New challenges for UE developers with voice transport over LTE

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Agenda

LTE, Voice and SMS – Overview
IMS, SIP, Network Protocol Considerations
UE and Radio Access Considerations
SRVCC, CSFB, SVLTE and interworking
Voice Quality Testing and Agilent Solutions
Support for Voice with LTE

2G/3G – normal Circuit Switched calls have an allocated resource even during times of inactivity - even when nothing is being said

• Inefficient use of available bandwidth
• Reduced flexibility for resource allocation

LTE – UE will generally only be provided resources when it is necessary – even for voice

• Allows efficient use of network resources. If we are saying nothing we will require no network resources
• Places stress on the network to ensure suitable access timing and quality of service (QoS).

LTE transportation is fully IP – no circuit switched services
Support for Voice with LTE

Most agree that the long term solution for voice is to use VoIP and an IMS based core network - However it will take time for networks to support this

For networks which do not support IMS several technologies are being considered, namely:-

- CSFB (Circuit Switched Fall Back) - single radio approach
- SVLTE or Dual Standby approach (Simultaneous Voice and Data LTE) - dual radio approach
- SRVCC (Single Radio Voice Call Continuity) – Voice on LTE with CS backup
- CSFB, SVLTE and SRVCC all involve some level of I-RAT behavior

VoLGA (Voice over LTE Generic Access), does not appear to be getting a lot of support.
IMS (IP Multimedia Subsystem)

- 3GPP defined IMS in 1999
- All-IP enabler for **value added services**
- After a slow start, growth accelerating 50-70% per annum.

- IMS-based VoLTE prevails over VoLGA
- IMS essential for **Voice over LTE**
- 2009 One Voice; AT&T, Orange, Telefonica, TeliaSonera, Verizon, Vodafone, Alcatel-Lucent, Ericsson, Nokia Siemens, Networks, Nokia, Samsung and Sony Ericsson
- 2010 VoLTE adopted by GSMA
SIP and IMS

- SIP (Session Initiation Protocol) was initially designed to work in an open homogeneous IP network
- SIP provides the signalling required to support call set-up procedures
- SIP also provides many other services (caller id, multi-party & emergency calls...)

PSTN
IP Network
GSM
3G
Telephone Network

SIP server
SIP and IMS

Protocols

- **SIP IPv6**:
  - QoS support
  - Legacy services
  - 999 services
  - Caller id
  - Lawful intercept
  - Link Quality Mgt
  - Firewall Needs

- **Accounting**: Billing, Security

- **Legacy Req**:
  - QoS support
  - Legacy services
  - 999 services
  - Caller id
  - Lawful intercept

- **Architectural req**:
  - SIP
  - ARCH

- **Commercial Req**:
  - SIGCOMP
  - Planning

- **P headers**: Feature extension for 3GPP

- **ENUM**: Maps telephone numbers to IP addresses

- **COPS**:

- **IPsec**: Protocol used to support AAA

- **MD5-AKA**: Security protocols used in 3G

- **DIAMETER**:

- **MD5-AKA**:

- **IPsec**: Security protocol for IP links

- **Common Open Policy Service**: Used to communicate a users QoS policy info to routers

- **SIGCOMP**: Used for compression purposes
IMS uses SIP
(Session Initiation Protocol)

SIP (Session Initiation Protocol)
e.g. INVITE, TRYING, RING, OK, BYE etc.

SDP (Session Description Protocol)
e.g. m (media), a (attribute) etc.
\[ m=audio 49120 RTP/AVP 98 97 \]
\[ a=rtpmap:98 AMR/8000 \]
\[ a=fmtp:98 mode-set=7 \]

RTP (Real-time Transport Protocol)
e.g. AMR encoded speech

RTCP (RT Control Protocol)
e.g. Send/receive quality metrics

CSCF = Call Session Control Function
RTP is used for the delivery of the user data,
RTCP

**RTCP (RFC3550)**
- Packet Loss Rate
- Jitter
- Timing Information

**RTCP XR (RFC3611)**
- Delay
- Signal Level
- Noise Level
- Call metrics
- Buffering
IP Sec

Tunnel Mode (VPN like)
- Entire original IP datagram encrypted
- Client to Server

Transport Mode
- Between two end points
AH added to protect against alteration of datagrams while in transit

Differences between the key generation between LTE and “Vanilla” network

More info in RFC4301, 4302, 4303
SigComp (SIP Signalling Compression)

- Compresses text based SIP and SDP messages
- Up to 3:1 compression
- Standardized by the IETF RoHC working group
SMS over SGs

What if my network does not support IMS, how do I get SMS?

SMS over SGs – Normal circuit switched “native” SMS is encapsulated message within NAS messaging

SGs is actually the interface between the MME and MSC, normally used for mobility management processes between 2G/3G and LTE

Requires combined attach with SMS only update IE.

Example from Agilent PXT, N6062A Message Editor
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VoIP QoS and Multiple PDN’s

Strict packet delay-based QoS

QoS will vary by allocation, by application and will be heavily dependent on system capacity.

UE’s can have multiple data streams, Multiple PDN’s, Addresses, Port numbers etc. – All with different parameters:

• Default DRB or Dedicated DRB
• Guaranteed or non Guaranteed Bit Rate
• Packet delay budget e.g. 50 to 300ms
• Packet Error rate e.g. \(10^{-2}\) to \(10^{-6}\)

Example from Agilent PXT, N6062A Message Editor
Traffic Flow Template

Traffic Flow Template (TFT)
Sort IP packets based on:
• IP protocol; e.g. UDP, TCP
• Port number
• IP address
• Priority

Example configuration

Internet

IMS-SIP

Dedicated

Default

e.g. Browser

IPv4/6 address 1

IPv6 address 2

e.g. RTP voice stream

GBR, high error

Non-GBR, low error

e.g. IMS-SIP signalling
GBR, TFT, QoS, DRB's signalled to UE

Dedicated bearer, linked to default bearer

Example from Agilent PXT, N6062A Message Editor
LTE scheduling optimizations for Voice

SPS
- LTE Web/data optimized scheduling adds overhead and latency for speech
- SPS reduces signalling overhead by making sticky allocations during talk bursts
- Reduces LTE PDCCH loading

TTI bundling
- Up to 4dB cell edge coverage improvement
- Repeat UL data multiple times, reducing probability of errors
- More signaling efficient, fewer errors and lower latency than segmentation and more robust MCS
Message Editor example
SPS and TTI Bundling

Example from Agilent PXT, N6062A Message Editor
Robust Header Compression (RoHC)

Typical VoIP Header for IPv4 = 40 bytes
Typical VoIP Header for IPv6 = 60 bytes

With a typical voice rate of 12 kbps, uncompressed IPv6 headers represent approximately 60% of the data sent / received

LTE network efficiency very poor without RoHC

Robust Header Compression is therefore required for LTE VoIP.

Example from Agilent PXT, N6062A Message Editor
RoHCv2 (Robust Header Compression)

- Cuts IP overhead e.g. RTP streams for speech; 2:1 for IPv4, 3:1 for IPv6
- RoHCv2 simplification robustness handling of out-of-sequence packets
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Phone x Network Configuration

Only one network actively using IMS with LTE today.

Challenges with interoperability between legacy non-3GPP and LTE

LTE designed from IP viewpoint, legacy systems optimised for Voice

Strict latency targets to ensure acceptable voice quality.

Different use models and progression towards VoLTE depending on
existing network RAT.
SVLTE Simultaneous Voice & LTE aka Dual Standby
- Two phones in one case
- 1xRTT (or GSM/W-CDMA) chipset for all voice calls (CS only)
- LTE/eVDO/eHRPD (and/or W-CDMA) separate chipset for data

E.g.

- LTE/eVDO/HRPD radio
- 1xRTT CS radio
- CS Voice Client
- GUI
Phone x Network Configuration

CSFB Circuit Switched Fall Back
• Handover from LTE to legacy 2G/3G for ALL voice calls
• Use Circuit switched voice (and if available parallel slower legacy data)

E.g.

LTE/W-CDMA/GSM radio

CS Voice Client

GUI
Phone x Network Configuration

SRVCC Single Radio Voice Call Continuity
- IMS voice calling in LTE coverage areas
- Quickly handover from LTE to legacy 2G/3G at LTE edge

E.g.

- LTE/W-CDMA/GSM radio
- IMS Voice Client
- CS Voice Client
- GUI
Phone x Network Configuration

LTE / IMS Only
• IMS voice calling in LTE coverage areas

E.g.

- LTE radio
- IMS Voice Client
- GUI

GSM/W-CDMA/ TDSCDMA
1xRTT/eVDO/ eHRPD

Greater insight. Greater confidence.
Accelerate next-generation wireless.
Phone x Network Configuration


1xRTT/eVDO/eHRPD

GSM/W-CDMA/TDSCDMA

SVLTE Simultaneous Voice & LTE Aka Dual Standby

CSFB Circuit Switched Fall Back

LTE/IMS islands

SRVCC Single Radio Voice Call Continuity

LTE / IMS only

Most Operators will skip some steps

World phones will need to roam with many network configurations

E.g.

LTE/eVDO/HRPD/GSM/W-CDMA/TDSCDMA radio

IMS Voice Client

GUI

1xRTT CS radio

CS Voice Client
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Various Test Forms

Voice quality – PESQ, POLQA etc.
- Codec testing, inter-codec rate testing
- Handling for packet delays, lost/un-ordered packets

SARS test

Signalling testing
- Origination, registration, notifications etc.
- Supplementary services – multi-party calls, call on hold etc. – voice and data interactivity
- Signalling Conformance Testing – 3GPP TS 34.229

Radio testing for VoIP functions
- Semi-persistent scheduling, TTI bundling, dynamic small allocation testing

Handovers, SRVCC, CSFB

Battery drain

SMS and Video call testing

Operator Specific Test Plans

Field Testing
Testing to improve user experience

Start with infrastructure IOT, quickly move to simulated bench-top networks then automated conformance testing of new UEs.
IFT for 2G/3G battery drain analysis

Server:
- FTP server
- UDP Server
- Apache HTTP server
- MMS/SMS server

Client:
- Interactive Functional Test (IFT)
- Wireless Protocol Advisor
- License Keys
- Modem drivers

PSU:
- Current monitoring

8960:
- 2G/3G BS emulation
IFT for 2G/3G battery drain analysis

- Server:
  - FTP server
  - UDP Server
  - Apache HTTP server
  - MMS/SMS server

- Client:
  - Interactive Functional Test (IFT)
  - Wireless Protocol Advisor
  - License Keys
  - Modem drivers

- IFT:
  - Activities: SMS, FTP, HTTP, USP, calls
  - Parallel threads
  - UE automation
  - Drag & Drop programming
  - VB scripting

Anticipate  Accelerate  Achieve
IFT for 2G/3G/LTE battery drain analysis

Client:
- Interactive Functional Test (IFT)
- Wireless Protocol Advisor
- License Keys
- Modem drivers

Server:
- FTP server
- UDP Server
- Apache HTTP server
- MMS/SMS server
- IMS-SIP server/client

IMS-SIP server and client:
- SMS-IMS
- VoIP, Video
- Logging

PSU:
- Current monitoring
- 2G/3G BS emulation

PXT:
- LTE BS emulation
- 100Mbps
- Multiple DRB
- IPv6

IMS - SIP server and client:
- SMS-IMS
- VoIP, Video
- Logging

IMS/SIP Server Client for VoLTE Test

Key VoLTE features

SIGCOMP
P headers...

Hello!

Multiple Dedicated DRBs
RoHC
SPS
TTI bundling

Logger

Message Log

CODEC support...

PXT LTE cell
E6621A

SIP client

UE

Hello!

Anticipate  Accelerate  Achieve
Audio test scenario 1
Human jury testing and/or PESQ with IP impairments

Agilent IMS-SIP server

Delay/Jitter/Loss insertion

Audio in/out headphone jack or HATS

VoLTE UE

RF

Audio Analyzer

Agilent PXT

Agilent PXT

VoLTE UE

RF
Audio test scenario 2
Parametric audio quality and noise suppression

Optiona audio noise

Agilent IMS-SIP server + Client

Audio Analyzer

Reference Audio I/O

Analogue audio I/O

Analogue audio to HATS Mic/Speaker or headphone jack

Optional Delay/Jitter/Loss insertion

Ethernet

VoLTE UE

RF
Functional test scenario x
IMS/CS voice calling & Inter-RAT scenarios

Agilent IMS server and several remote clients

Agilent 8960 1xRTT cell
Agilent 8960 eHRPD cell
Agilent PXT LTE cell

Test automation

Ethernet

RF

Agilent IMS server and several remote clients

Agilent 8960 1xRTT cell
Agilent 8960 eHRPD cell
Agilent PXT LTE cell

Test automation

Ethernet

RF
Verizon Inter-RAT Test Automation

www.agilent.com/find/N5973A

Leading Inter-RAT test coverage

• Efficient, repeatable testing: Fully automated execution including UE control and report generation, runs unattended

• Includes IMS-SIP server and IPv6

• Extend to SVD/SVLTE Inter-RAT as UE available

• Builds on Agilent Interactive Functional Test Software, expandable to other test plans

Supports Agilent’s established 8960 and PXT test sets
Useful references

ITU-T P.862. Perceptual evaluation of speech quality (PESQ)
ITU-T P.863. Perceptual Objecting Listing Quality Assessment (POLQA)

PXT Website.  www.agilent.com/find/PXT

Agilent IMS/SIP  www.agilent.com/find/E6966A

Interactive Functional Test (IFT).  www.agilent.com/find/IFT

GSMA IR.92 IMS Profile for Voice and SMS

3GPP TS 34.229 IP Multimedia call control protocol based on SIP and SDP, UE conformance specification

3GPP TS 33.178 Security Aspects of early IP Multimedia Subsystems (IMS)

3GPP TS 26.114 IP Multimedia Subsystems (IMS) Multimedia telephone: Media handling and interaction

3GPP TS 26.132 Speech and video telephony terminal acoustic test specification

3GPP TS 22.173 IP Multimedia Core Network Subsystem Multimedia Telephony Service and supplementary services

3GPP TS 23.228 IP Multimedia Subsystem (IMS) Stage 2
Summary

IPv6, IP Sec, SigComp, Multiple DRB’s, QoS, Multiple IP addresses, IMS, SIP, TTI Bundling, Semi Persistent Scheduling, VoIP, POLQA, PESQ, RTP, RTCP, SRVCC, CSFB, SVLTE, SDP, SMS over SGs

Enough reasons to start testing VoLTE now?

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LTE 3GPP Stack overview – PDCP

Robust Header Compression (RoHC)

- For more info see IETF RFC 4995.
- Reduced overhead, more efficient

Once RoHC has been applied the whole packet (data AND header) are ciphered as per 35.201 (data only)

Headers and Message Authentication codes are added
POLQA ITU-T P.863
Perceptual Objective Listening Quality Assessment

http://www.polqa.de/index.html
© Optimcom GmbH
Rich Communication Suite (RCS)

RCS is an industry effort focused on the use of IMS (IP Multimedia Subsystem) for providing mobile phone communication services. "Rich Communication" in itself is meaningless jargon, which refers to the use of more than just voice for communication, but has long been touted as a benefit of IMS. It is to be noted that much of the capability of RCS is already available from Internet service providers.

According to the press release,[1] from the end-user point of view RCS would enable communication such as instant messaging, video sharing and buddy lists. These capabilities would be available on any type of devices using an open communication between devices and networks. Main features of RCS are:

Enhanced Phonebook, with service capabilities and presence enhanced contacts information

Enhanced Messaging, which enables a large variety of messaging options including chat and messaging history

Enriched Call, which enables multimedia content sharing during a voice call. For many years Nokia and Ericsson have been pushing a "see what I see" capability, but this has failed to be taken up by operators, or the way it was deployed failed to result in great take-up by consumers. Bundling this feature with the two other features may help to increase uptake.

Wider and large scale IMS deployment, Interoperability between different terminal vendor RCS clients and RCS service interworking between operators are the key aims of RCS Initiative. However, RCS is considered by some to be not fit for purpose [2] and has not been rolled out commercially by any major operator as of September 2010. [1] http://www.nokia.com/A4136001?newsid=1189463
LTE VoIP Test Building Blocks

Agilent

Agilent IMS-SIP Server

Agilent IMS-SIP Client

Agilent PESQ Measurement IP

Agilent PXT LTE Test Set

Agilent U8903A Audio Analyzer

Partners of Agilent

IMS-SIP TTCN-3 Signaling Test

Other Measurement IP e.g. POLQA

Standards compliant HATS testing
CSFB

- Procedure

  - PXT sends either RRC release or Mobility from E-UTRA command
  - UE joins 3G cell
  - The UE is redirected to the 8960 and automatically completes the call setup. The 8960 receives a page response from the UE and continues to setup the call.
CSFB (c2k requires s101 control plane tunnel)

- CSFB requires 2 baseband ICs (LTE + alternative RAT) with some opportunity to share RF resources
- For c2k there is significant additional protocol complexity vs. SV-LTE
- For c2k CSFB a control plane tunnel is required in order to pre-register with the c2k cell (via LTE signaling) – this is known as an optimized handover
- 2G/3G CSFB does not need this as all RATs share the same core network
- CSFB is being considered for voice on many 2G/3G/LTE networks
- VZW will ultimately support voice using IMS and will then use optimized handovers between LTE and eHRPD
- Agilent are developing optimized handovers for the PXT/8960.