for supplementary service management.

Voice Media
IR.92 requires the use of the well-known adaptive multi-rate (AMR) 3GPP voice codec, and also specifies support for AMR-wideband (AMR-WB). This choice simplifies interoperability with legacy 3GPP networks, in particular, in the context of transcoder-free operation/tandem-free operation. Codecs are permitted to operate in octet-aligned or bandwidth-efficient modes, but the bandwidth-efficient mode is preferred in order to minimize network resource utilization. In order to ensure interoperability with all legacy telephony networks – in addition to the necessary voice transcoders – support for transmission of DTMF tones is mandatory.

AMR voice, being real-time in nature, is transported via the Real-Time Protocol (RTP). IR.92 specifies that, normally, one RTP packet should contain a single AMR frame, though up to 12 AMR frames per RTP packet are possible. Intuitively, receivers of RTP packets are required to perform packet reordering according to the RTP packet timestamp, duplicate detection in case the RTP sender transmitted multiple copies for redundancy, and appropriate jitter buffer management.

The Real-Time Control Protocol (RTCP) is also utilized, but its messages are normally suppressed during voice calls, only being transmitted when a call is put on hold, in order to serve as a link keep-alive.

Short Messaging
In conjunction with support for voice services, VoLTE-compatible devices also must support sending and receiving SMS over IP, and the IMS in VoLTE-compatible networks must be capable of functioning as an IP short message gateway (IP-SM-GW).

Note that SMS over IP represents a second mechanism for providing short messaging services to users, the other of which being SMS over the SGs NNI. (This is the mechanism that is used by default in the absence of IMS.) Since mobile devices might support both of these SMS over LTE solutions, carriers are required to provision the preferred SMS transport via the IMS Management Object.

Radio Access Network

Physical Layer
IR.92 addresses only a few physical layer aspects. Network and mobile device support for discontinuous reception (DRX) is mandatory as it helps to extend the mobile's battery life. The base station must be capable of semi-persistent scheduling (SPS), i.e., semi-permanent radio resource allocation, and is required to take into consideration the mobile device's reported transmit buffer status and its radio conditions when making scheduling decisions. This facilitates the delivery of guaranteed bit rate (GBR) bearers.

Transport Layer
IR.92 requires the use of a particular set of radio bearers, specifically, two signaling radio bearers (SRBs) and two or three data radio bearers (DRBs). The standard SRBs – SRB1 and SRB2 – transport the dedicated control channel (DCCH) which is used for signaling message exchange between the mobile device and the base station (RRC messages), as well as between the mobile device and the MME (NAS messages). These radio bearers always operate in RLC acknowledged mode (AM).

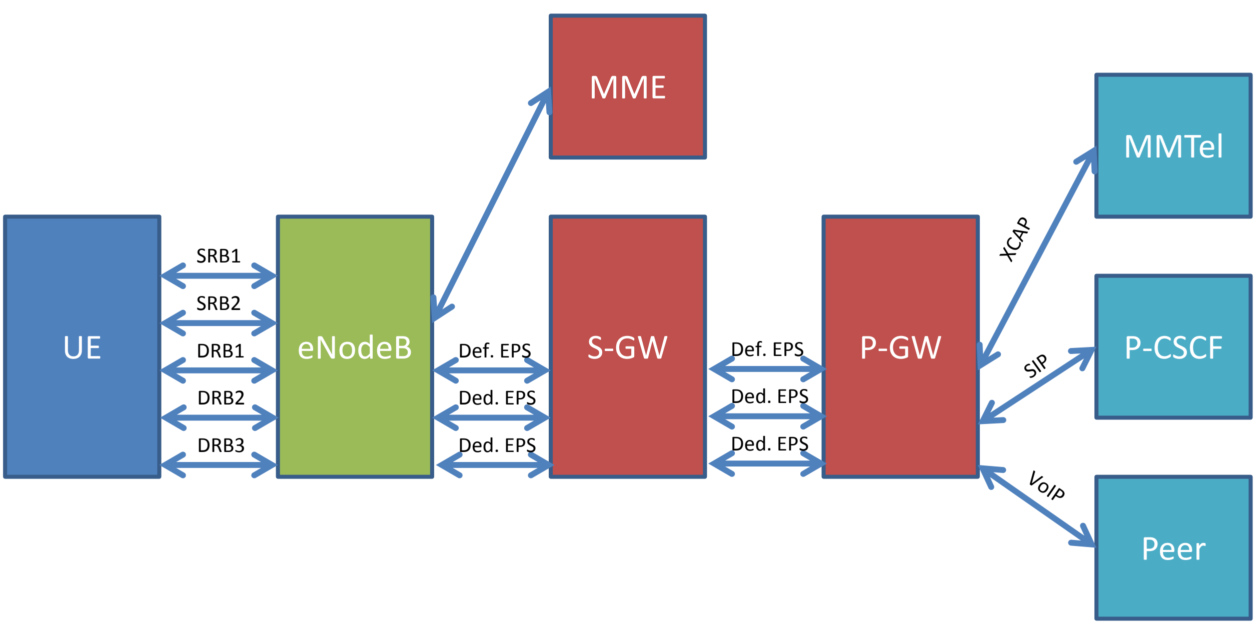
The two (or three) DRBs carry SIP (call control) and XCAP (service management) signaling as well as voice packets. The first DRB corresponds to the default EPS bearer (see below) and is used to transport SIP and XCAP signaling messages. The associated QCI is 5, which means that the bearer operates in non-guaranteed bit-rate (NGBR) mode, but with high priority, low latency, and very low packet loss. The second DRB corresponds to a dedicated EPS bearer and is used to transport voice packets. The associated QCI is 1, which means that the bearer operates in GBR mode with high priority and low latency, but with relatively higher packet loss. There may also be an additional DRB operating with QCI 8/9 (low-priority, high-latency, very low packet loss) used for the transport of XCAP messages. A summary of QCI definitions appears below:

QCI	Packet Delay Budget (ms)	Packet Loss Rate (%)
1	100	1
2	150	0.1
3	50	0.1
4/6	300	0.0001
5	100	0.0001
7	100	0.1
8/9	300	0.0001

Core Networking

Access Points and EPS Bearers
All networks and mobile devices are required to utilize a common access point name (APN) for VoLTE, namely, "IMS". Unlike many legacy networks, LTE networks employ the "always-on" conception of packet connectivity: Devices have PDN connectivity virtually from the moment they perform their initial attach to the core network. During the initial attach procedure, some devices choose to name the access point through which they prefer to connect. However, mobile devices are *not* permitted to name the VoLTE APN during initial attach, i.e., to utilize the IMS as their main PDN, but rather to establish a connection with the IMS AP separately. Thus, VoLTE devices must support multiple simultaneous default EPS bearers.

Note that because the VoLTE APN is universal, mobile devices will always connect through the *visited* PLMN's IMS PDN-GW. This architecture also implies the non-optionality of the P-CSCF:



As stated, VoLTE sessions employ two or three DRBs. This, in turn, implies the use of one default EPS bearer plus one or two dedicated EPS bearers. The default EPS bearer is always used for SIP signaling and exactly one dedicated EPS bearer is used for voice packets (regardless of the number of active voice media streams.) XCAP signaling may be transported on its own dedicated EPS bearer – for a total of three active EPS bearers – or it may be multiplexed with the SIP signaling on the default EPS bearer, in which case only two EPS bearers are utilized.

IP and Header Compression

By default, the mobile device is required to operate in “dual-stack” IPv4/IPv6 mode. If the IMS AP does assign an IPv6 address, the device is required to prefer that address and specifically to utilize it during P-CSCF discovery.
Since IP overhead in VoIP calls can be substantial – and since air interface capacity is precious – IR.92 mandates IP header compression – specifically robust header compression (RoHC) – for the voice packet stream. Prior to transmitting a packet over the voice DRB, the mobile device and PDN-GW employ RoHC which reduces the 40 or 60 header bytes per IP packet to one or three bytes per packet.

Service Continuity

Within the LTE/SAE network, mobility is handled for VoLTE implicitly, by the defined mobility management mechanisms. A challenge arises, however, during mobility to or from UMTS or GSM. UMTS and GSM utilize a wholly different call model than VoLTE. The call processing and service delivery entities are the MSC/VLR and HLR rather than the CSCFs and MMTel servers. Furthermore, the bearer/transport channels used to exchange voice packets are CS, not PS.

Part of the solution to this problem is known as IMS Centralized Services (ICS). With ICS, voice calls are *always* anchored at the IMS, regardless of in which domain they are originated. This solves the problem of transfer of call control as the mobile device moves between radio access technologies (there is no transfer). But the PS bearer challenge remains.

With UMTS, PS services were supported from inception, so simply transporting voice over a PS bearer is supportable naturally. However, the PS QoS mechanisms in UMTS may not be able to provide a satisfactory user experience. Further, in addition to QoS challenges, GSM/GPRS is extremely limited in terms of its multiple bearer support. This gave rise to the concept of “PS over CS” in UMTS and GSM, in which a CS bearer is utilized to transport voice packets for calls anchored in the IMS. As a workaround to the limited bearer support in GSM, USSD can be used used to transport the IMS call setup signaling.

Voice call continuity during radio handover between LTE and the UMTS/GSM CS domain is accomplished via Single Radio Voice Call Continuity (SRVCC).

Voice over LTE Verification

Again, the GSMA IR.92 reference document does not define new technologies; it simply indicates how the existing specifications should be implemented and configured in order to deliver the requisite voice/video calling and messaging functionality over LTE. But even in the context of this narrowly-defined mission, comprehensive testing of mobile devices is crucial to ensure VoLTE functionality, interoperability, and performance.

Device Test Approaches

The approaches available for VoLTE verification are the traditional ones: There is bench testing utilizing test equipment that emulates the LTE network in terms of RF, protocols, and peer devices. There is lab testing with network infrastructure vendors. And there is field/drive testing. Each approach has its benefits and all three will be utilized during a VoLTE service launch.

Emulator-based bench testing, in particular, is flexible in that it permits testing at successively higher levels of system integration. At the lower levels, 3GPP itself provides test cases covering general LTE functional areas such as RF, radio resource management, and protocols. These particular areas are covered in 3GPP TS 36.521-1, 36.521-3, and 36.523-1, respectively. (Of course, there are numerous additional test suites covering ancillary functionality such as the UICC.) Then there are test plans covering VoLTE itself, and finally there are carrier-specific test plans which take into account the carrier’s actual network functionality and use cases of interest.

Functional Testing, Performance Testing, and Quality Testing

At each level of system integration, tests may be focused on basic functionality, or performance, or quality. For VoLTE, some of the more important functional tests include:

- IMS registration
- IMS security procedures

- Device (SIP) addressing
- Call processing including abnormal call handling
- Call conferencing and other supplementary services such as call forwarding
- Call drop and reestablishment
- Emergency calls
- Codecs and DTMF
- Video calls

During performance testing, some areas of key interest are:

- Jitter buffer management
- Lost packet and erroneous packet handling
- Call setup latency
- Call completion rate
- Ringback tones

Note that performance can be particular to the calling scenario – VoLTE-to-VoLTE and VoLTE-to-non-VoLTE – as well as to the mobility scenario – inter-RAT and inter-domain – and it is important to understand a device's performance in each.

Finally, important quality tests include acoustic audio test as defined in 3GPP TS 26.132, PESQ/POLQA scores, acoustic echo, and the effects of transcoders as would occur in VoLTE-to-non-VoLTE calling.
Anritsu provides a full line of test solutions to support VoLTE functional, performance, and quality verification including:

- ME7873L RF and RRM
- MD8475A Signaling tester
- MD8430A Signaling tester
- ME7834L Carrier acceptance and conformance
- MX847570A SmartStudio

Summary

3GPP Long Term Evolution networks are now a commercial reality. As cellular network operators continue to plan LTE coverage and capacity expansion, the need to migrate traditional voice calling services to LTE becomes ever more pressing. The emergence of GSMA IR.92 – IMS Profile for Voice and SMS – as the *de facto* voice-over-LTE standard has cleared the way for global adoption of a single LTE voice solution. IR.92's forward-looking foundation on IMS/SIP ensures its longevity, but it also introduces functional, performance, and quality challenges. Comprehensive testing and verification of VoLTE-enabled mobile devices is crucial to ensuring consumer adoption and commercial success of voice services on LTE.

References

GSMA IMS Profile for Voice and SMS
GSMA LTE Roaming Guidelines
3GPP TS 36.521-1: E-UTRA; UE conformance specification; Radio transmission and reception; Part 1: Conformance testing
3GPP TS 36.521-3: E-UTRA; UE conformance specification; Radio transmission and reception; Part 3: RRM conformance testing
3GPP TS 36.523-1: E-UTRA and EPC; UE conformance specification; Part 1: Protocol conformance specification
Anritsu MD8430A LTE Signaling Tester (Base Station Simulator)
Anritsu ME7873L LTE RF Conformance Test System
Anritsu MD8475A Signaling Tester (Base Station Simulator)
Anritsu ME7834 Mobile Device Test Platform
Anritsu MX847570A SmartStudio

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