White Paper

IMS Architecture

The LTE User Equipment Perspective

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1. EXECUTIVE SUMMARY

The IP Multimedia Subsystem (IMS) dates from 3GPP release 5 over a decade ago, but is now becoming a reality with the rollout of IMS-based LTE networks. IMS enables convergence on multiple fronts, including access types (fixed, mobile), service types, application control functions and convergence between telephony and traditional data delivery.

This paper presents a high-level technical view of the IMS architecture as seen by LTE-capable User Equipment (UE), and is part of a suite of associated literature available from Spirent. Others include:

- White Paper - VoLTE Deployment Challenges and the Radio Access Network – this paper examines the challenges of VoLTE deployment with specific attention on the RAN features required to deliver carrier-grade voice services on the LTE network.

- Reference Guide - IMS Procedures and Protocols: The LTE User Equipment Perspective – discusses related procedures, protocols and sample call flows (including VoLTE).

- IMS/VoLTE posters:
    - LTE and the Mobile Internet – a high-level multi-generational architectural diagram connecting radio access networks, core networks and application servers.
    - IMS/VoLTE Reference Guide – a convenient reference relating industry specifications to the topics most often addressed by mobile device designers.

CORRESPONDING LITERATURE

WHITE PAPER
VoLTE Deployment Challenges and the Radio Access Network

REFERENCE GUIDE
IMS Procedures and Protocols: The LTE User Equipment Perspective

POSTERS
LTE and the Mobile Internet
IMS/VoLTE Reference Guide

2. INTRODUCTION

Over the past several years the IMS has been a topic of discussion for anyone connected with the wireless industry. Since the introduction of IMS has most significantly affected wireless network equipment and its deployment, much of the attention has been paid to the network itself. However, IMS and the deployment of LTE have a significant effect on the operation of mobile devices.

This paper provides an overview of IMS, its architecture and applications from the perspective of the LTE User Equipment (UE). It also provides a look at the evolution to a data-only LTE network and includes a discussion of the challenges and requirements to support delivery of voice services (including VoLTE) over an all-IP network.

3. WHY IMS?

The history of IMS began with the 3G.IP, a now-defunct consortium of major industry influencers. In the late 1990's AT&T, BT, Rogers Cantel, Ericsson, Lucent, Nokia, Nortel Networks, Telenor TIM and others banded together to bring an all-IP network to UMTS systems. The stated plan was to build on an evolved GPRS core network and W-CDMA and EDGE air interfaces. At that time, IMS was thought to be solely intended for wireless communications.

As IMS evolved, it became clear that the original stated requirements (such as voice transcoding, interconnection between domains, access independence and a rudimentary concept of presence) could lend itself to bridging gaps between wireless and wired networks, addressing one of several definitions of "convergence".

3.1. THE ALL-IP NETWORK

For years, any mention of IMS simply referred to it as the "flat, all-IP network". The evolution of communications makes it clear that we are trending towards the efficiency offered by all-digital networks. Yet the Public Switched Telephone Network (PSTN) implements concepts that have been in use since the early days of telephony… circuit-switching is the classic example. An all-IP network promises vast cost savings and greatly increased efficiency.

As a result of the gradual evolution of telephony, digital traffic is often packaged as payload data in other protocols. While the development of LANs and the Internet made IP the ubiquitous de-facto method of data transfer, digital telephony often requires that IP packets are distributed as payload over other switching & distribution techniques. For example, IP packets eventually delivered to a mobile device may have been packaged into ATM cells which were transmitted within SONET frames. The realization of a true all-IP network eliminates the overhead associated with multiple types of switching at multiple connection layers.

3.2. THE BIG CONVERGENCE

The term “convergence” is so widely used in technological circles that it has taken on many different meanings, several of which are being addressed by IMS.

Convergence of telephony and IP services – while today’s subscriber may see this convergence as one that has already taken place within the mobile device, there are costs and inefficiencies involved, due to the fact that these two functions of a phone require connections to multiple networks using separate methodologies of delivery. IMS provides a single network subsystem for all service types, including voice telephony.

Convergence of access technologies – IMS promises to make access technologies almost immaterial, converging common access types (e.g. cellular, Wi-Fi, landline audio, LAN, etc.) around the IMS core.

Convergence of service types – today’s voice, audio and video services each use specific service-to-service protocols, offering the opportunity for IMS to create efficiencies.

Convergence of location – today’s global traveler is connected to mobile applications through complex interfacing of multiple networks and network types. The IMS concept of “presence” addresses the issue of presenting communications and applications consistently and efficiently, without regard to the user’s physical location.

Convergence of control functions – To address tremendous growth in mobile applications, IMS offers a single set of control and routing functions that can be shared by applications, rather than the application-specific control and routing used today.

3.3. VALUE ADDED BY CELLULAR OPERATORS

IMS offers mobile operators a chance to offer added value in the delivery of data and applications. The most prominent example today is the emergence of voice traffic. Voice-over-IP (VoIP) codecs make it possible for any IP-based system, even the public Internet, to deliver better-than-POTS quality audio as a commodity.

However, the Internet is not equipped to guarantee levels of service consistent with the public’s expectations for voice telephony. On a generic IP-based Public Data Network (PDN), load-balancing, latency and a host of other parameters are done on a best-effort basis. By controlling the IMS core, cellular network operators are able to offer specific Quality-of-Service (QoS) based on purchased service levels and on the requirements of the applications themselves (e.g. latency requirements for voice).

4. IMS ARCHITECTURE

Most discussions of the IMS include a graphic portrayal of its architecture in terms of a single flat network[1] or as three separate layers[2]: the transport layer, the IMS layer and the service/application layer. While it is useful to note that IMS is a multi-layered architecture (minimizing the number of connections required when compared to a truly flat architecture), for the purposes of this paper the network is best understood as the combination of user equipment (UE), transport, control functions and the applications. Figure 1 depicts a simplified view of the related network from the point of view of the UE. For a more detailed depiction of the related network connections, a poster titled LTE and the Mobile Internet is available for download from Spirent Communications.

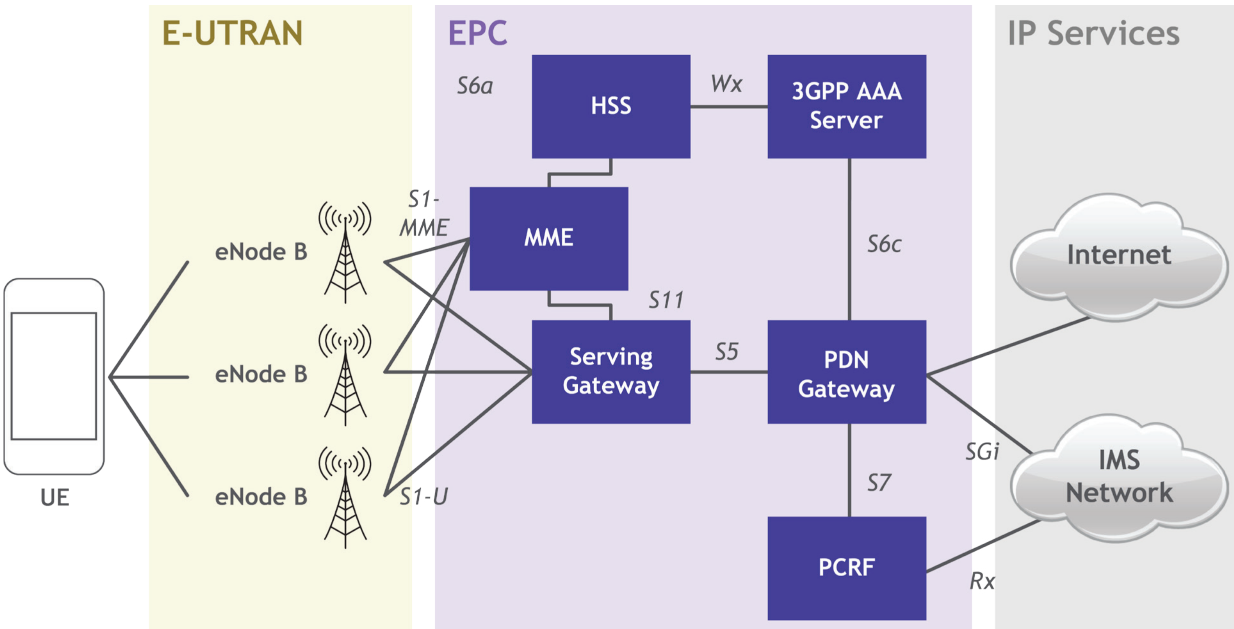

Figure 1 - IMS with the LTE Evolved Packet Core

4.1. The UE

The UE is the terminal of the IMS architecture, and resides with the user. In IMS, the UE contains a Universal Integrated Circuit Card (UICC) and a Session Initiation Protocol User Agent (SIP UA).

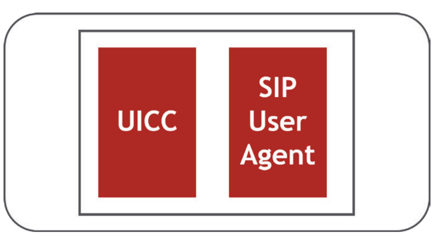

Figure 2 - The IMS-capable UE

SIP, the protocol used for IMS messaging, is defined in the IETF's RFC 3261[3]. It is described in detail in a Spirent reference guide titled IMS Procedures and Protocols: The LTE User Equipment Perspective.

1 3GPP TS 23.228: "IP Multimedia Subsystem (IMS); Stage 2"
2 TISPAN ES 282 007: "IP Multimedia Subsystem (IMS); Functional architecture"
3 Internet Engineering Task Force (IETF) RFC 3261: "SIP: Session Initiation Protocol"

4.1.1. Universal Integrated Circuit Card (UICC)

Each UE includes a UICC, a smart card that contains one or more applications. The applications may be any or all of the following:

- Subscriber Identity Module (SIM) – identity information used by a GSM network.
- UMTS Subscriber Identity Module (USIM) – identity information used by a UMTS or LTE network.
- CDMA Subscriber Identity Module (CSIM) or Re-Useable Identification Module (R-UIM) – identity information used by a CDMA network.
- IP Multimedia Services Identity Module (ISIM) – identity information used by the IMS subsystem.

The ISIM contains:

- IP Multimedia Private Identity (IMPI) – Permanently allocated global identity assigned by a user's home operator. It is analogous to the International Mobile Subscriber Identity (IMSI) used in legacy technologies and is transparent to the subscriber. It includes the home operator's domain information.
- The home operator's domain name.
- IP Multimedia Public Identity (IMPU) – Used to request communication with another user, the IMPU can be roughly thought of as analogous to a telephone number. It can be either a sip URI, which resembles an email address in appearance (sip:‹username›@‹host›:‹port›) or a tel URI as defined in RFC 3966[4] (tel:‹country_code›‹national_destination_code›‹subscriber_number›). A device may have multiple IMPUs, and multiple devices may share an IMPU.
- A long-term secret used to authenticate and calculate cipher keys. IMS actually does multiple levels of authentication: with the transport network, with the radio access network (RAN), with the IMS core, etc. This long-term secret is used in SIP registration.

If an ISIM is not present, a UE will default to using the USIM or CSIM.

4.1.2. The SIP User Agent (SIP UA)

The SIP UA is the logical terminal of the SIP network and both transmits and receives SIP messaging. It also manages the SIP session from the terminal end.

In general, the SIP UA can be thought of as providing typical telephone functionality (e.g. dial, answer, hold, transfer, etc.) via two separate roles:

- UAC (User Agent Client) – Sends SIP requests.
- UAS (User Agent Server) – Received requests and sends SIP responses.

4 Internet Engineering Task Force (IETF) RFC 3966: "The tel URI for Telephone Numbers"

4.2. THE EVOLVED PACKET CORE (EPC)

The all-IP EPC used in LTE is a part of the transport block of the architecture, where “transport” is the entity through which the overall network (e.g. the LTE Evolved Packet System [EPS]) is accessed. The transport block includes backhaul/backbone as well as the access network.

4.2.1. The Public Data Network Gateway (PDN-GW or PDG)

The PDN-GW is a well-known entity in legacy digital networks, offering the UE access to public digital networks (e.g. the Internet). In IMS there are typically separate PDN-GWs offering access to the Internet and the IMS network.

In the case of LTE, the PDN-GW also serves as a mobility anchor point for users moving between LTE services and non-3GPP services.

4.2.2. Policy and Charging Rules Function (PCRF)

The PCRF provides real-time determination of what types of traffic are allowed under what conditions, and also determines how to account for this traffic (for billing purposes). Based on requests for IMS services, the PCRF also initiates the appropriate bearers. Examples of PCRF functions might be:

- If a multi-user game is offered and the user attempts to start the service, the PCRF will determine whether that user is authorized for the service.
- A network operator may determine that third-party VoIP services are allowed to use Wi-Fi connections but not cellular connections. When a VoIP application is launched, the PCRF will determine whether the application may continue.
- If a user attempts to launch a VoLTE call (and is authorized to do so), the PCRF will initiate the setup of the dedicated bearer.

4.3. THE IMS CORE

The IMS core provides session and media control.

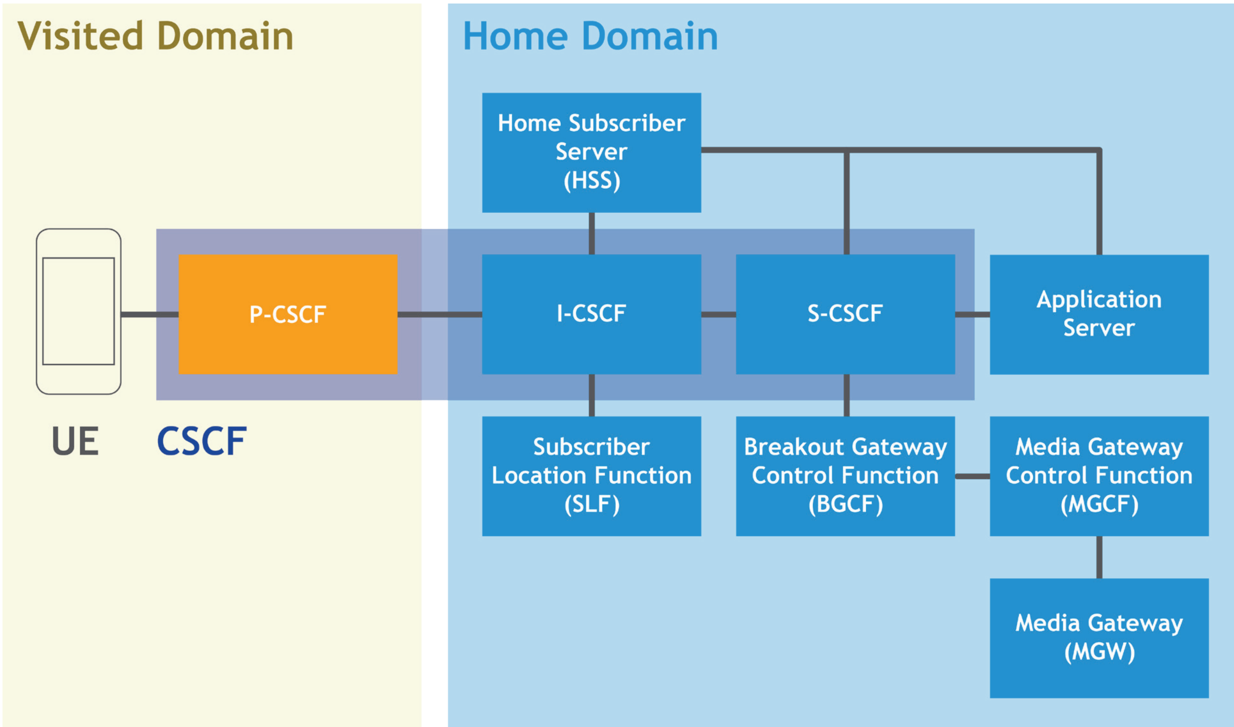

Figure 3 - Interaction between the CSCF, HSS and other elements

4.3.1. Call Session Control Function (CSCF)

The CSCF is responsible for establishing, monitoring, supporting and releasing multimedia sessions. It is comprised of three separate entities which may or may not be separate physical entities:

4.3.1.1. Proxy CSCF (P-CSCF)

The P-CSCF is seen as the initial point of contact from any SIP User Agent. It handles all requests from the UE and is, from the UE's point of view, the "SIP proxy" to the entire subsystem (via the I-CSCF and/or S-CSCF). It may include a Policy Control Function (PCF) responsible for enforcing QoS policies on media. In terms of policy-based networking outlined in RFC 2753[5], the PCF is the policy server, or Policy Decision Point (PDP). This is separate from the PCRF described earlier, which enforces policy on the transport network. Logically, the P-CSCF is considered part of the visited network.

4.3.1.2. Serving CSCF (S-CSCF)

The S-CSCF is a SIP server logically seen as part of the home network and is analogous to the Home Location Register (HLR) used in GSM. It "knows" about the user and what applications are available to the user, and is a decision point as to whether or not the user's SIP messages will be forwarded to the application servers.

The S-CSCF also stores addresses used for contacting the UE, so that it can be used in future sessions. It is also the enforcement point of the network operator's policies.

5 Internet Engineering Task Force (IETF) RFC 2753: "A Framework for Policy-based Admission Control"

4.3.1.3. Interrogating CSCF (I-CSCF)

The I-CSCF is the entity that initiates the assignment of a user to an S-CSCF (by querying the HSS) during registration. It is "seen" by the IMS core as a proxy to an individual user and is a liaison for SIP messaging between the user (via the P-CSCF) and the S-CSCF.

4.3.2. Home Subscriber Server (HSS)

The HSS is a database that maintains user profile and location information and is responsible for name/address resolution. It is also responsible for authentication and authorization, but unlike in legacy technologies, authentication with the radio access network and the core can be different.

4.3.3. Subscriber Location Function (SLF)

The SLF keeps track of multiple HSSes in a home network, and is responsible for assigning one to a user.

4.3.4. Media Gateways

For detailed descriptions of the gateway interfacing between SIP-based networks and the legacy PSTN, see RFC 3372[6].

4.3.5. Media Gateway Control Function (MGCF)

The MGCF controls media gateways (MGWs), performs transcoding (converting codecs, for example from EVRC to WB-AMR) and the conversion of media between the Real-time Transport Protocol (RTP) used in IMS and the Pulse-Coded Modulation (PCM) used by a circuit-switched network.

Depending on how a network equipment manufacturer decides to implement, the MGCF may also serve as the breakout to a circuit-switched network. In that case the MGCF is also responsible for managing the conversion of signaling messages, converting SIP messaging to the Bearer Independent Call Control (BICC) and ISDN User Part (ISUP) protocols used in legacy systems.

4.3.6. Breakout Gateway Control Function (BGCF)

If an MGCF does not include the breakout to a circuit-switched network, that functionality is performed by the BGCF. When the BGCF does control this breakout it does so by selecting an MGCF (either in the same IMS network or another IMS network) or by selecting an MGW (on a non-IMS-based network).

6 Internet Engineering Task Force (IETF) RFC 3372: "Session Initiation Protocol for Telephones (SIP-T): Context and Architectures"

5. VOICE SERVICE OVER LTE

5.1. EVOLUTIONARY STEPS

A primary goal of LTE is to provide telco-grade voice services over a data-only LTE network.

Until VoLTE is ready for widespread commercial deployment, operators are faced with the challenge of providing call continuity between LTE and legacy circuit-switched networks. As shown in Figure 4, 3GPP2 (CDMA) and 3GPP (legacy UMTS) voice services have evolved in slightly different ways.

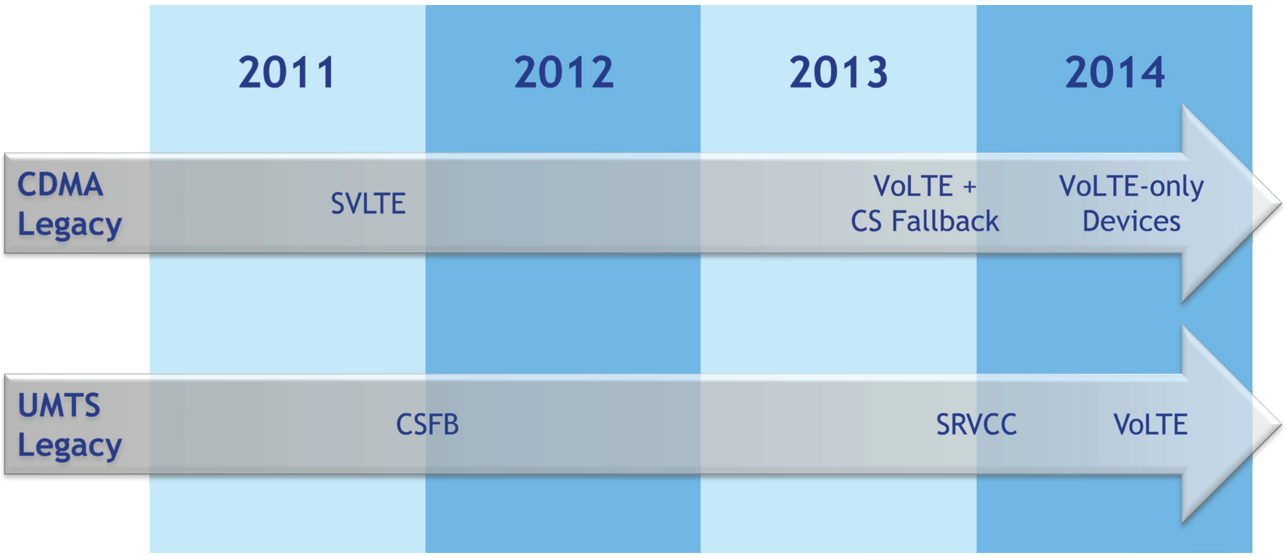

Figure 4 - Evolution of Voice Services with LTE Deployment

5.1.1. Simultaneous Voice and LTE (SVLTE)

For legacy 3GPP2 operators, SVLTE uses two radios to simultaneously communicate with:

- 1X network for services such as CS Voice, SMS, Emergency Services
- LTE network for high-rate PS data services

While this approach enables rapid deployment, it is not meant to be more than an interim measure. For one thing the cost of two radios is absorbed into each and every SVLTE capable device. Other potential issues involve interference between the radios, concern for exceeding maximum allowable output power levels (enforced per device, not per band or per radio) and, of course, battery life.

5.1.2. Circuit Switched Fallback (CSFB)

CSFB provides 3GPP network operators with a means to move from LTE to UMTS/GSM (or even 1X) services when circuit-switched services (voice, SMS) are needed.

CSFB does allow for a single-radio (or single transmitter, dual receiver) design. Like SVLTE, it offers complete set of circuit-switched services and features, even though the device is primarily operating in LTE mode. However, packet-switched services are degraded when used on the slower legacy packet-switched network… this is an issue because depending on the type of CSFB being used, packet-switched bearers may be interrupted. Finally, the fallback mechanism takes some time, which translates into longer call setup times. Using this scheme, call setup can take as long as a half a second.

The CSFB type used depends on the network available to fall back on, as well as the specifications release being adhered to, as outlined in Table 1.

Destination RAT	Option	3GPP Release
UMTS	RRC Connection Release with Redirection (w/o Sys Info)	Release 8
UMTS	RRC Connection Release with Redirection (w/ Sys Info)	Release 9
UMTS	PS Handover with DRBs	Release 8
GSM	RRC Connection Release with Redirection (w/o Sys Info)	Release 8
GSM	RRC Connection Release with Redirection (w/ Sys Info)	Release 9
GSM	PS Handover with DRBs	Release 8
GSM	Cell Change Order (w/o NACC)	Release 8
GSM	Cell Change Order (w/ NACC)	Release 8

Table 1 - CSFB Techniques

5.1.3. Voice Over LTE (VoLTE)

VoLTE is the service that once widely deployed, enables operators to provide improved QoS over legacy circuit-switched voice service and "best-effort" Over The Top (OTT) services. VoLTE is defined in the GSM Association's (GSMA's) Permanent Reference Document IR.92[7]. The document is intended to ensure interoperable SIP-based IMS VoIP and SMS for UE's and the LTE EPC. It defines basic IMS capabilities and supplementary services for telephony, real-time media negotiation, transport and codecs, LTE radio and EPC capabilities (such as establishing bearers and QoS) and functionality that is relevant across the protocol stack and subsystems.

Note that IR.92 provides a profile of *minimum* mandatory 3GPP capabilities.

A second related document, the GSMA's IR.88[9] provides guidance for LTE roaming scenarios.

5.1.4. Single Radio Voice Call Continuity (SRVCC)

SRVCC allows a PS/IMS-based (VoLTE) Voice Call to transition to a legacy CS network. Unlike SVLTE and CSFB, SVRCC does enable call continuity. SRVCC uses a single radio, and allows an operator to provide ubiquitous voice coverage, even when LTE coverage is not complete.

However, the signaling required is complicated. The result is that there may be a brief break in audio service when the call is transitioning to the circuit-switched network.

7 GSM Association Official Document IR.92: "IMS Profile for Voice and SMS"
8 3GPP TS 24.229: "Technical Specification Group Core Network and Terminals; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3"
9 GSM Association Official Document IR.88: "LTE Roaming Guidelines"

5.2. Requirements for Supporting VoLTE

From the UE's point of view, there are four non-obvious requirements for a network to support VoLTE. The first three of these, Semi-Persistent Scheduling, Transmission Time Interval Bundling and Discontinuous Reception, are implemented at the MAC sub-layer. The fourth, Robust Header Compression, is implemented in the Packet Data Convergence Protocol (PDCP) sub-layer.

5.2.1. Semi-Persistent Scheduling (SPS)

In LTE, DL and UL traffic channels are dynamically shared. The control channel (PDCCH) must be used to identify which sub-frames should be decoded on the downlink PDSCH and which users are allowed to transmit in each UL sub-frame (on the PUSCH). Without SPS, every Physical Resource Block (PRB) on the downlink and uplink must be explicitly granted; the resulting overhead is inefficient for traffic that requires continual allocations of small packets (such as VoIP).

This issue is addressed by SPS, which defines a transmission pattern and, based on that pattern, assigns a pattern for PRBs to use going forward (unless there is a reason to change the pattern). As an example, suppose a voice service uses one coded packet every 20ms. During silent periods, PRB assignments can be canceled. In the uplink they can be implicitly canceled after a defined number of empty UL transmissions. In the downlink they can be canceled with a Radio Resource Control (RRC) message.

5.2.2. Transmission Time Interval (TTI) Bundling

In order to reduce end-to-end latency, LTE introduced the idea of the short TTI (1 ms). This means that the Hybrid Automated Request (HARQ) process is meant to acknowledge transmissions every 1 ms. However, at cell edges a UE might not have enough time available to reliably deliver an entire VoIP packet in one TTI.

The solution is to bundle multiple TTIs together without waiting for HARQ feedback. A VoIP packet is sent as a single packet data unit (PDU) during a bundle of subsequent TTIs, and the HARQ feedback is only expected after the last transmission of the bundle. As in legacy technologies, RRC protocol is used to configure TTI bundles.

5.2.3. Discontinuous Reception (DRX)

A constantly-on voice session can quickly reduce battery life. Since VoLTE traffic is highly predictable (e.g. 20ms codec packets), a UE receiver does not have to constantly monitor the PDCCH, and the receiver can essentially be turned off between receptions. This must be carefully configured, though, since missing acknowledgements or HARQ messages can add unacceptable latency.

5.2.4. Robust Header Compression (RoHC)

IP header information can be disproportionately large when compared to the relatively small VoLTE codec packets being transmitted, creating inefficiency in terms of the air interface bandwidth.

For example, a combination of RTP, UDP and IP headers can total 40 to 60 bytes of header data, while using AMR-WB at 14.4 kpbs yields payload data of about 50 bytes per 20 ms frame. In this case there may be more overhead being transmitted than actual payload data. RoHC can sometimes compress headers down to the 2-4 byte range, providing greatly improved efficiencies on the air interface.

6. VIDEO SERVICES OVER LTE

Mobile video streaming and video chat are two services that are surging in growth driven by rapid smartphone and tablet adoption and the rollout of LTE worldwide. As operators implement LTE, IR.94[10]-based video calling will generally follow IR.92 VoLTE in their service portfolio. Operators are keen to establish sufficient levels of service quality and user experience to compete with the increasingly popular Over-the-Top (OTT) video chat services. Various Key Performance Indicators (KPIs) such as Video Mean Opinion Score (V-MOS), frame loss, audio-video sync and video impairment metrics, measured under real and varying network conditions, are used to make improvements in devices, software and infrastructure.

7. CONCLUSION

IMS will, for the first time, shift carriers' voice service offerings to the data realm. VoLTE is the first major IMS-related application being rolled out on a large scale and the stakes are high. The combination of IMS, SIP and RAN features as described in this document are essential in delivering the "carrier-grade" VoLTE experience. UE testing and measurement, initially focused on IMS and SIP functional testing, is now concentrating on both industry and operator-specified test requirements for VoLTE call performance and VoLTE user experience evaluation. Aside from dealing with a network that is literally new to the core, UE designers must consider the layered complexity of a multi-RAT, multi-band IMS-capable UE.

Spirent is a global leader in LTE device testing and is well positioned to support the industry with the many IMS/VoLTE test challenges on the horizon. This white paper is the first in an ongoing series of tools aimed to educate and support UE developers as they contribute to the deployment of IMS/VoLTE. A second white paper, sVoLTE Deployment and the Radio Access Network: The LTE User Equipment Perspective" provides an overview of the complexity of IMS/VoLTE deployment and a detailed understanding of the significant testing challenges.

Please see the Spirent website (www.spirent.com) for other free white papers, recorded seminars, posters and other resources that may be helpful to the UE developer.

10 GSM Association Official Document IR.94: "IMS Profile for Conversational Video Service"

8. ACRONYMS

ATM	Asynchronous Transfer Mode
BGCF	Breakout Gateway Control Function
BICC	Bearer Independent Call Control
CS	Circuit-Switched
CSCF	Call Session Control Function
CSFB	Circuit Switched Fallback
CSIM	CDMA Subscriber Identity Module
DRX	Discontinuous Reception
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
EPC	Evolved Packet Core
EVRC	Enhanced Variable Rate Codec
HARQ	Hybrid Automated Request
HLR	Home Location Register
HSS	Home Subscriber Server
I-CSCF	Interrogating Call Session Control Function
IMPI	IP Multimedia Private Identity
IMPU	IP Multimedia Public Identity
IMS	IP Multimedia Subsystem
ISIM	IP Multimedia Services Identity Module
ISUP	ISDN User Part
MGW	Media Gateway
MME	Mobility Management Entity
OTT	Over the Top
PCF	Policy Control Function
PCM	Pulse-Coded Modulation
PCRF	Policy and Charging Rules Function
P-CSCF	Proxy Call Session Control Function
PDCCH	Physical Downlink Control Channel
PDCP	Packet Data Convergence Protocol
PDG	Public Data Network Gateway
PDN	Public Data Network
PDN-GW	Public Data Network Gateway
PDP	Policy Decision Point
PDU	Packet Data Unit
PRB	Physical Resource Block
PSTN	Public Switched Telephone Network
PS	Packet-Switched
QoS	Quality-of-Service
RoHC	Robust Header Compression
RRC	Radio Resource Control
RTP	Real-time Transport Protocol
S-CSCF	Serving Call Session Control Function
SIM	Subscriber Identity Module
SIP	Session Initiation Protocol
SLF	Subscriber Location Function
SONET	Synchronous Optical Networking
SPS	Semi-Persistent Scheduling
SRVCC	Single Radio Voice Call Continuity
SVLTE	Simultaneous Voice and LTE
TTI	Transmission Time Interval
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UE	User Equipment
USIM	UMTS Subscriber Identity Module
VoLTE	Voice over LTE
WB-AMR	Wideband Adaptive Multi-Rate

SPIRENT

1325 Borregas Avenue
Sunnyvale, CA 94089 USA

AMERICAS 1-800-SPIRENT | +1-818-676-2683 | sales@spirent.com
EUROPE AND THE MIDDLE EAST +44 (0) 1293 767979 | emeainfo@spirent.com
ASIA AND THE PACIFIC +86-10-8518-2539 | salesasia@spirent.com

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