Validating VoLTE

A Definitive Guide to Successful Deployments

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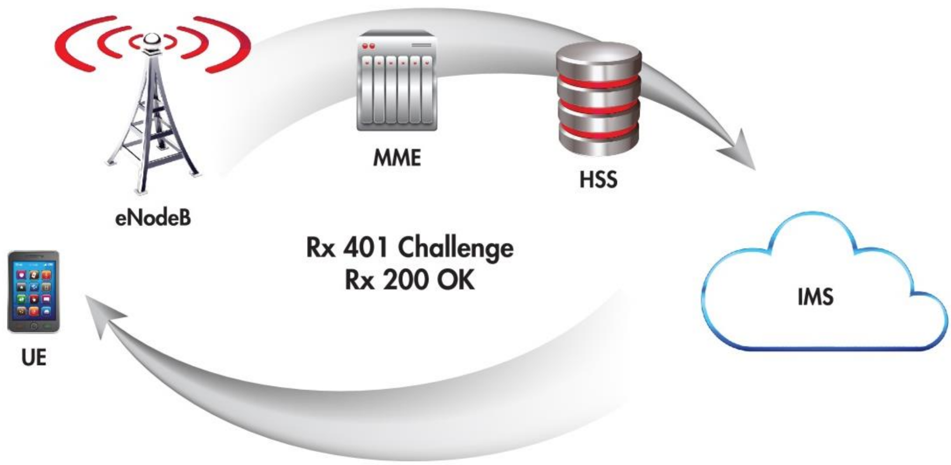
Introduction

This book outlines a VoLTE (Voice over Long Term Evolution) test plan that ensures a correct, stable, and effective VoLTE deployment. These scenarios cover major functional and characterization requirements of a VoLTE network. Each test provides a description, test steps, and expected results. The test plan provides significant benefits when executed before deployment, and also as part of an ongoing regression environment as network elements are upgraded and expanded over the network lifetime. This book is a collection of input gathered from our work with leading equipment vendors and mobile operators globally.



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Overview

VoLTE will be among the most critical and complex technologies mobile operators will ever deploy. The first major service to leverage LTE's comprehensive QoS and policy control framework, VoLTE sets the stage for operators to offer differentiated services, adopt new pricing structures, and partner or compete with over the top (OTT) players.

But with voice services still accounting for a significant percentage of revenues, the quality delivered via 4G networks will need to be toll-grade, as good as or better than 3G. Additionally, VoLTE is not just for voice, but rather a complex mix of voice, video, and messaging services.

With so much at stake, ***mobile operators MUST move away from relying on equipment vendors to validate performance.*** While it is not uncommon for vendors to state performance and capacity numbers for specific or "best case" configurations, the diverse new challenges posed by VoLTE require operators to supplement the testing performed by vendors with their own.

Getting the framework—and network design—right from the beginning plays a major role in delivering the ultimate quality subscribers expect, and freeing creative operators to be fully adaptive in rolling out advanced services. The ability to validate the functionality, quality, resiliency, and scalability of these new networks and services on their own gives operators a clear advantage, and ranks among the most significant challenges facing the industry today.

Validating VoLTE: A Definitive Guide to Successful Deployments

This new book, authored by Ixia VoLTE experts, creates a framework and outlines a step-by-step approach for thoroughly evaluating all aspects of a VoLTE implementation. The book delivers critical insight and understanding of:

- VoLTE's unique challenges
- Potential pitfalls of poor device selection and network design
- Best practices and methodologies for guaranteeing mobile subscribers' QoE
- How leading operators around the world are validating VoLTE in the lab before going live

Validating VoLTE

Validating VoLTE helps ensure a stable, high-performing VoLTE deployment with scenarios covering major functional and performance characterization requirements. By following the approach outlined in the guide, operators gain significant benefits prior to deployment, and in ongoing regression test environments as the network evolves and changes over time.

A collection of input gathered from Ixia's work with leading mobile operators and equipment vendors worldwide, the book features dozens of detailed test procedures covering all major functional and performance elements. Operators can benefit from more than 40 test cases, each containing a description, test steps, and expected results.

Validating VoLTE overviews major categories of system evaluation including:

- Functional validation
- Capacity planning and network scalability
- Effectiveness of QoS and policy control mechanisms
- Voice and video quality
- System interoperability
- Network resiliency and error handling

Mobile operators need to know the scalability and bottlenecks of their specific network designs, service offerings, and call profiles. And to compete in today's high-profile LTE market, they require the flexibility to quickly and confidently roll out new services and charging plans without having to go back to their vendors every time the network changes.

Market Need for VoLTE

Even with the fast rise of web and data services in mobile networks, voice and SMS services still generate around 70 percent of total operator revenues globally. With these revenues at stake, LTE operators must have a path to provide high-quality, toll-grade voice services natively on LTE networks. Unlike previous mobile systems, LTE is designed as a packet switched all-IP system; it does not include a separate circuit-switched domain currently used for voice and SMS services in 3G networks.

Additionally, there is a technology requirement to allow mobile operators to move beyond basic voice and texting. For several years, OTT application providers have delivered very popular voice, video, messaging, and location services that are shifting consumers' attention and usage. Skype and GoogleTalk have nearly a billion registered users world-wide. Apple has sold countless iPhones and iPads, many of which are capable of FaceTime video-calling.

With VoLTE (GSMA VoLTE IR.92 specification, based on global 3GPP standards), mobile operators can deliver a new conversation experience of enriched voice, enlivened video, and intuitive messaging. VoLTE unlocks these richer conversation services and lays a foundation for operators to offer them with telecom-grade quality using well-define quality of service (QoS) mechanisms. VoLTE is based on the multimedia telephony (MMTel) service, which is the standardized IMS-based (IP Multimedia System) VoIP service designed to replace existing circuit-switched voice. Rich Communications Services (RCS) marketed under joyn™ by GSMA, complements VoLTE and provides definition for:

- **Enhanced phonebook:** service capabilities and enhanced contacts information such as presence and service discovery
- **Enhanced messaging:** enables a large variety of messaging options including chat, emoticons, location share, and file sharing
- **Enriched calls:** enables multimedia content sharing during a voice call, video call, and video sharing

With VoLTE, operators can:

- Simultaneously deliver data with high-definition (HD) quality voice
- Create attractive communication services, blending mobile voice with video, converged IP messaging, the web, and social networking
- Harmonize today's fragmented communications
- Benefit from LTE's improved spectral efficiency
- Quickly re-farm 2G and 3G spectrum to LTE for capacity gains

VoLTE Deployments and Circuit Switched Fallback

After several years of debate and discussion over alternative technology proposals, VoLTE has emerged as the industry's preferred method of supporting new rich-media services. VoLTE has broad industry support from both the vendor and operator communities. Initial launches and trials started in 2012 and will continue in 2013, with increasing deployment and usage in 2014 and 2015.

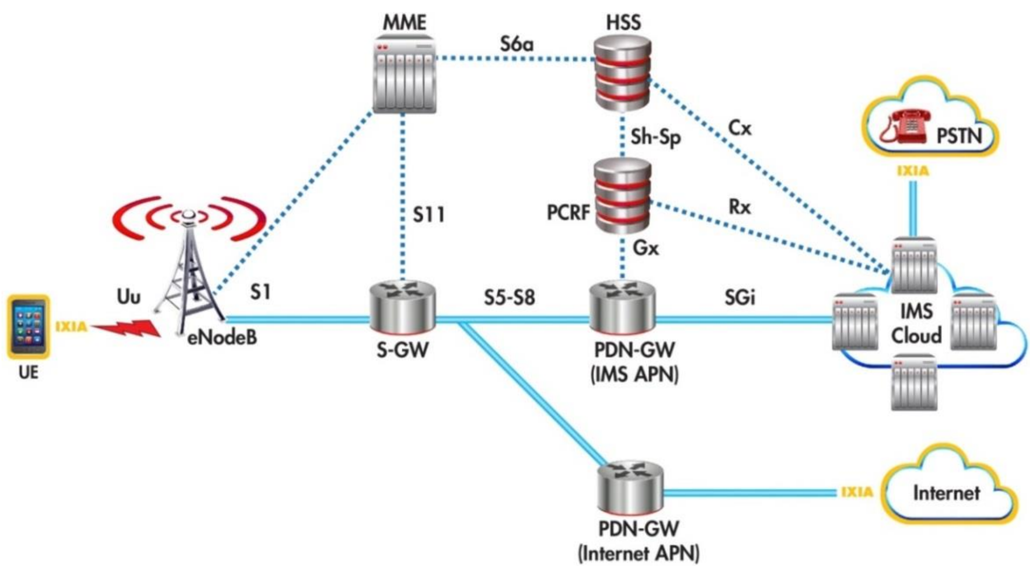
Initial LTE deployments will be data only. But LTE operators have an immediate need to supply voice and messaging services to their customers. Circuit switched fallback (CSFB) will be the method for interim voice services for LTE subscribers, until VoLTE is widely available. With CSFB an incoming or outgoing voice call forces a radio fallback from LTE to the legacy 2G or 3G service. Any 4G data is stopped at this point. This means that during voice calls the subscriber is limited to legacy circuit-switched (CS) voice and texting, and slower 3G mobile data services.

CSFB will primarily use narrowband codecs, and not the AMR-WB (Adaptive Multi-Rate Wideband) used for HD audio. Additionally, it will not enable all-IP services such as video calls. Another significant drawback of CSFB is call setup latency. CSFB call setup time is expected to be greater than 4 seconds (approximately 1 second more than 3G call setup), and will be noticeably longer than VoLTE's sub-second call setup time. In the field, times of over 10 seconds have been seen to connect a CSFB call.

Importance of QoS and Policy Control to VoLTE

For operators to provide better rich media services compared to OTT players, end-to-end QoS is an absolute requirement. On 3G networks, delays tend to be high and variable, packets arrive out of order, and some packets are lost or discarded. The QoS framework defined for 3G networks was not implemented in practice. This cannot be the case with VoLTE. For high-quality, real-time services the “best-effort” packet data services in 3G will not be sufficient. LTE networks must be capable of identifying and treating service flows with well know QoS characteristic for latency, jitter, packet loss, and error rates. The network must deliver guaranteed QoS end-to-end from the UE (User Equipment), or the client device, all the way to external TDM-based networks.

This is exactly what the specifications for LTE, the evolved packet core (EPC), and VoLTE have defined in great detail. Differentiated services, or the ability to treat various data services differently from voice and video services, are one of the fundamental tenets of LTE. VoLTE will be the first true test of this architectural design.



VoLTE Network Architecture

VoLTE Test Challenges and Requirements

Testing VoLTE's complex mix of voice, video, and messaging services in terms of functionality, quality, resiliency, and scalability is a significant challenge facing the industry. Some major categories of VoLTE testing include:

- Functional validation
- Capacity planning and network scalability
- Effectiveness of QoS and policy control mechanisms
- Voice and video quality
- System interoperability
- Network resiliency and error handling

VoLTE testers, including network engineering groups at operators or system test groups at equipment vendors, are often responsible for rolling out new services or recreating and debugging network problems. They understand that most issues occur under load, so in addition to basic feature testing, they must validate network infrastructure under load. It is far too expensive for mobile operators to learn about scaling problems after network deployment, or product release in the case of a vendor. Another common theme is the need for service validation with real application, voice, and video traffic. Here the focus should not be on protocol testing, but instead to closely model the services and subscriber mobility the live network will handle.

There are a number of reasons why operators are moving away from relying solely on equipment vendors to perform network validation. Most operator networks are now multi-vendor sourced, that every network is unique and operators want to test their specific configuration, services, and traffic mixes. It is common for vendors to state performance and capacity numbers for specific "best case" configurations. Whereas operators need to know the scalability and bottlenecks of their specific network designs and call profiles. The flexibility to quickly roll out new services or charging plans, without going back to the vendor every time the network changes, is another desirable position.

In many regards, VoLTE poses a significant challenge, driving operators to supplement the testing performed by vendors. For VoLTE to work effectively, many nodes in an LTE network must inter-work together; LTE Base Station (eNodeB), the Mobility Management Entity (MME), the Serving and Packet Data Gateways (S-GW/P-GW), Home Subscriber databases (HSS), Policy Server, Charging Systems, and IMS components. There is a complex sequence of actions and interaction between nodes to authenticate, route, bill, and provide QoS. There are many moving parts in a VoLTE network, and more often than not, networks are multi-vendor sourced. The testing focus should be on end-to-end service validation, not individual device (node) testing. Operators do not want to duplicate development testing; they want to measure the expected user experience on the network and the network's overall scalability. To do this effectively, testing must be performed with all the components that will be used in the live network.

Since a lab-based replica of the live network is not practical or cost-effective, operators rely on test systems that provide line-rate emulation of video, voice over IP (VoIP), VoLTE, data, and peer-to-peer protocols and application traffic.

Key VoLTE Test Capabilities

Rather than rely on vendor data sheets, mobile operators can use purpose-built test systems and strategic methodologies to subject devices and configurations to high stress, high scale conditions. In general, a VoLTE test system must support functional, performance, and stability testing of SIP-based VoIP network components as well as a wide mix of voice, video, and data applications.

The complexities and variables of wireless networks must be fully replicated and validated in the lab using an automated, repeatable, and proven approach. Operators must be able to assess the subscriber experience in the face of mobility, system overload, and even device failure on a large-city scale.

End-to-end test coverage encompasses everything from the wireless base stations to the Internet core, and operators need to be able to evaluate the entire LTE/VoLTE network as a whole, and also isolate individual subsystems such as the RAN, EPC, and the IMS core. Test tools must support SIP, SDP, H.323, MGCP, H.248, SKINNY, and RTP/RTCP protocols with voice codecs, video telephony, and fax, in addition to video and data/web protocols.

They must also be capable of testing a variety of network components in LTE and IMS topologies, including:

- SIP proxies and registrar servers
- Media gateways
- Call agents
- Session border controllers (SBCs) and application-layer gateways

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- Application servers
- EUTRAN components (eNodeB)
- EPC components (S-GW, P-GW, MME, HSS, PCRF)
- IMS components (x-CSCF)

Important traffic emulation and SIP test functionality includes the ability to:

- Emulate real-world application traffic at extremely high scale
- Simultaneously-support data, voice, and video protocols to emulate a multiplay subscriber environment
- Simulate SIP endpoints behind one or many SIP proxies
- Simulate SIP proxy and SIP registrar server
- Maintain full control over SIP state machines, messages, and contents
- Create any test case, including negative testing

VoLTE Test Configurations

To ensure a reliable VoLTE service, testing must occur under a variety of VoLTE test configurations. Operators will test end-to-end from the UE to the external networks, they will also need to isolate a sub-system, such as the EPC, to understand its scalability or to isolate a network problem. Generally, product development teams are more concerned with testing individual components like an eNodeB, whereas system test groups need to look at the network more holistically.

Common test configurations include:

1) **End-to-end system test.** In this scenario the test tool must emulate UEs with SIP agents over the Uu interface (RF), generating voice and SMS traffic into the LTE eNodeB. On the other side of the network under test, the tool emulates land-based SIP user agents or the PSTN.

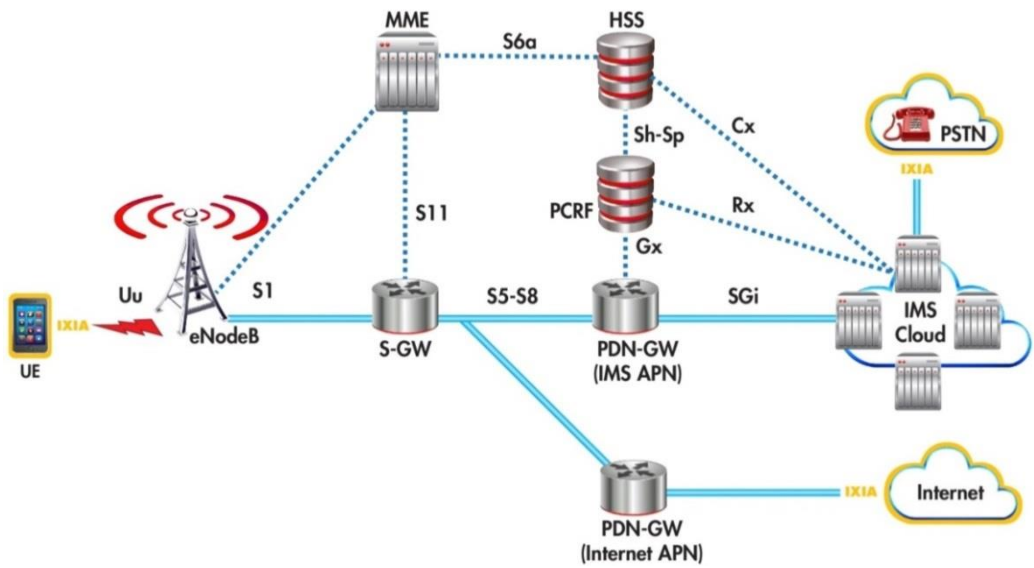
2) **Isolation of the evolved packet core (EPC).** In this configuration the test tool supports the emulation of millions of mobile subscribers and hundreds of eNodeB's generating voice and SMS traffic directly into the EPC. On the other side of the network the entire core network is emulated by replicating the behavior of the Proxy Call Session Control Function P-CSCF and all other devices behind it.

3) **Isolation of the combined EPC and IMS core.** In this scenario mobile subscribers and eNodeB's are emulated on one side and land-based SIP user agents are emulated on the other side.

The following VoLTE test cases can be run from a number of points within the LTE network. To ensure full functionality, it is important that full end-to-end (VoLTE client to VoLTE client) tests are performed. This will validate that all network elements are operating correctly across the VoLTE network. It is possible and preferable in certain cases to test under different configurations. These are considered below.

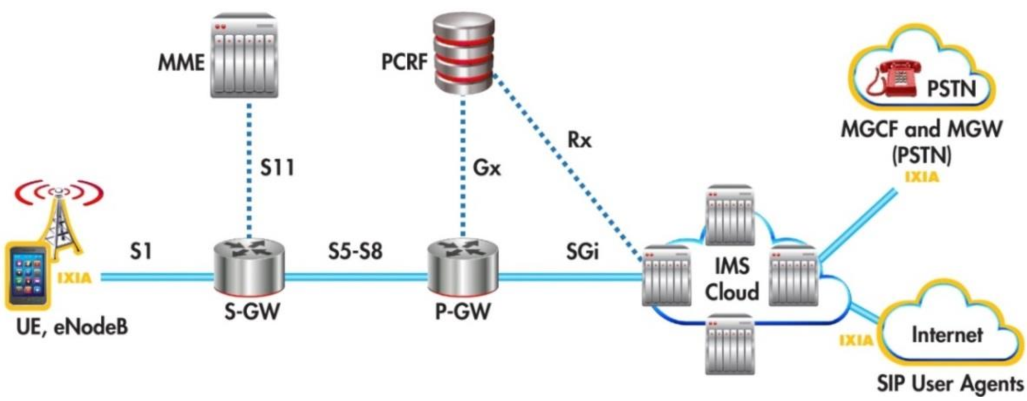
Test Configuration: VoLTE Client End to End Across the Network

This configuration represents the deployed network, with UEs connecting end to end across the network. Testing under this scenario involves Ixia emulation (indicated in diagrams using yellow highlights) of the LTE UEs and possibly SIP/PSTN clients or additional core servers e.g., web/video servers available through the Internet Access Point Name APN, all other elements are real. The Test Case section is written with this configuration.



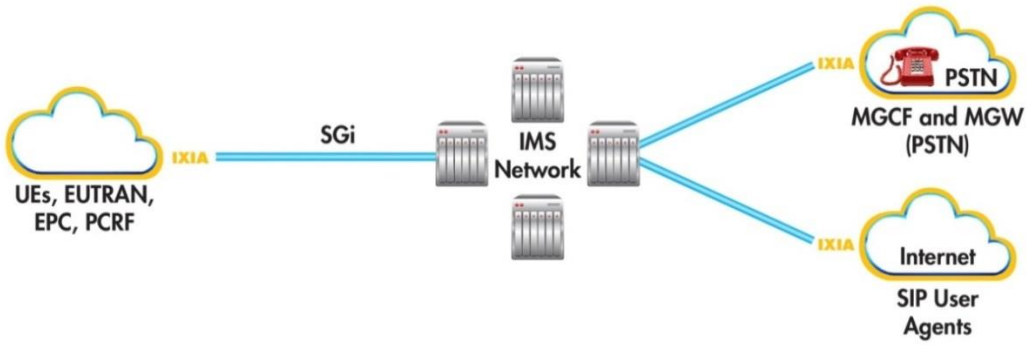
Test Configuration: UE/eNodeB Emulation Across the VoLTE Network

This configuration brings in the emulation of the eNodeB (eNodeB) with the UE. The eNodeB, with its air interface operation and scheduling challenges, can often be the key element of impairment. This test configuration allows direct validation that the rest of the wired network is operating correctly. This configuration also is useful for testing VoLTE load in the network, by allowing much greater loading of the EPC and IMS than is possible solely through UE emulation. Each eNodeB is limited to less than a Gbps of traffic, whereas the LTE EPC can handle tens to hundreds of gigabits per second (Gbps) of traffic, along with the IMS and MME servers that can handle thousands of signaling transactions per second.



Test Configuration: IMS Isolation

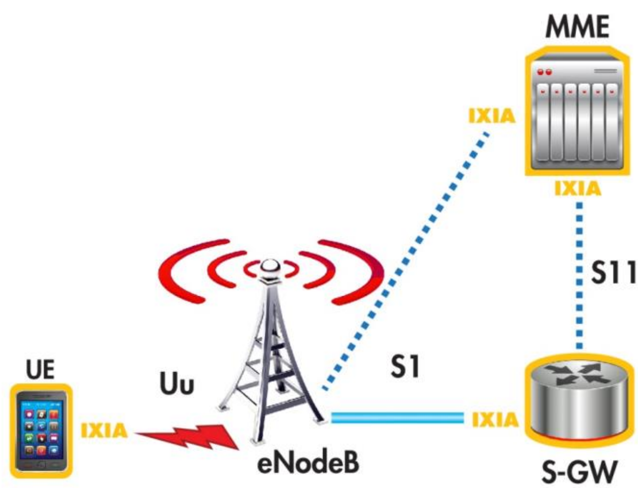
This configuration involves emulation of the PCRF directly to the IMS core. This will validate the signaling functionality and also can validate the expect IMS load capacity.



Test Configuration: eNodeB Isolation

While the eNodeB does not directly process the IMS (SIP) signaling used for VoLTE, it is the element most likely to introduce impairment of VoLTE operation from achieving high signal quality. Ensuring fast and effective eNodeB operation is essentially to VoLTE operation.

The eNodeB has strong challenges in scheduling the UEs for their Uplink (UL) and Downlink (DL) transmissions. By its nature, a VoLTE stream has small voice packets with small inter-arrival times. eNodeB functionality like Semi-Persistent Scheduling, TTI Bundling, and Robust Header Compression RoHC are all designed to assist VoLTE operation, but add complexity that must be examined. By testing an eNodeB in isolation, additional focus can be applied to validating this functionality.



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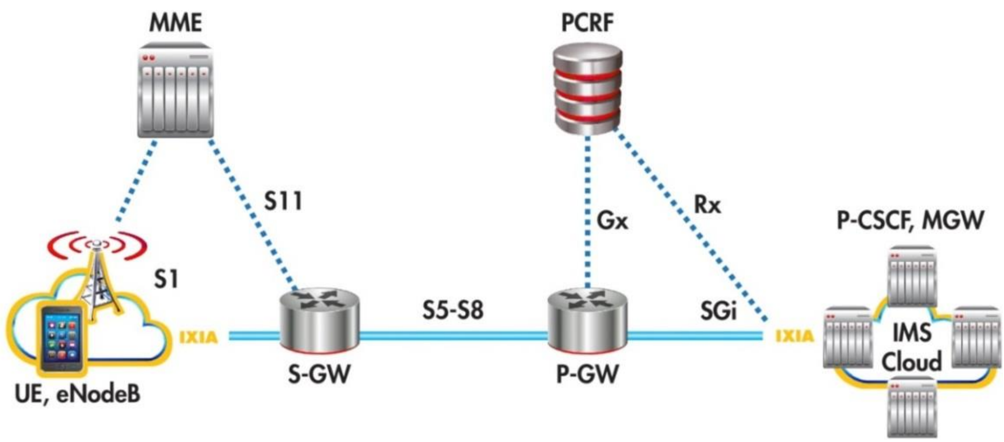
Test Configuration: EPC Isolation

Isolating the EPC with emulation of the UE/eNodeBs and IMS network allows focused investigation of issues that may be affecting VoLTE operation from within the EPC.



VoLTE Test Cases

This section proposes a set of VoLTE test cases that ensure correct operation across a VoLTE Network.

All the test cases proposed are taken from the perspective of a full end-to-end VoLTE test across the network from UE to UE. This is the first test configuration illustrated in the Test Configuration section. All other Test Configurations can be used to run these scenarios.

The test cases can be divided into the following sections:

- Section 1: Setup/Registration
- Section 2: Voice
- Section 3: SMS
- Section 4: Video
- Section 5: Advanced
- Section 6: Subscriber Modeling

Test Case Format

Each of the test cases presented will follow the following format.

Test Case Number: Title

Title describes the key scenario of the test (e.g. VoLTE voice-only call)

Description

- Scenario overview
- Diagram that shows the critical network elements and positive results for the test case
    - UE_1 is the default for result pass criteria if another UE is not identified

Initial State

- Numbered pretest configuration details
- May include earlier test scenarios denoted by number

Test Steps

- Numbered bullets of each test step in the call flow
- Messages can be shown in a different font with key parameters identified:

  `SIP REGISTER … FROM:<PhoneNumber>@ims… P:ASSERTED_ID Cell_ID`

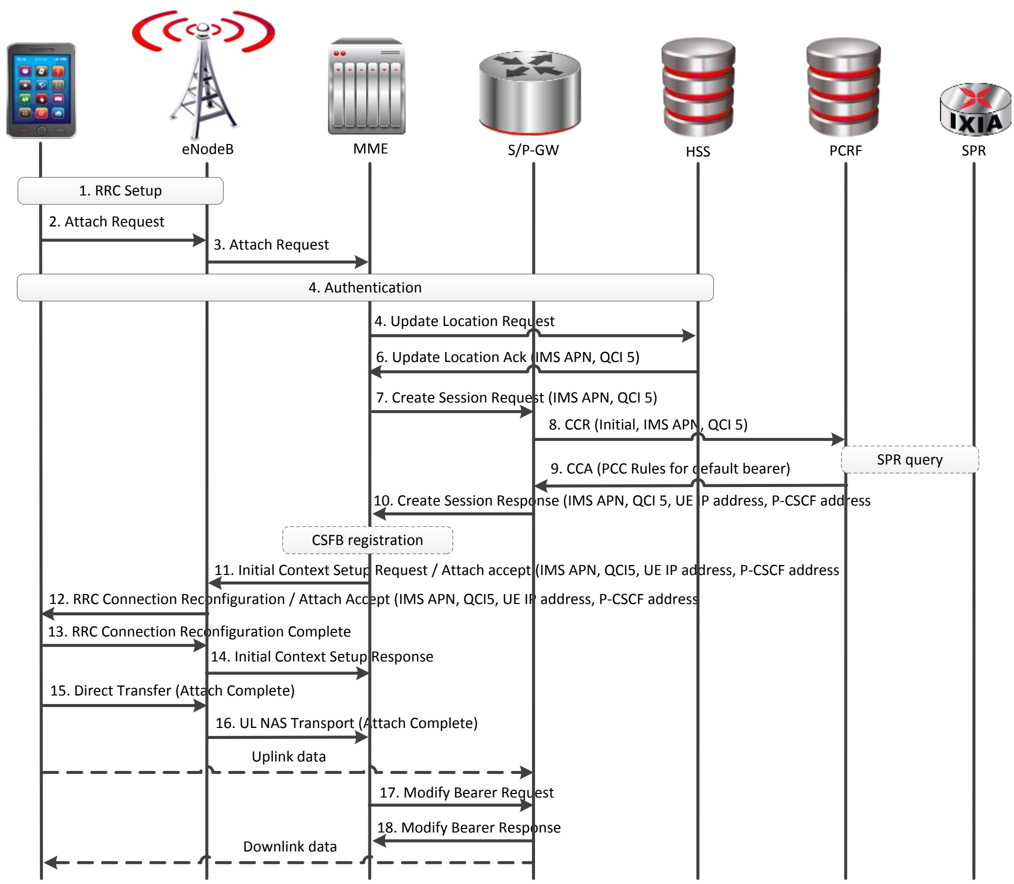
Results

- Numbered checks to be met to pass the test case

Section 1: VoLTE Setup

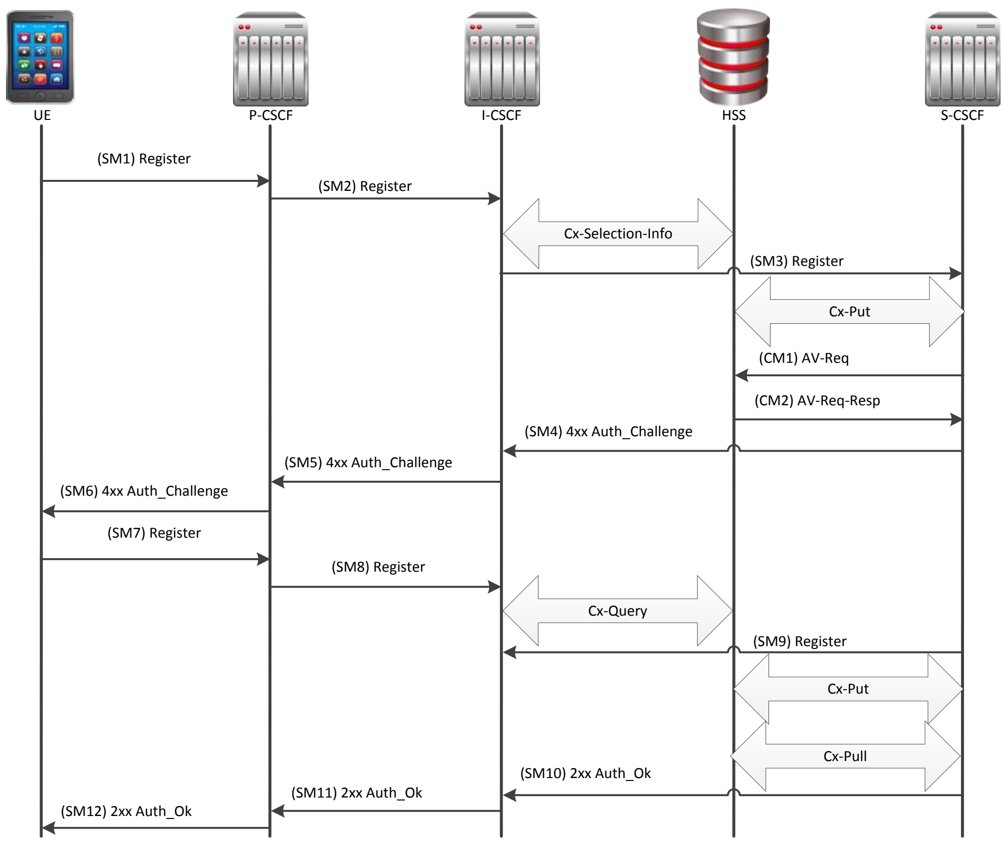
LTE Attach Call Flow

Before a VoLTE test scenario can be run, it is assumed that the UE has gone through the Attach process. This handles the underlying LTE-based signaling and configuration before the VoLTE (IMS) signaling is started. This call flow is provided below.



IMS Registration Call Flow

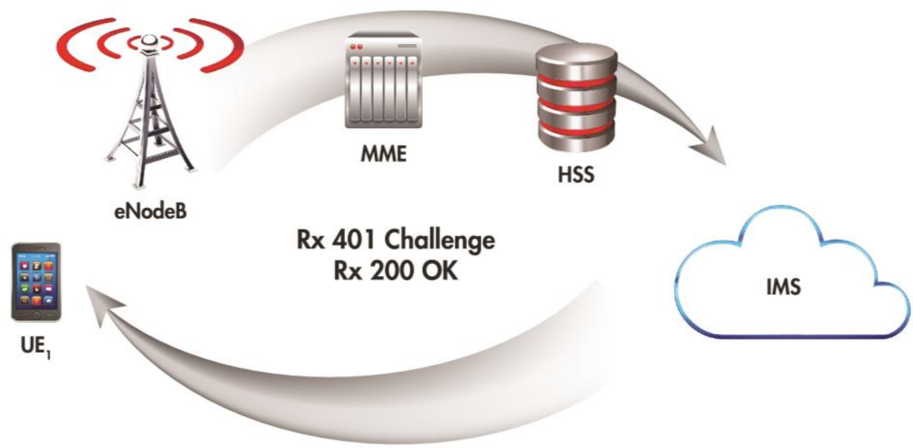
The UE must register with the IMS network. This allows its presence to be updated. This call flow looks like this:



Test Case 1: IMS Registration

Description

A UE will register with the IMS to allow VoLTE services using MD5 Digest Authentication.



Initial State

1) UE is EPS attached and has the default bearer activated for VoLTE APN

Test Steps

1) UE sends a REGISTER message to home network (to S-CSCF via P-CSCF and I-CSCF) to perform SIP registration with a public user identity:

   `REGISTER FROM:<PhoneNumber>@ims… P:ASSERTED_ID Cell_ID`

2) UE receives SIP “401 Unauthorized” response from the IMS core (P-CSCF):

   `401 Challenge Random challenge (RAND)`

3) UE sends a 2nd REGISTER with calculated response (RES) based on a shared secret and previously received RAND

4) UE receives a SIP 200 OK response from the S-CSCF routed back via all the CSCFs

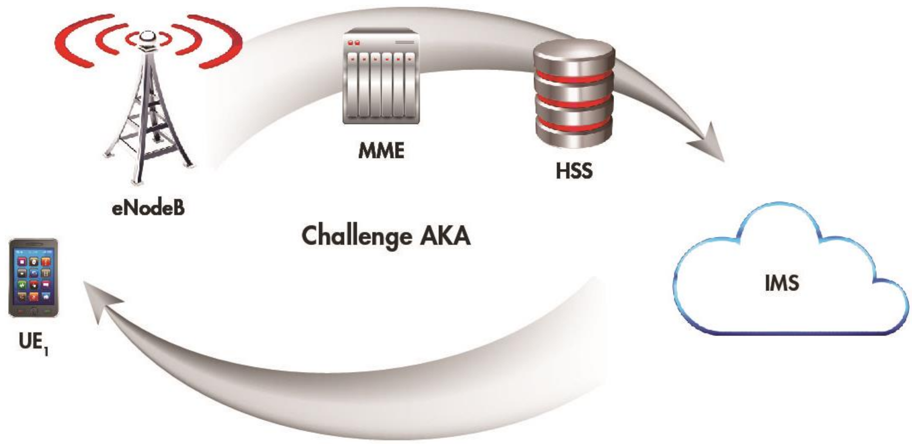
Results

1) UE successfully receives 401 with Random Challenge (RAND)
2) UE successfully receives SIP 200 OK

Test Case 2: IMS Registration with IMS AKA

Description

A UE will register with the IMS to allow VoLTE services using IMS AKA authentication.



Initial State

1) UE is EPS attached and has the default bearer activated for VoLTE APN

Test Steps

1) UE sends a REGISTER message to home network (to S-CSCF via P-CSCF and I-CSCF) to perform SIP registration with a public user identity:

   `REGISTER FROM:<PhoneNumber>@ims… P:ASSERTED_ID Cell_ID`

2) UE receives SIP "401 Unauthorized" response from the IMS core (P-CSCF):

   `401 Challenge`
   `Random challenge (RAND)  Network authentication token (AUTN) Authentication scheme (IMS authentication and key agreement AKA)`

3) UE sends a 2nd REGISTER with calculated response (RES) based on a shared secret and previously received RAND

4) UE receives a SIP 200 OK response from the S-CSCF routed back via all the CSCFs

Results

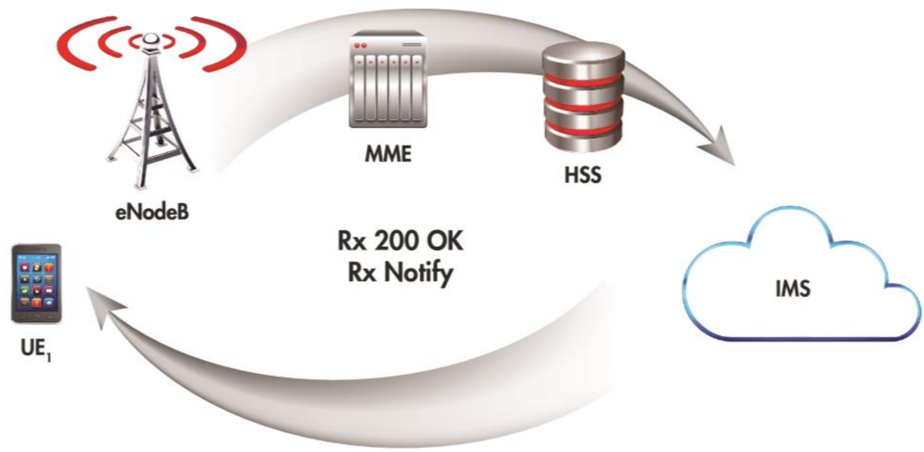
1) UE successfully receives 401 with Random Challenge (RAND) and AKA fields

2) UE successfully receives SIP 200 OK

Test Case 3: SIP Subscribe Procedure

Description

A UE Subscribes for an Event of Notification from the IMS network. Event notifications are used in services including automatic callback services (based on terminal state events), buddy lists (based on user presence events), message waiting indications (based on mailbox state change events), and PSTN and Internet Interworking status (based on call state events).



Initial State

1) UE is EPS attached and has the default bearer activated for VoLTE APN

2) UE has successfully Registered to the IMS network

Test Steps

1) UE initiates Subscribe indicating Event subscription:

   `SIP SUBSCRIBE (REG-EVENT) (Event: reg), new Dialog`

2) UE receives 200 OK indicating subscription has been accepted and containing Expires header field to indicate the actual duration for which the subscription will remain active (unless refreshed)

3) UE receives SIP NOTIFY(REG-EVENT), sends 200 OK

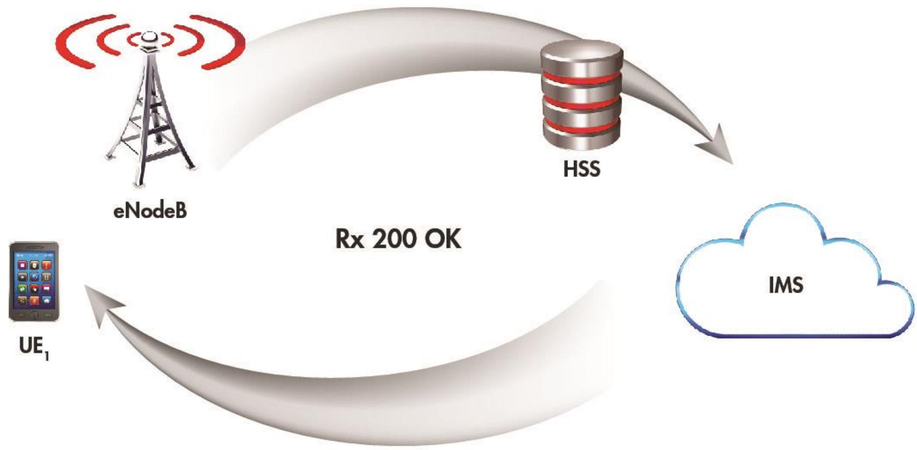
Results

1) UE receives 200 OK in response to SUBSCRIBE
2) UE receives SIP NOTIFY(REG-EVENT)

Test Case 4: SIP De-Registration on UE Power Down

Description

A Registered UE powers down and causes de-registration from the IMS network.



Initial State

1) UE is EPS attached and has the default bearer activated for VoLTE APN

2) UE is Registered with the IMS network and is in Idle or Connected state

Test Steps

1) UE is powered down causing a de-registration, by sending a REGISTER request to the S-CSCF with expiry time set to 0:

   ```
   REGISTER Expiry Time:0
   ```

2) The S-CSCF clears all temporary information it has for UE and updates the HSS

3) UE receives a “200 OK” response from the S-CSCF

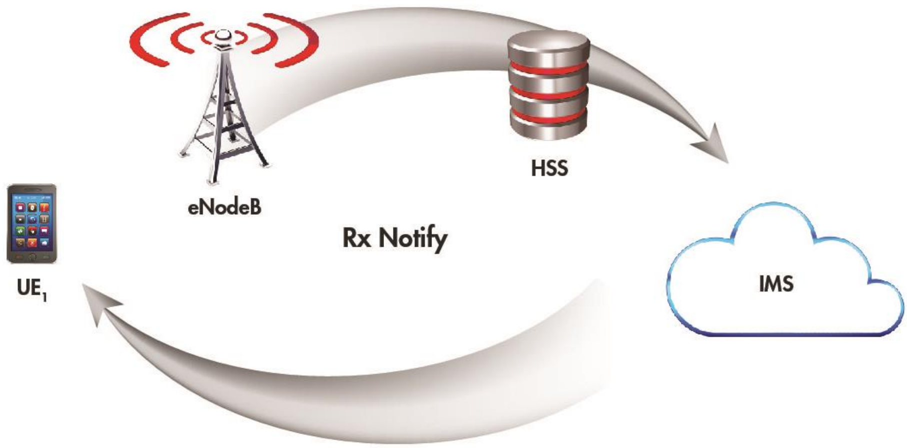
Results

1) UE successfully receives 200 OK to deregister

Test Case 5: SIP De-Registration on Network Release

Description

A Registered UE is released from the Network. This can happen for various reasons (reported stolen, unpaid bills, administrative reasons).



Initial State

1) UE is EPS attached and has the default bearer activated for VoLTE APN

2) UE is Registered with the IMS network and is in Idle or Connected state

Test Steps

1) S-CSCF sends a NOTIFY message with registration-state information to UE:

   `NOTIFY Deactivated Event`

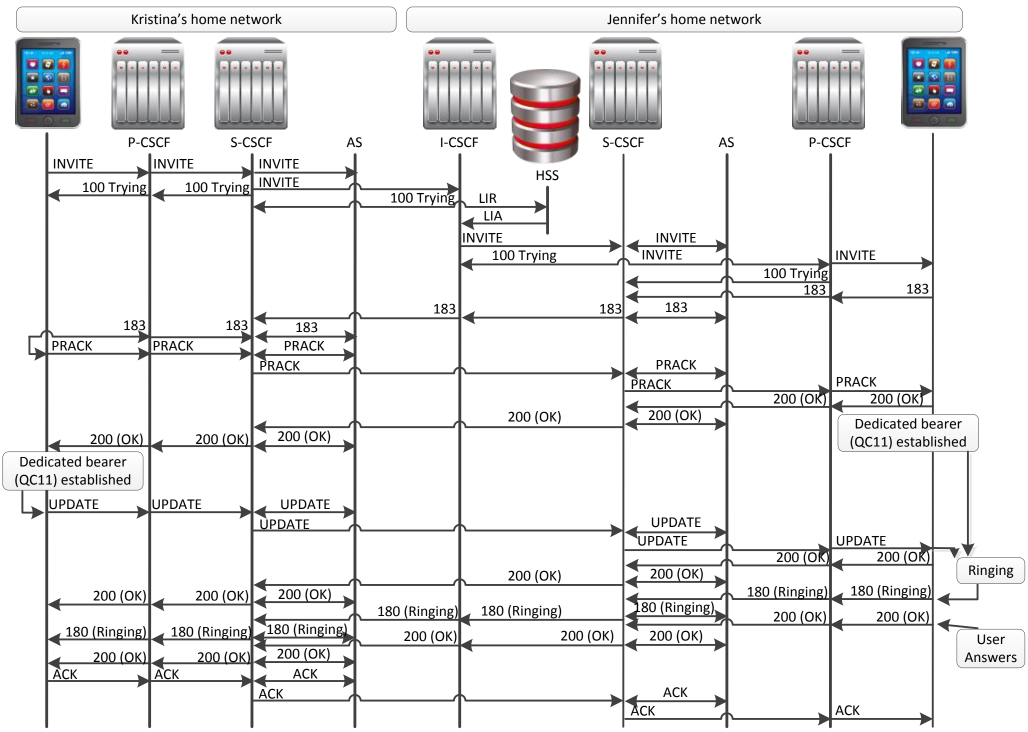
2) UE sends a “200 OK” back to S-CSCF to acknowledge de-registration

Results

1) UE Successfully receives Notify Deactivated Event

Section 2: VoLTE Voice-Only Call

This section provides test cases that focus on IR.92 Voice calls only. VoLTE through dedicated bearers provides the quality difference to VoIP over the top traffic.



Measuring Quality of Voice

Speech quality in a telephony system is a subjective judgment that depends on technical attributes of the system and the participants' speaking and listening preferences. A mean opinion score (MOS) provides a numerical indication of the quality of transmission, in the range of 1 to 5 (see table).

Mean Opinion Score

MOS	Quality
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

There are 2 types of MOS algorithms:

1) **Perceptual/Subjective (POLQA and PESQ)**

   a. **Perceptual Objective Listening Quality Analysis**
   Perceptual Objective Listening Quality Analysis (POLQA®) is the voice quality testing standard for fixed, mobile, and IP-based networks that was adopted as ITU-T Recommendation P.863 and successor to P.862/PESQ. POLQA provides strong support for testing of new wideband 4G/LTE networks delivering HD-quality voice services. POLQA is particularly important for measuring QoE of wideband HD voice mobile calls in the 7 kHz and 14 kHz frequency range.

   The POLQA perceptual measurement algorithm is the joint development of OPTICOM, SwissQual, and TNO.

   b. **Perceptual Evaluation of Speech Quality**
   Perceptual evaluation of speech quality (PESQ) is a mechanism that measures the quality of speech in the automated way as defined by ITU-T standard P.862. PESQ is an objective measurement method that predicts the results of subjective listening tests. The PESQ algorithm produces results analogs with the subjective MOS standard. A mapping between PESQ results and MOS was defined after the release of the P.862 recommendation. PESQ-LQ (PESQ- listening quality) is defined in ITU-T Rec. P.862.1, and improves the correlation with subjective test results at the high and low ends of the scale. It is important to measure both values: PESQ-LQ and PESQ-LE (listening effort).

PESQ is a full-reference method designed for end-to-end quality of voice assessment using a psycho-acoustic and cognitive model. PESQ analyzes the degraded audio signal (the signal after passing through the communication system) versus the reference (the signal injected in the system). The method requires access to actual audio information in both reference and degraded signals, performs level and time alignment to accommodate attenuations and delays, process the disturbances and finally applies the cognitive model. This is done using signal processing algorithms requiring considerable processing power.

The ITU-T P.800 specification defines methods for subjective determination of transmission quality. These methods use a large number of human subjects who listen to sentences read aloud by professional male and female speakers and transmitted over the telephony system. The listeners rate the quality of the individual audio signals, which are averaged into a MOS score.

2) Network Packet Transmission Based (E-Model and R-Factor)

In packet networks, quality of voice measurements can be performed by assessing the packets transmission using the E-Model and then generating the metric R-Factor. As defined by ITU-T G.107, R-Factor combines a number of values measuring the effect of various impairments; some of these are:

- The effect of coding/decoding – defined as constants for every codec
- Jitter, packet loss, and delay
- The effect of audio signal capture (mouth to microphone) and reproduction (speaker to ear), defined as a constant

The E-Model does not require reference signal information as it does not look at the actual audio content of the degraded signal. This method requires far less processing power than PESQ.

The R-Factor method predicts a user satisfaction on a scale from 0 to 100, where 100 is the highest satisfaction. A formula is defined for conversions between R-Factor and MOS. For example, a perfect transmission with the codec G.711 has an R-Factor of 94 and a MOS of 4.4.

R-Factor

R-Factor	User Satisfaction
91-100	Very satisfied
81-90	Satisfied
71-80	Some users dissatisfied
61-70	Many users dissatisfied
51-60	Nearly all users dissatisfied
0-50	Not recommended

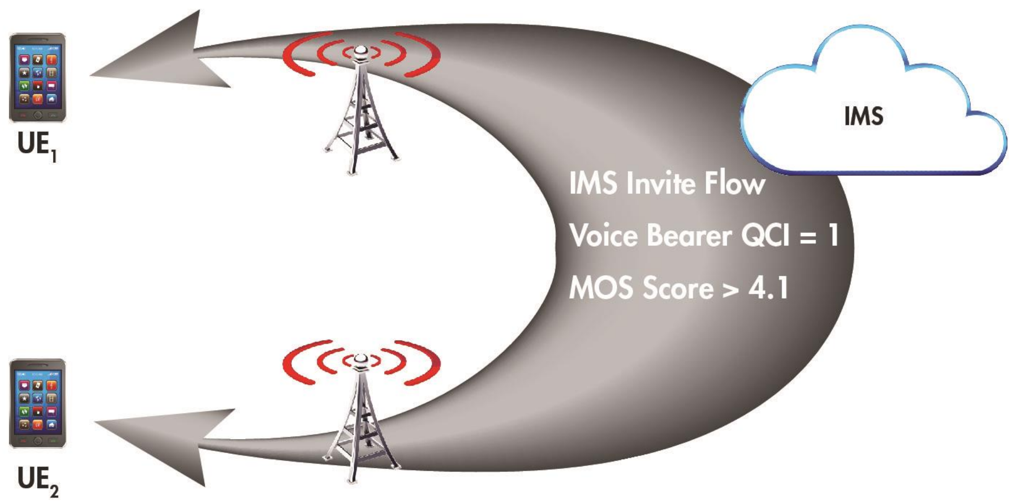
R-Factor and PESQ both characterize a voice transmission system; the former considers the packet networks impairments, while the latter compares the received audio signal with the expected signal. Beside the effect of the impairments of the transmission network, PESQ also captures the effects of trans-coding, voice activity detector, echo cancelation, and any other type of audio signal alteration.

Because the PESQ algorithm is computationally intensive, it is not practical to use it for testing the speech quality on high-scale devices or systems. However, if E-Model is used exclusively, some issues may remain hidden if the system performs audio signal processing. The best practice is to combine the two methods by performing E-Model measurement on all calls and PESQ on a smaller percentage of them.

Test Case 6: VoLTE Voice-Only Call with Wideband AMR

Description

This test case details the fundamental VoLTE scenario of a user-to-user call using Wideband AMR.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are subscribed with the IMS network and are available

Test Steps

1) UE_1 initiates the call by sending INVITE message to UE_2:

```
INVITE SDP
a=rtpmap
99 AMR-WB/16000
```

2) UE_2 responds with 183 Session Progress

3) UE_1 sends PRACK for reliability of the provisional response

4) UE_2 sends 180 Ringing

5) UE_2 responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the voice codec to be used

6) SIP call is established, RTP streaming starts; Voice Dedicated Bearer is configured at this stage

7) PCRF receives the 183 response and forwards it to UE_1, but in addition sends a Diameter AA Request (AAR) with session information from SDP (IPs, ports, media and codecs, bandwidth)

8) PCRF creates policy rules based on the session information from AAR and sends the Rx AA Answer (AAA) with a success value towards P-CSCF

9) PCRF also sends the created policy rule to P-GW using a diameter re-auth request (RAR); P-GW uses this information to enforce QoS and to apply traffic policy for voice media (PCEF)

10) P-GW also recognizes that is no bearer established for the provided QCI and ARP pair so it initiates a new dedicated bearer creation using this QoS information; for this P-GW sends Create Bearer Request to MME

11) MME allocates EPS bearer identity for this dedicated bearer and sends it together with EPS bearer QoS and TFT info to the UE in a Session Management Request inside the Bearer Setup Request to the eNodeB

12) eNodeB maps the EPS bearer QoS to the radio bearer QoS and sends a RRC Connection Reconfiguration message to the UE; this RRC Reconfig contains also a session management request sent by MME

13) UE stores the new bearer QoS settings and EPS bearer identity and uses the TFT to identify voice traffic flow coming from the application layer and matches uplink traffic to right radio bearer

14) After configuration, the UE returns the RRC Connection Reconfiguration Complete message to the eNodeB

15) eNodeB acknowledges the bearer activation to the MME with a Bearer Setup Response

16) MME acknowledges the bearer activation to the S/P-GW by sending a Create Bearer Response with Success outcome

17) Moving from the default bearer to dedicated if QCI matched and MBR received in Create Bearer Request is higher than configured value

18) Call is maintained for 3 minutes

19) UE_1 on hangup initiates a BYE

20) UE_2 responds with 200 OK

Results

1) Successful SIP Signaling execution to complete the INVITE process; ACK is received by UE_2

2) Successful initiation of the Voice stream, with correct properties, on a dedicated bearer (QCI=1)

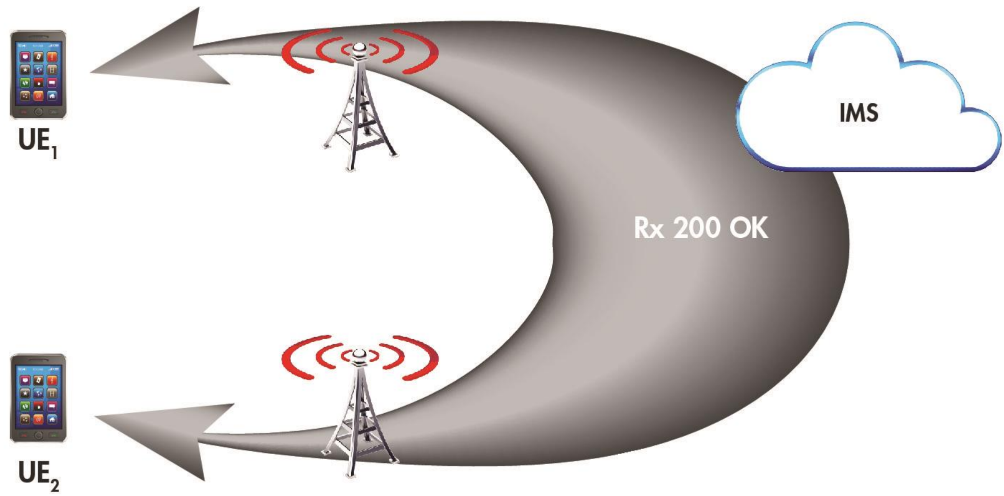
3) QoE assessment of MOS and PESQ for voice quality

    a) MOS should have a value > 4.1 (Excellent)

Test Case 7: VoLTE Call Release Initiated by the Calling Party

Description

In this scenario the VoLTE call release is initiated by the Calling party.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are subscribed with the IMS network and are involved in a bidirectional voice call for the last 3 minutes, with UE_1 being the call initiator

Test Steps

1) UE_1 sends a BYE that is propagated through IMS proxies to UE_2

2) UE_2 responds with 200 OK, which is propagated through the network to UE_1

Results

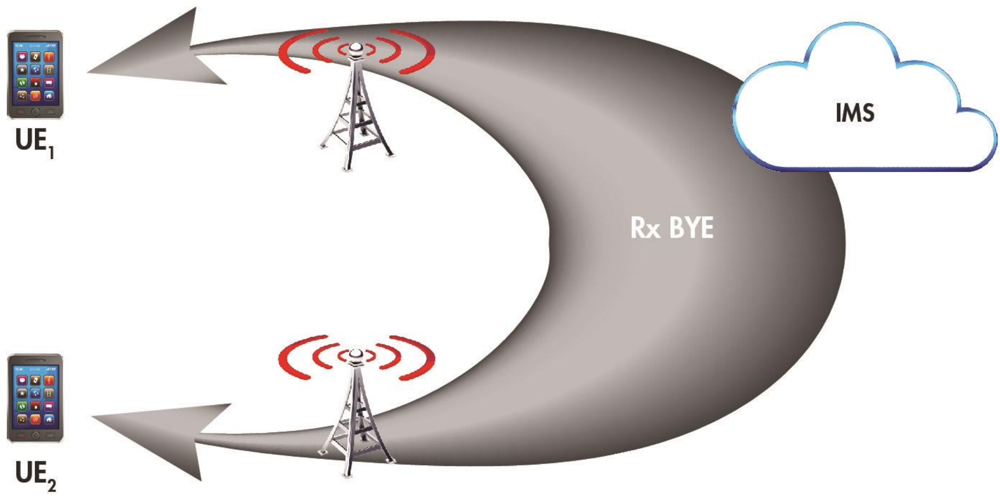
1) UE_2 receives BYE from his P-CSCF

2) UE_1 receives 200 OK from his P-CSCF

Test Case 8: VoLTE Call Release Initiated by the Called Party

Description

In this scenario the VoLTE call release is initiated by the Called party.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are subscribed with the IMS network and are involved in a bidirectional voice call for the last 3 minutes, with UE_1 being the call initiator

Test Steps

1) UE_2 sends a BYE that is propagated through IMS proxies to UE_1

2) UE_1 responds with 200 OK, which is propagated through the network to UE_1

Results

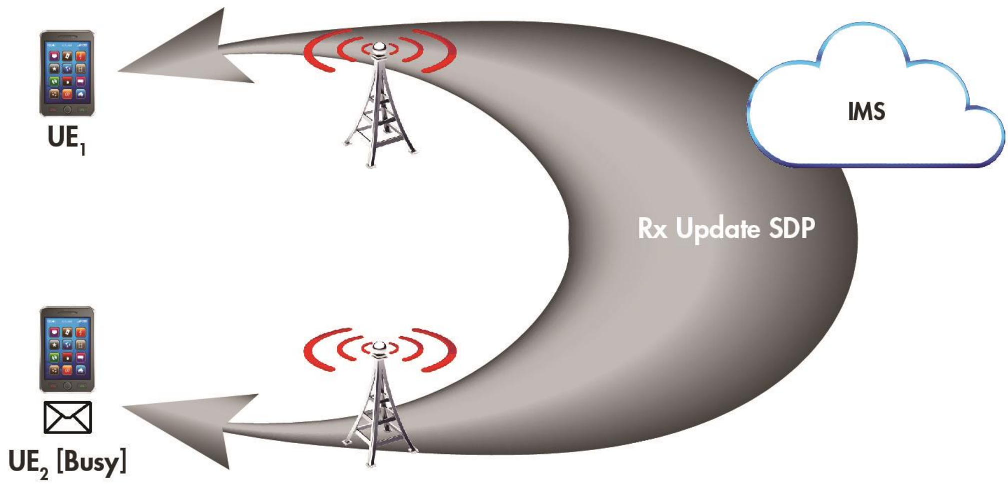
1) UE_1 receives BYE from P-CSCF

2) UE_2 receives 200 OK from P-CSCF

Test Case 9: Redirect VoLTE Call to Voice Mail after Called Party is Busy

Description

This test explores a case where the called party rejects incoming call (BUSY scenario) and the voice mail forwarding is triggered.



Initial State

1) UE$_1$ and UE$_2$ are VoLTE Registered and available

Test Steps

1) UE$_1$ initiates the call by sending INVITE message containing SDP with 2 way audio/video info towards UE$_2$ via IMS network (P,I,S-CSCF proxies)

2) UE$_2$ responds with 180 Ringing without SDP, which is propagated back to UE$_1$

3) UE$_2$ sends 486 Busy Here response

4) Upon receiving 486 Busy Here message S-CSCF proxy on UE$_2$ side will forward it to TAS (Telephony Application Services) module

5) TAS will trigger the Call Forwarding Procedure for UE$_2$

6) UE$_1$ receives UPDATE (with SDP offer) from S-CSCF

7) UE_1 responds with 200 OK UPDATE (with SDP answer) to S-CSCF

8) A voice prompt is played (RTP stream is sent from Voice Mail Server to UE_1); after the prompt, UE_1 can deposit his voice message to be delivered to UE_2 voice mail box (RTP stream is sent from UE_1 to Voice Mail Server)

9) After RTP streams for voice mail stop, UE_1 receives BYE from S-CSCF

Results

1) UE_2 sends 486 Busy Here

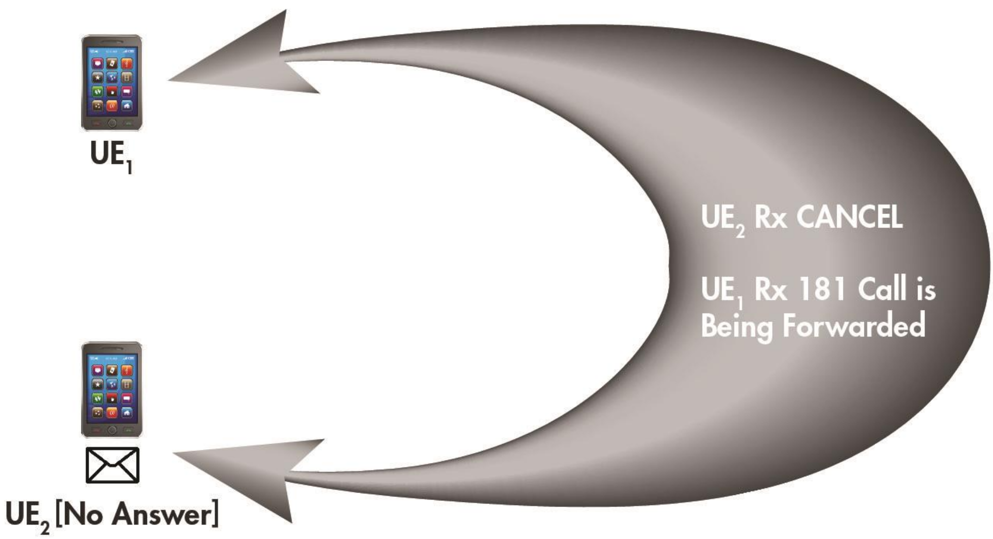
2) UE_1 receives UPDATE with SDP offer

3) UE_1 sends 200 OK UPDATE

Test Case 10: Redirect VoLTE Call Not Answered to Voice Mail

Description

This test explores a case where the called party doesn't answer incoming call and forwarding to voice mail is triggered.



Initial State

1) UE_1 and UE_2 are VoLTE Registered and available

Test Steps

1) UE_1 initiates the call by sending INVITE message containing SDP with 2-way audio (a=sendrecv) info towards UE_2 via IMS network (P,I,S-CSCF proxies)

2) UE_2 responds with 180 Ringing with SDP, which is propagated back to UE_1

3) UE_2 does not answer the incoming call, as a result Call Forward No Answer timer expires at TAS for UE_2

4) TAS sends CANCEL for the call to UE_2 through S-CSCF

5) UE_2 sends 200 OK to S-CSCF as response to CANCEL; S-CSCF forwards the 200 OK to TAS module

6) UE_2 sends 487 Request Terminated to S-CSCF; which forwards 487 to TAS module

7) TAS responds with an ACK, which is propagated through S-CSCF to UE_2

8) TAS triggers Call Forward procedure for Voice Mail

9) UE_1 receives 181 Call is Being Forwarded from S-CSCF

10) Voice Mail receives INVITE with SDP offer

11) Voice Mail responds with 180 Ringing and 200 OK (with SDP answer) to UE_1 via S-CSCF

12) UE_1 sends ACK

13) A voice prompt is played (RTP stream is sent from Voice Mail Server to UE_1); after the prompt, UE_1 can deposit his voice message to be delivered to UE_2 Voice Mail (RTP stream is sent from UE_1 to Voice Mail Server)

14) After RTP streams for voice mail are stopped, UE_1 receives BYE from S-CSCF

Results

1) UE_2 receives CANCEL

2) UE_2 sends 487 Request Terminated

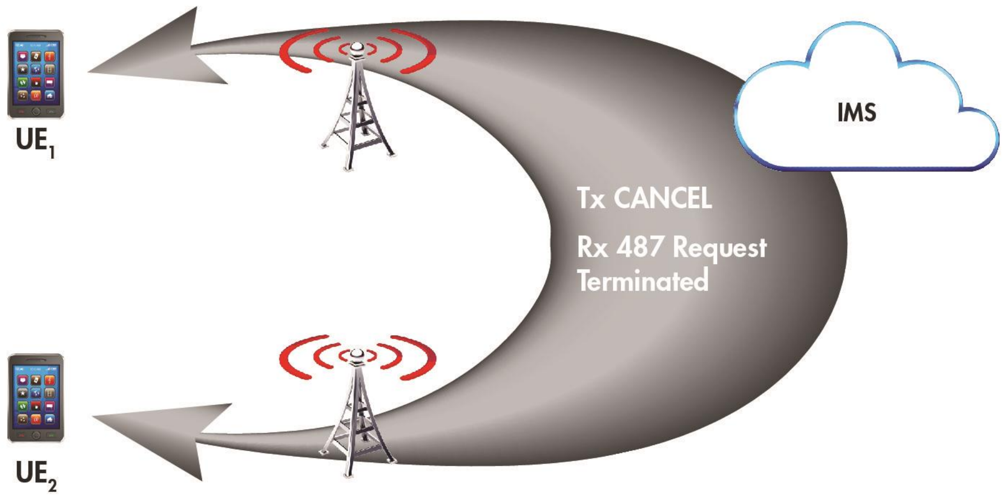
3) UE_1 receives 181 Call is Being Forwarded

4) UE_1 receives 200 OK

Test Case 11: Originator Cancels the Call Before Ringing

Description

This test case explores a VoLTE scenario where the originator changes his mind and cancels the call before receiving “180 Ringing.”



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are subscribed with the IMS network and are available

Test Steps

1) UE_1 initiates the call by sending INVITE message to UE_2

2) UE_2 responds with 100 Trying

3) UE_1 sends CANCEL

4) UE_2 responds with 200 OK

5) UE_2 sends 487 Request Terminated

6) UE_1 sends ACK

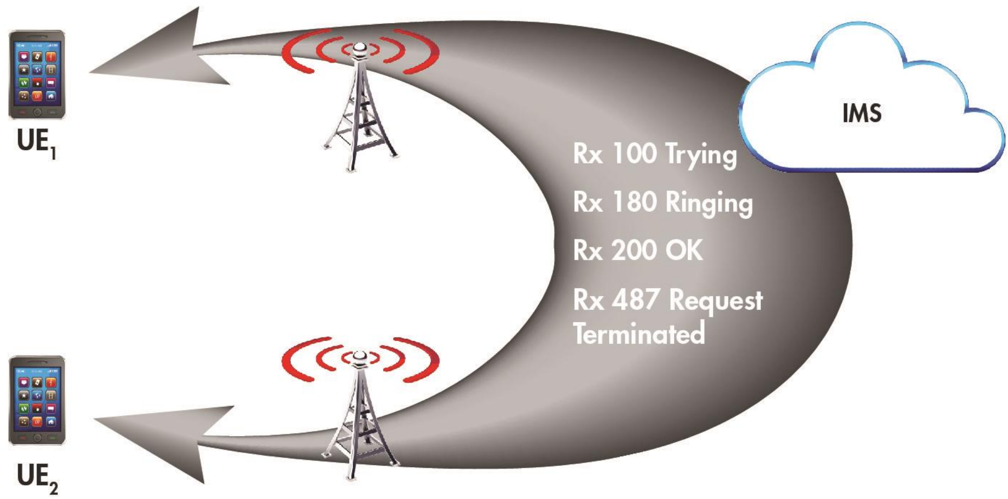
Results

1) UE_2 receives CANCEL
2) UE_1 receives 200 OK
3) UE_1 receives 487 Request Terminated

Test Case 12: Originator Cancels Call after Ringing

Description

This test case explores a VoLTE scenario where the originator changes his mind and cancels the call after receiving “180 Ringing.”



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are subscribed with the IMS network and are available

Test Steps

1) UE_1 initiates the call by sending INVITE message to UE_2

2) UE_2 responds with 100 Trying

3) UE_2 sends 180 Ringing

4) UE_1 sends CANCEL C_seq=1

5) UE_2 responds with 200 OK

6) UE_2 sends 487 Request Terminated

7) UE_1 sends ACK

Results

1) UE_1 receives 180 Ringing

2) UE_2 receives CANCEL

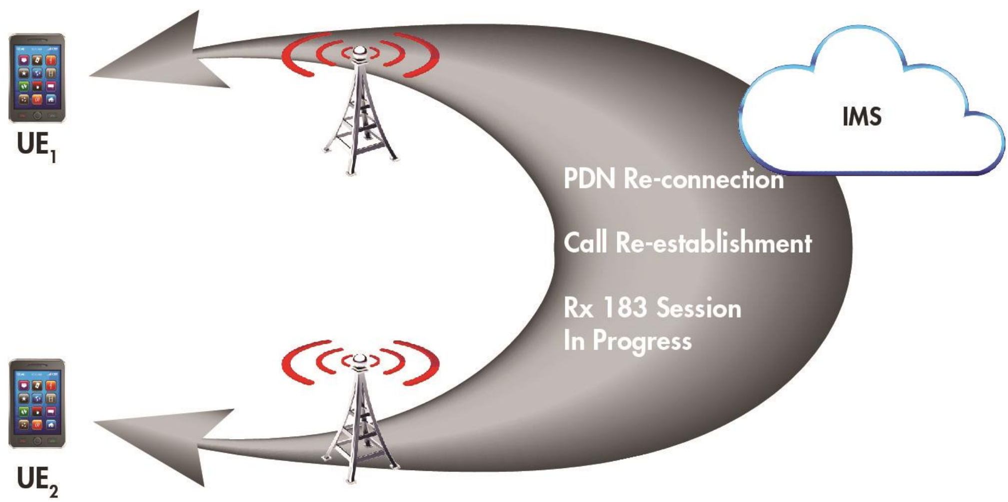
3) UE_1 receives 200 OK

4) UE_1 receives 487 Request Terminated

Test Case 13: Loss of PDN Connectivity

Description

In this Negative Test scenario, a call is re-established after PDN connectivity loss. This is an implicit detach by the network.



Initial State

1) UE_1 and UE_2 are VoLTE Registered

2) UE_1 has initiated an ongoing voice call with UE_2

3) RTP voice media packets are using dedicated bearer (QCI=1 and GBR)

Test Steps

1) UE_1 loses PDN connectivity

2) UE_2 receives BYE request with 503 "Service Unavailable" response code:

```
BYE: Reason header
503 Service Unavailable
```

3) UE_1 re-establishes PDN connection by sending NAS Attach request + PDN Connectivity request

4) After bearer configuration the UE returns the RRC Connection Reconfiguration Complete message to the eNodeB

5) UE_1 sends REGISTER and completes the IMS Registration procedure

6) UE_1 re-initiates the call by sending INVITE message to UE_2

7) UE_2 responds with 183 Session In Progress

8) UE_1 sends PRACK for reliability

9) UE_2 sends 180 Ringing

10) UE_2 responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the voice codec to be used in this call

11) PGW initiates a new dedicated bearer, UE_1 receives RRC Connection Reconfiguration

12) SIP call is established, RTP streaming starts. Voice Dedicated Bearer is configured at this stage

13) Call is maintained for 3 minutes

14) UE_1 on hangup initiates a BYE

15) UE_2 responds with 200 OK

Results

1) Successful PDN Re-Connection

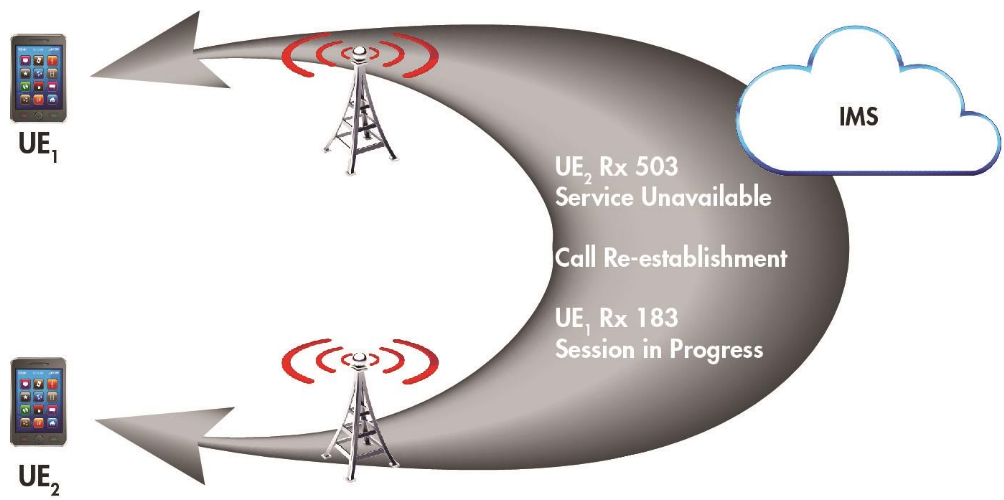
2) Successful Call Re-establishment

3) UE_1 receives 183 Session In Progress

Test Case 14: Loss of SIP Signaling

Description

In this Negative Test scenario, a call is re-established after a signaling bearer loss.



Initial State

1) UE_1 and UE_2 are VoLTE Registered

2) A voice call is established between UE_1 and UE_2 (UE_1 initiated) and is ongoing for 1 minute

3) RTP voice media packets are using dedicated bearer (QCI=1 and GBR)

Test Steps

1) UE1 loses SIP signaling

2) UE2 receives BYE request with 503 "Service Unavailable" response code

3) BYE 503 Service Unavailable

4) PGW/network initiates a new dedicated bearer for SIP signaling, UE1 receives RRC Connection Reconfiguration

5) After bearer configuration the UE returns the RRC Connection Reconfiguration Complete message to the eNodeB

6) UE1 sends REGISTER and completes the IMS Registration procedure. SIP signaling uses dedicated bearer (QCI=5)

7) UE1 re-initiates the call by sending INVITE message to UE2

8) UE2 responds with 183 Session Progress

9) UE1 sends PRACK for reliability

10) UE2 sends 180 Ringing

11) UE2 responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the voice codec to be used in this call; SIP call is established, RTP streaming starts

12) Call is maintained for 3 minutes

13) UE1 on hangup initiates a BYE

14) UE2 responds with 200 OK

Results

1) UE_2 Receives 503 Service Unavailable

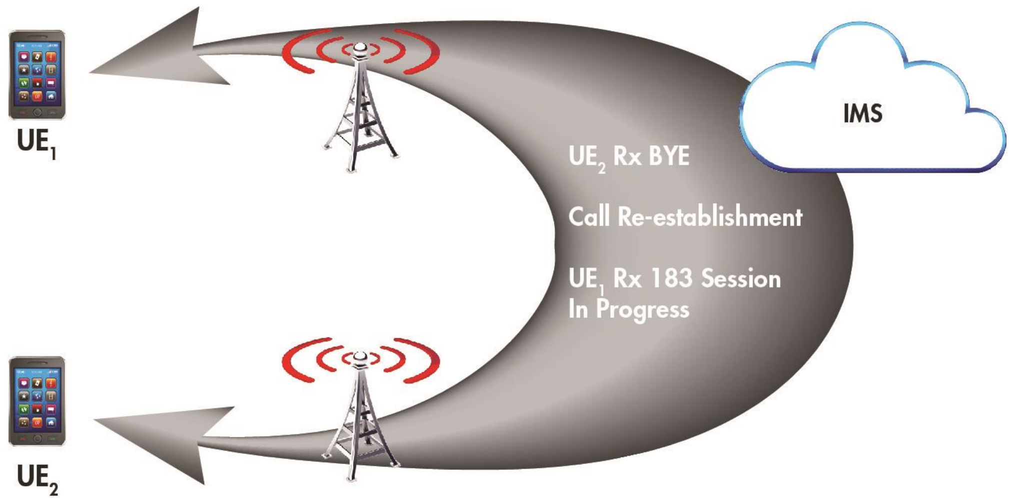
2) Call Re-establishment

3) UE_1 receives 183 Session In Progress

Test Case 15: Loss of Media Bearer

Description

In this Negative Test scenario, a call is re-established after loss of the Media Bearer.



Initial State

1) UE_1 and UE_2 are VoLTE Registered

2) A voice call is established between UE_1 and UE_2 (UE_1 initiated) and is ongoing for 1 minute

3) RTP voice media packets are using dedicated bearer (QCI=1 and GBR)

Test Steps

1) UE_1 loses the media bearer, this could be the result of a radio connection loss

2) UE_2 receives BYE request with 503 "Service Unavailable" response code:

```
BYE 503 Service Unavailable
```

3) UE_1 regains radio connectivity

4) UE_1 re-establishes PDN connection by sending NAS Attach request + PDN Connectivity request

5) After bearer configuration UE_1 returns the RRC Connection Reconfiguration Complete message to the eNodeB

6) UE_1 sends REGISTER and completes the IMS Registration procedure

7) UE_1 re-initiates the call by sending INVITE message to UE_2

8) UE_2 responds with 183 Session Progress

9) UE_1 sends PRACK for reliability

10) UE_2 sends 180 Ringing

11) UE_2 responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the voice codec to be used in this call

12) PGW initiates a new dedicated bearer. UE_1 receives RRC Connection Reconfiguration with Activate dedicated EPS bearer context

13) SIP call is established, RTP streaming starts

14) Call is maintained for 3 minutes

15) UE_1 on hangup initiates a BYE

16) UE_2 responds with 200 OK

Results

1) UE_2 Receives BYE

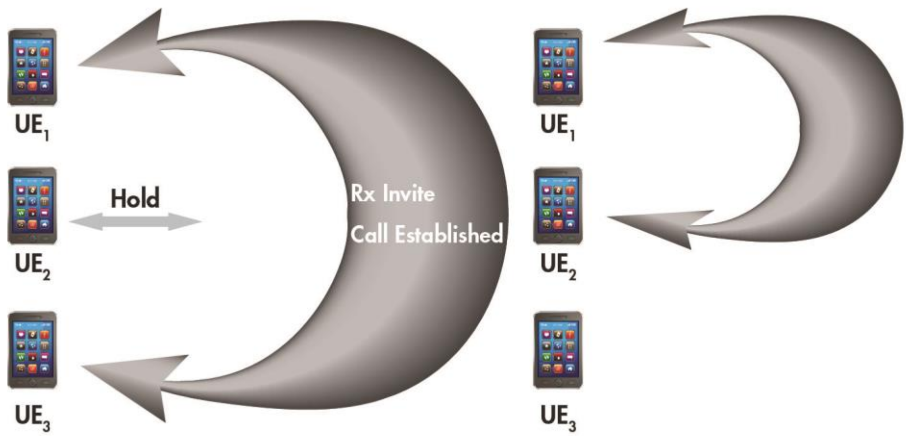
2) Call Re-establishment

3) UE_1 Receives 183 Session In Progress

Test Case 16: Voice Call Waiting, Second-Party Hold

Description

This scenario describes a user in a call who accepts an incoming call putting the current user on hold.



Initial State

1) UE_1, UE_2, and UE_3 are VoLTE Registered

2) A voice call is established between UE_1 and UE_2 and is ongoing for 1 minute

3) RTP media packets are using dedicated bearer (QCI=1 and GBR)

Test Steps

1) UE_3 initiates a voice call to UE_1 and sends INVITE with SDP Offer:

```
INVITE SDP
audio=AMR-WB, AMR
```

2) A “Call-Waiting” screen indication is presented on UE_1, which sends 180 Ringing to UE_3

3) UE_3 responds with PRACK for reliability

4) UE_1 responds with 200 OK

5) UE_1 accepts the voice call from UE_3 and sends Re-INVITE with SDP Offer to UE_2, which is put on “Hold”

```
Re-INVITE: SDP
audio=sendonly
```

6) UE_2 sends 200 OK with SDP Answer to UE_1

```
200 OK: SDP Answer: SDP
audio=recvonly
```

7) UE_1 responds with ACK

8) Audio streams between UE_1 and UE_2 are put on Hold

9) UE_1 sends 200 OK with SDP Answer to UE_3:

```
200 OK: SDP Answer
audio=AMR-WB
```

10) At this point RTP voice streams are exchanged between UE_1 and UE_3 on dedicated bearer (QCI=1 and GBR), establishing an ongoing Voice call

11) UE_1 hangs-up voice call with UE_3 and sends BYE to UE_3

12) UE_3 responds with 200 OK

13) UE_1 switches back to the voice call with UE_2 by sending UE_2 RE-INVITE with SDP offer to UE_2:

```
RE-INVITE: SDP Offer
audio=sendrecv
```

14) UE_2 responds with 200 OK containing SDP Answer:

```
200OK: SDP Answer
audio=sendrecv
```

15) At this point RTP voice streams are exchanged again between UE_1 and UE_2 on a dedicated bearer (QCI=1 and GBR). The call takes 60 seconds

16) UE_1 on hangup initiates a BYE

17) UE_2 responds with 200 OK, which reaches UE_1

Results

1) UE_1 receives INVITE from UE_3

2) UE_2 receives Re-INVITE with SDP:

   `audio=sendonly`

3) UE_1 receives 200 OK with SDP:

   `audio=recvonly`

4) UE_3 receives 200 OK with SDP:

   `audio=AMR-WB`

5) UE_3 receives BYE

6) UE_1 receives 200 OK

7) UE_2 receives Re-INVITE with SDP:

   `audio=sendrecv`

8) UE_1 receives 200 OK with SDP:

   `audio=sendrecv`

9) UE_2 receives BYE

Test Case 17: Voice Call Switch Hold

Description

A user puts existing call on hold for incoming call. The user then switches hold back to the original call and resumes.

Initial State

1) UE_1, UE_2 and UE_3 are VoLTE Registered

2) A voice call is established between UE_1 and UE_2 and is ongoing for 1 minute; RTP voice packets are sent in both ways

Test Steps

1) UE_1 places the call on hold by sending INVITE to UE_2 with SDP media attributes:

```
INVITE SDP
audio=AMR-WB, AMR
m=audio
a=sendonly
```

2) UE_2 responds with 200 OK with SDP media attributes:

```
m=audio
a=recvonly
```

3) UE_1 sends INVITE to UE_3

4) UE_3 responds with 200 OK

5) RTP voice packets are sent both ways between UE_1 and UE_3 on dedicated bearer (QCI=1, GBR)

6) UE_1 puts the call with UE_3 on hold and switches back to UE_2 by sending INVITE to UE_3 with SDP media attributes:

```
audio=AMR-WB, AMR
m=audio
a=sendonly
```

and sending INVITE to UE2 with new SDP media attributes:

```
audio=AMR-WB, AMR
m=audio
a=sendrecv
```

7) UE_2 responds with 200 OK with SDP media attributes:

```
audio=AMR-WB, AMR
m=audio
a=sendrecv
```

8) UE_3 responds with 200 OK with SDP media attributes:

```
audio=AMR-WB, AMR
m=audio
a=recvonly
```

9) RTP voice packets are sent both ways between UE_1 and UE_2 on dedicated bearer (QCI=1, GBR)

10) UE_1 ends the call by sending BYE to UE_1

11) UE_2 responds with 200 OK

12) UE_1 sends INVITE to UE_3 with SDP media attributes

13) UE_3 responds with 200 OK with SDP media attributes

14) RTP voice packets are sent both ways between UE_1 and UE_3 on dedicated bearer (QCI=1, GBR)

15) UE_1 ends the call by sending BYE to UE_3

16) UE_3 responds with 200 OK

Validating VoLTE

Results

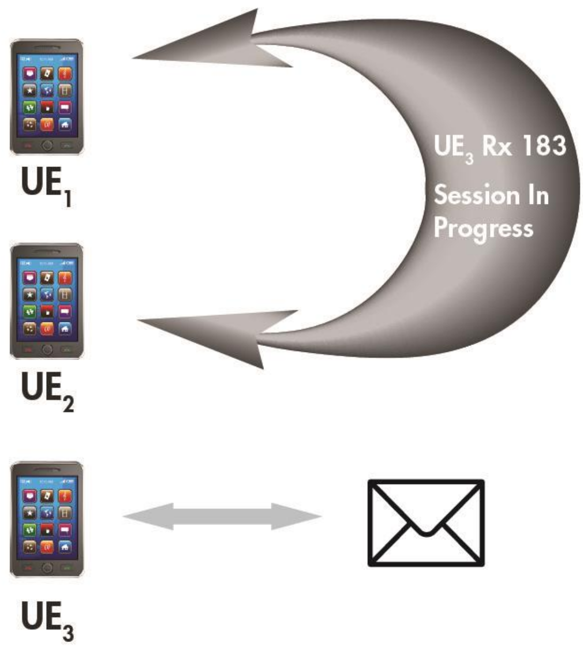
1) UE_1 receives INVITE with SDP media attributes

2) UE_3 receives INVITE with SDP media attributes

Test Case 18: Voice Third Call Redirect to Voice Mail

Description

The User in a call allows an incoming call to go to Voice Mail.



Initial State

1) UE_1, UE_2 and UE_3 are VoLTE Registered

2) A voice call is established between UE_1 and UE_2 and is ongoing

3) RTP voice media packets are using dedicated bearer (QCI=1 and GBR)

Test Steps

1) UE_3 initiates a voice call to UE_1 and sends INVITE with SDP Offer:

```
INVITE SDP
audio=AMR-WB, AMR
```

2) A "Call-Waiting" screen indication is presented on UE_1

3) UE_1 sends 180 Ringing to UE_3

4) UE_1 selects “Ignore” action

5) UE_1 sends 486 Busy Here, which propagates to TAS; TAS will initiate INVITE, which is routed through S-CSCF to voice mail system

6) UE_3 receives 183 Session Progress with SDP Answer:

```
183 Session Progress SDP
audio=AMR-WB, AMR
```

7) UE_3 responds with PRACK

8) A voice prompt is played to UE_3 indicating to record voice message after the signal

9) At this stage RTP voice packets are sent both ways between UE_3 and voice mail system. RTP voice media packets are using dedicated bearer (QCI=1 and GBR)

10) On hangup UE_2 receives BYE and UE_1 receives 200 OK

Results

1) UE_1 receives INVITE with SDP Offer:

```
audio=AMR-WB, AMR
```

2) UE_1 sends 486 Busy Here

3) UE_3 receives 183 Session In Progress with SDP Answer:

```
audio=AMR-WB, AMR
```

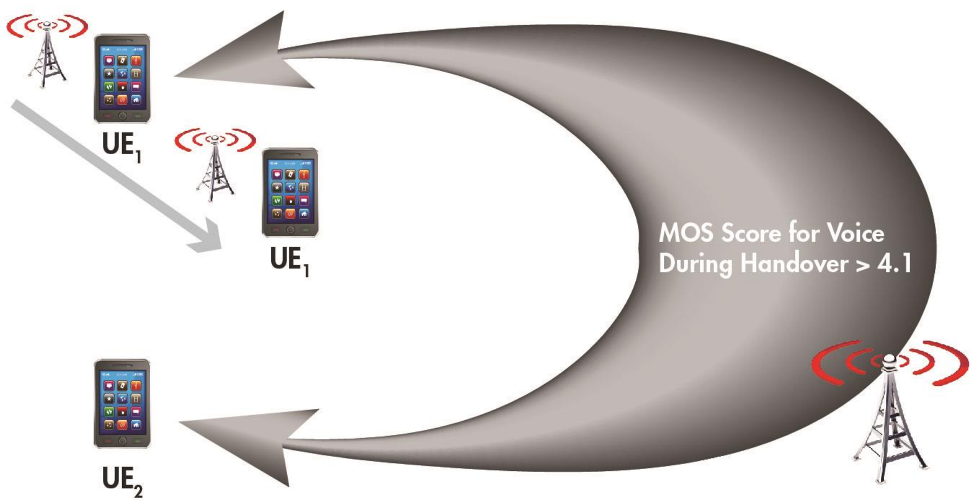
4) RTP packets are sent using Dedicated Bearers (QCI=1,GBR)

Test Case 19: Handover During VoLTE Voice-Only Call

Description

This test case explores user VoLTE quality under a handover situation.

This handover may be Intra-eNodeB, Inter-eNodeB handover across frequencies, bands or duplexing (FDD/TDD). This may invoke an X2, S11 or S1 with MME/S-GW reallocation.



Initial State

1) UE_1 and UE_2 are VoLTE Registered and available

Test Steps

1) UE_1 calls UE_2 successfully establishing SIP call and RTP packets are sent and received on Dedicated Bearer (QCI=1)

2) After 90 seconds UE_1 provides RRC Measurement Reports that detail better reception of Sector C than Sector A

3) The eNodeB then directs UE_1 to perform a handover across these sectors by sending RRC Connection Reconfiguration (Handover Command)

4) UE_1 performs a RACH procedure on the target sector/eNodeB and should send a RRC Connection Reconfiguration Complete message to the target if the procedure was successful

5) A DRB is established successfully between UE_1 and target eNodeB

6) Voice call continues for 30-60 seconds more

7) UE_1 ends the call by sending BYE; UE_2 responds with 200 OK

Results

1) UE_1 receives RRC Connection Reconfiguration (Handover Command)

2) Handover signaling is successful

3) New Radio Bearer to the target Sector is setup successfully

4) RTP packets are using Dedicated Bearers (QCI=1)

5) QoE assessment of MOS and PESQ for voice quality

    a) MOS is expected to have a value > 4.1

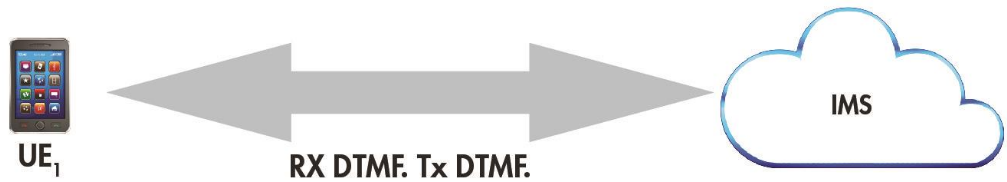
6) Voice stream Packet Loss < 0.001

7) Voice stream Jitter unaffected

Test Case 20: DTMF Tone Handling

Description

Here the user accepts and responds with DTMF during a VoLTE call.



Initial State

1) UE_1 is VoLTE Registered

Test Steps

1) UE_1 initiates a call by sending INVITE message to a SIP client (DTMF host)

   SDP contains narrowband and wideband codec support for both speech and DTMF (telephone-event):

```
INVITE SDP
m=audio 49152 RTP/AVPF 97 98 99 100 101 102
a=rtpmap:97 AMR-WB/16000/1
a=fmtp:97 mode-change-capability=2; max-red=220
a=rtpmap:98 AMR-WB/16000/1
a=fmtp:98 mode-change-capability=2; max-red=220; octet-
align=1
a=rtpmap:99 telephone-event/16000/1
a=fmtp:99 0-15
a=rtpmap:100 AMR/8000/1
a=fmtp:100 mode-change-capability=2; max-red=220
a=rtpmap:101 AMR/8000/1
a=fmtp:101 mode-change-capability=2; max-red=220; octet-
align=1
a=rtpmap:102 telephone-event/8000/1
a=fmtp:102 0-15
a=ptime:20
a=maxptime:240
a=sendrecv
```

2) UE_2 responds with 183 Session Progress

3) UE$_1$ sends PRACK for reliability

4) UE$_2$ sends 180 Ringing

5) UE$_2$ responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the codec to be used in this call:

```
200 OK SDP
m=audio 49152 RTP/AVPF 97 98 99 100 101 102
a=rtpmap:97 AMR-WB/16000/1
a=fmtp:97 mode-change-capability=2; max-red=220
a=rtpmap:98 AMR-WB/16000/1
a=fmtp:98 mode-change-capability=2; max-red=220; octet-
align=1
a=rtpmap:99 telephone-event/16000/1
a=fmtp:99 0-15
a=rtpmap:100 AMR/8000/1
a=fmtp:100 mode-change-capability=2; max-red=220
a=rtpmap:101 AMR/8000/1
a=fmtp:101 mode-change-capability=2; max-red=220; octet-
align=1
a=rtpmap:102 telephone-event/8000/1
a=fmtp:102 0-15
a=ptime:20
a=maxptime:240
a=sendrecv
```

6) SIP call is established, RTP streaming starts

7) PGW initiates new Dedicated Bearers (QCI=1, GBR). UE$_1$ and UE$_2$ receive RRC Connection Reconfiguration with Activate dedicated EPS bearer context

8) Call is maintained for as long as 5 digits are sent as DTMF tones (bidirectional)

   a) The tone duration of a DTMF event shall be at least 65ms and the pause duration in-between two DTMF events shall be at least 65ms

   b) RTP media packets are using dedicated bearer (QCI=1, GBR)

9) UE$_1$ on hangup initiates a BYE

10) UE$_2$ responds with 200 OK

11) UE$_1$ receives 200 OK

Results

1) DTMF are correctly decoded at UE_1

2) DTMF are correctly sent by UE_1

Test Case 21: VoLTE Emergency Registration

Description

This scenario describes a user performing an Emergency Registration in preparation for an Emergency call.

Initial State

1) UE_1 is already attached and creates an Emergency PDN or UE is not attached and it needs to perform an Emergency Attach to LTE network

2) UE_1 received the DNS address and the dedicated P-CSCF IP addresses from P-GW

Test Steps

1) UE_1 sends a REGISTER to P-CSCF, which forwards it through IMS network to S-CSCF

2) UE_1 receives 401 UNAUTHORIZED challenge:

```
SIP 401 Challenge
Random challenge (RAND) Network authentication token (AUTN)
Authentication scheme (IMS authentication and key agreement
AKA)
```

3) UE_1 sends a 2nd REGISTER with calculated response (RES) based on a shared secret and previously received RAND

4) UE_1 receives a SIP 200 OK response from the S-CSCF routed back via IMS proxy-servers

Results

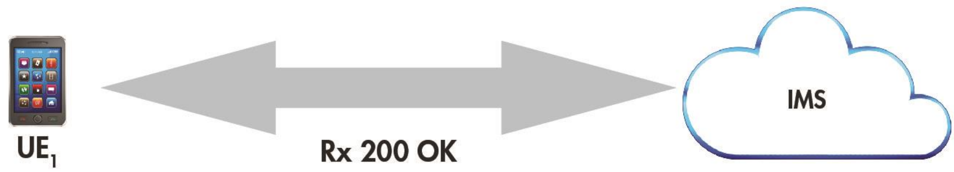
1) UE_1 successfully receives 401 with Random Challenge (RAND) and AKA fields

2) UE_1 successfully receives SIP 200 OK

Test Case 22: VoLTE Emergency Call

Description

This scenario describes the situation where a user needs to perform an emergency call. The US uses 911, the European Union has standardized 112 for dialing an Emergency call.



Initial State

1) UE$_1$ has completed an emergency registration

Test Steps

1) UE$_1$ sends INVITE with Emergency Service URN (Uniform Resource Name):

```
INVITE urn
service:sos
```

2) UE$_1$ receives 200 OK and responds to P-CSCF with ACK

Results

1) UE$_1$ successfully receives 200 OK

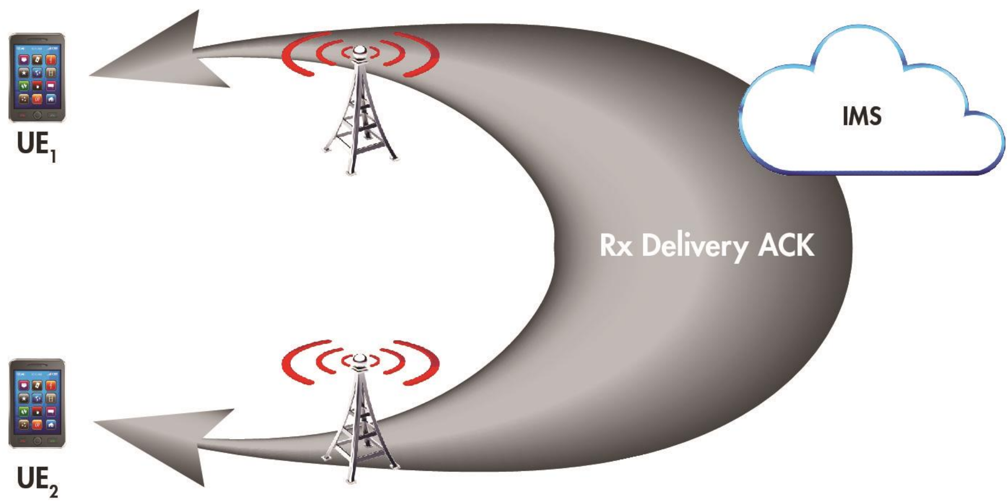
Section 3: VoLTE SMS

The Short Message Service (SMS) allows standardized short text messages over the network. This allowed data transmission over the Circuit Switched 3G network. In LTE all transmissions are data packets. SMS still remains a key customer feature. VoLTE encompasses SMS support.

Test Case 23: VoLTE Send SMS

Description

This scenario describes an SMS scenario between VoLTE-enabled UEs.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are VoLTE Registered and available

Test Steps

1) UE_1 sends MESSAGE request to UE_2:

    `MESSAGE Content-Type=application/vnd.3gpp2.sms, SMS encapsulated SMS P2P`

2) UE_1 receives 200 OK

3) UE_2 receives MESSAGE request

4) UE_2 sends 200 OK

5) UE_1 receives Delivery ACK for MESSAGE request

6) UE_1 sends 200 OK

Results

1) UE_2 receives MESSAGE request

2) UE_1 receives Delivery ACK for MESSAGE request

Section 4: VoLTE Video and Voice Call

IR.94 defines video support in addition to the IR.92-specified voice operation. Video is supported through the configuration of an additional Dedicated Bearer with QCI=2 to handle the video RTP stream. Similar to voice, a variety of codecs may be used. A video call implies both voice and video operation.

Measuring IP Video Quality

The perceptual quality of video transmitted across IP networks is susceptible to degradation from a number of transmission network sources, including frame errors caused by packet loss, discard of packets due to excessive delay/jitter, and discard of packets due to arrival sequencing errors. Simply relying on packet loss statistics, however, is not an accurate way to measure video quality as perceived by viewers. The same degree of packet loss may cause obvious distortion or may not even be noticed by the end user, depending on which video frame types are impaired.

In addition, impairments can be introduced during the encoding/decoding process by the codec itself or an inappropriately low bitrate. The video content (the level of detail and motion on-screen) can also have a significant impact on the visibility of problems. Furthermore, perceptual quality is affected by subjective factors, including human reaction time and the ‘recency effect’. Coupled with the type of content, such as fast motion, high detail, or frequent scene changes, the quality of experience for the viewer will vary even under the same impairment conditions.

Each of these objective and subjective factors must be taken into consideration to accurately estimate IP video perceptual quality.

Video Quality Metrics

Although there is no video quality ITU standard, video quality methods are provided by some companies. Telchemy is one such company that is recognized by the industry. VQmon/HD provides real-time perceptual quality scores, performance statistics, and extensive diagnostic data for monitored video streams in the form of the TVQM (Telchemy Video Quality Metrics) data set.

VQM metrics reported by VQmon/HD fall into two main categories:

1) **Perceptual Quality Metrics –** including mean opinion scores (MOS) for picture quality (MOS-V), audio quality (MOS-A), and combined audio-video quality (MOSAV), expressed in a range of 1 to 5, with 5 being best. For picture quality, both “relative” MOS (which does not consider the resolution of the display, frame rate, or progressive vs. interlaced scanning) and “absolute” MOS (which includes consideration of these factors) are reported.

   TVQM perceptual quality metrics also include an estimated peak signal-to-noise ratio (ESPR) in dB, and a set of metrics indicating the severity level (on scale of 0-10) of several degradation factors, including packet loss, jitter, codec type, etc.

2) **Video Stream Metrics –** including video stream description (image size, codec type, frame rate, etc.); content and scene analysis (detail and motion level) metrics; frame statistics indicating the number and proportion of each frame type (I, B, P, SI, and SP) received/impaired/lost/discarded; average and maximum bandwidth for each frame type and for the stream overall; video stream jitter and delay metrics; and interval metrics.

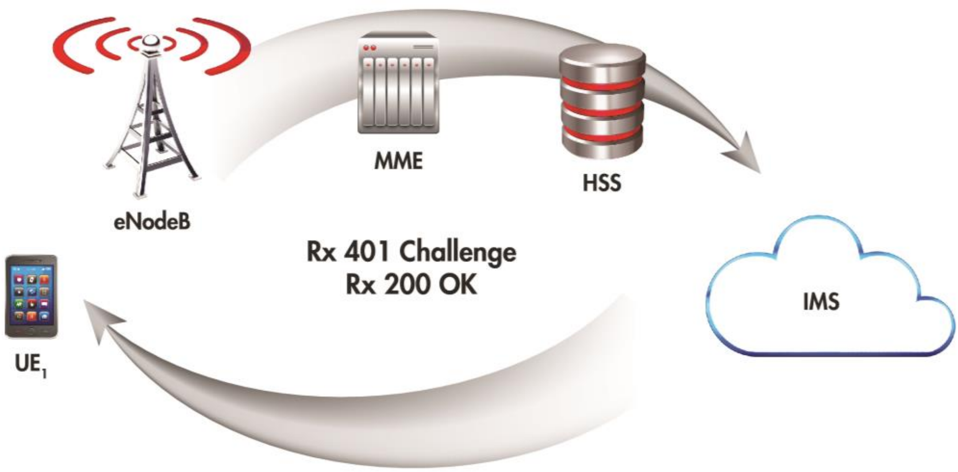
The table below lists some of the perceptual quality metrics reported by VQmon/HD, including acceptable ranges for each.

Acronym	Permitted Range	VQmon Scaled Range	Description
MOS-V	1-5	1-5	VQmon Picture Quality. A VQmon MOS representing video service picture quality. The score also considers the original video quality (before encoding and transmission) and the video content's sensitivity against video packet loss/discard.
EPSNR	0-60 dB	0-60 dB	VQmon/HD Estimated Peak Signal to Noise Ratio. A measurement of the quality of a video signal. This corresponding to the maximum possible signal energy versus the energy of the noise.

Test Case 24: VoLTE IR.94 Registration with IMS AKA Authentication

Description

This test case describes an IR.94 UE IMS Registration flow.



Initial State

1) UE_1 is EPS attached and has the default bearer activated for VoLTE APN

2) UE_1 has obtained P-CSCF IP address

Test Steps

1) A SIP Register is initiated from UE to P-CSCF, uses Contact header field to indicate video media support:

```
Contact: sip:user@example.com
audio
video
```

2) The rest of the steps are similar with those for an IR.92 only UE registration; see Test Case 2

Results

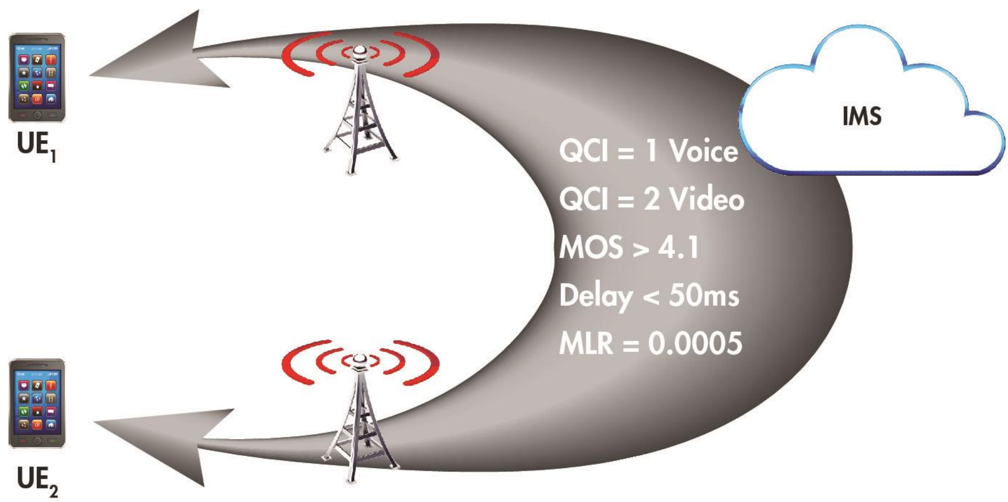
1) UE_1 successfully receives 401 with Random Challenge (RAND) AKA

2) UE_1 successfully receives SIP 200 OK

Test Case 25: VoLTE 2-Way Video Call with 2-Way Audio

Description

This scenario describes a 2-way video VoLTE call with 2-way audio.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are VoLTE Registered and available

Test Steps

1) UE_1 sends SIP INVITE message containing SDP offer (UE_1's IP address and ports for bi-directional audio and video media) to P-CSCF

2) UE_2 responds with 183 Session Progress

3) UE_1 sends PRACK for reliability

4) UE_2 sends 180 Ringing

5) UE_2 responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the voice codec and H.264 as the video codec to be used in this call

6) SIP call is established, RTP streaming starts. Voice and video dedicated bearers are configured at this stage

7) The rest of the steps are similar with a simple 2-way VoLTE voice call per Test Case 6

Results

1) Successful SIP Signaling execution to complete the INVITE process; ACK is received by UE_2

2) Successful initiation of a voice stream, with correct properties, on a dedicated bearer (QCI=1)

3) Successful initiation of a video stream, with correct properties, on a dedicated bearer (QCI=2)

4) RTP packets are using dedicated bearers

5) QoE assessment of MOS and PESQ for voice quality and MDI (Media Delivery Index) with DF (Delay Factor) and MLR (Media Loss Rate) components

    a) MOS is expected to have a value > 4.1

    b) Maximum acceptable DF is between 9 and 50ms

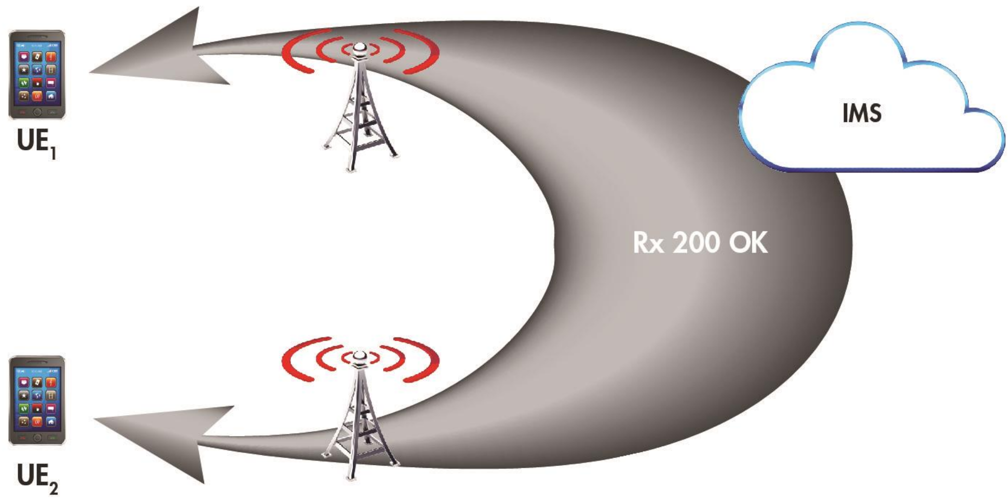
    c) Maximum acceptable MLR = 0.0005

6) Successful teardown of the call by UE_1, BYE is responded with 200 OK

Test Case 26: VoLTE Video Call Originator Terminates

Description

This scenario describes a 2-way video and audio VoLTE call where the call originator terminates it.



Initial State

1) UE$_1$ and UE$_2$ are EPS attached and have the default bearer activated on VoLTE APN

2) UE$_1$ and UE$_2$ are VoLTE Registered and available

Test Steps

1) UE$_1$ sends SIP INVITE message containing SDP offer (UE$_1$'s IP address and ports for bi-directional audio and video media) to P-CSCF

2) UE$_2$ responds with 183 Session Progress

3) UE$_1$ sends PRACK for reliability

4) UE$_2$ sends 180 Ringing

5) UE$_2$ responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the voice codec and H.264 as the video codec to be used in this call

6) SIP call is established, RTP streaming starts and lasts for 2 minutes. Voice and video dedicated bearers are configured at this stage

7) UE_1 sends BYE

8) UE_2 responds with 200 OK

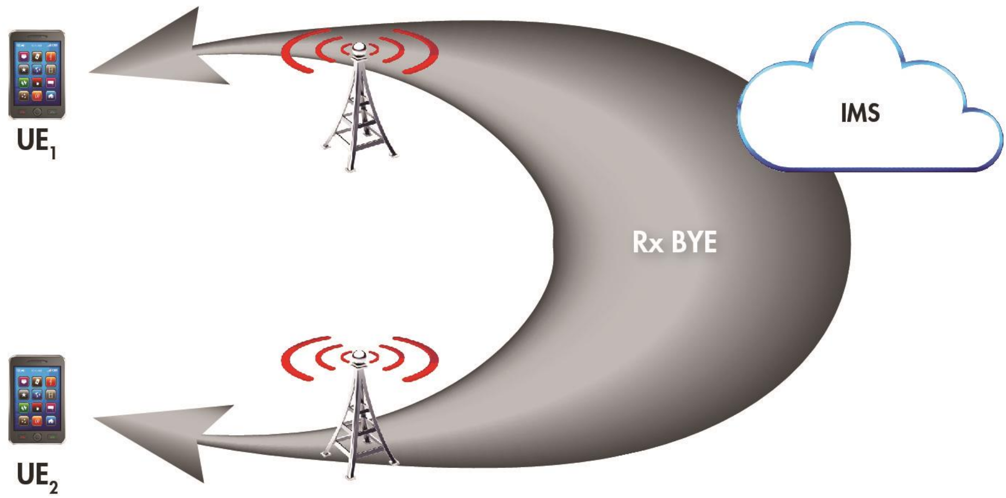
Results

1) Successful teardown of the call by UE_1, BYE is responded with 200 OK

Test Case 27: VoLTE Video Call, Called Party Terminates

Description

This scenario describes a 2-way video and audio VoLTE call where the called party terminates the call.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are VoLTE Registered and available

Test Steps

1) UE_1 sends SIP INVITE message containing SDP offer (UE_1's IP address and ports for bi-directional audio and video media) to P-CSCF

2) UE_2 responds with 183 Session Progress

3) UE_1 sends PRACK for reliability

4) UE_2 sends 180 Ringing

5) UE_2 responds with 200 OK and receives ACK; 200 OK contains a SDP answer that selects AMR-WB as the voice codec and H.264 as the video codec to be used in this call

6) SIP call is established, RTP streaming starts and lasts for 2 minutes. Voice and video dedicated bearers are configured at this stage

7) UE_2 sends BYE

8) UE_1 responds with 200 OK

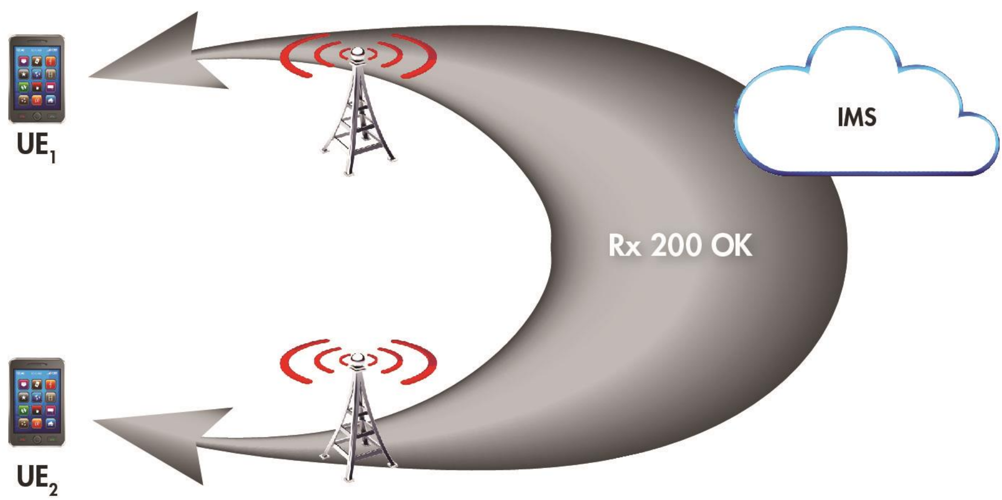
Results

1) Successful teardown of the call by UE_2, BYE is responded with 200 OK

Test Case 28: Change of Video Parameters

Description

During a call, UE_1 initiates a change of video parameters, driven by the IMS application, negotiating up or down depending on the call quality. Network initiation of reconfiguration is possible by the PCEF to Bandwidth shape or the PCRF to Service downgrade.



Initial State

1) UE_1 and UE_2 are VoLTE Registered
2) A video call is established between UE_1 and UE_2, AMR-WB and H.264 are used

Test Steps

1) UE_1 sends UE_2 Re-INVITE with SDP Offer
2) UE_2 responds with 200 OK with SDP answer accepting the new parameters
3) Updated MBR/GBR of the media streams is now established
4) After 3 minutes UE_2 ends the call by sending BYE to UE_1
5) UE_1 responds with 200 OK

Results

1) UE_2 receives Re-INVITE with SDP Offer containing new video (H.264) parameters
2) UE_1 receives 200 OK
3) UE_1 receives BYE
4) UE_2 receives 200 OK

Test Case 29: Video Call – a User Stops Video

Description

The user in this video call disables the video functionality but continues with voice. This allows users an important privacy capability.

Initial State

1) UE_1 and UE_2 are VoLTE Registered

2) A video call is established between UE_1 and UE_2 and is ongoing

3) RTP media packets are using dedicated bearers (QCI=1, GBR for voice and QCI=2, GBR for video)

Test Steps

1) UE_1 sends RE-INVITE to UE_2 with SDP Offer:

```
RE-INVITE SDP
m=video RTP/AVP 21
a=recvonly
```

2) UE_2 accepts stream stop and sends 200 OK with SDP Answer:

```
200 OK SDP
m=video RTP/AVP 21
a=sendonly
```

3) RTP video media stream from UE_1 to UE_2 stops. Between UE_1 and UE_2 exists a 2-way voice call and a video stream from UE_2 to UE_1

4) Call is maintained for 3 minutes

5) UE_1 on hangup initiates a BYE

6) UE_2 responds with 200 OK, which reaches UE_1

Results

1) UE_1 sends RE-INVITE to close video

2) UE_1 receives a 200 OK

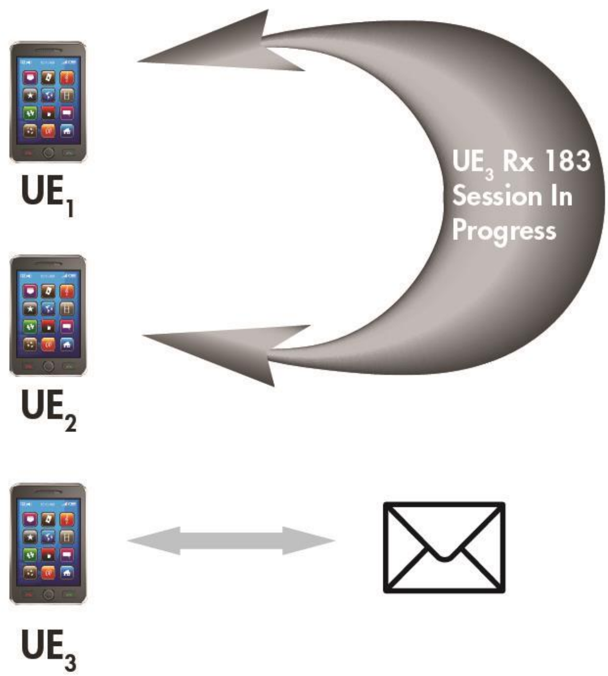
3) Voice continues between UE_1 and UE_2

4) RTP video packets are sent from UE_2 to UE_1 only

Test Case 30: Video Third Call Redirect to Voice Mail on Ignore

Description

This scenario describes an incoming call going to voice Mail.



Initial State

1) UE_1, UE_2, and UE_3 are VoLTE Registered

2) A video call is established between UE_1 and UE_2 and is ongoing

3) RTP media packets are using Dedicated Bearers (QCI=1, GBR for voice and QCI=2, GBR for video)

Test Steps

1) UE_3 initiates a video call to UE_1 and sends INVITE with SDP Offer:

```
INVITE SDP
audio=AMR-WB, AMR
video=H.264
```

2) A “Call-Waiting” screen indication is presented on UE_1, which sends 180 Ringing to UE_3

3) UE_1 selects “Ignore” action

4) UE_1 sends 486 Busy Here, which propagates to TAS; TAS will initiate INVITE, which is routed through S-CSCF to voice mail system

5) UE_3 receives 183 Session Progress with SDP Answer:

```
183 Session Progress SDP
audio=AMR-WB, AMR
```

6) UE_3 responds with PRACK for reliability

7) A voice prompt is played to UE_3, which can deposit his voice only message after the signal

8) At this stage RTP voice packets are sent both ways between UE_3 and voice mail system. RTP voice media packets are using dedicated bearer (QCI=1 and GBR)

9) On hangup UE_2 receives BYE and UE_1 receives 200 OK

Results

1) UE_1 receives INVITE with SDP Offer:

```
audio=AMR-WB, AMR
video=H.264
```

2) UE_1 sends 486 Busy Here

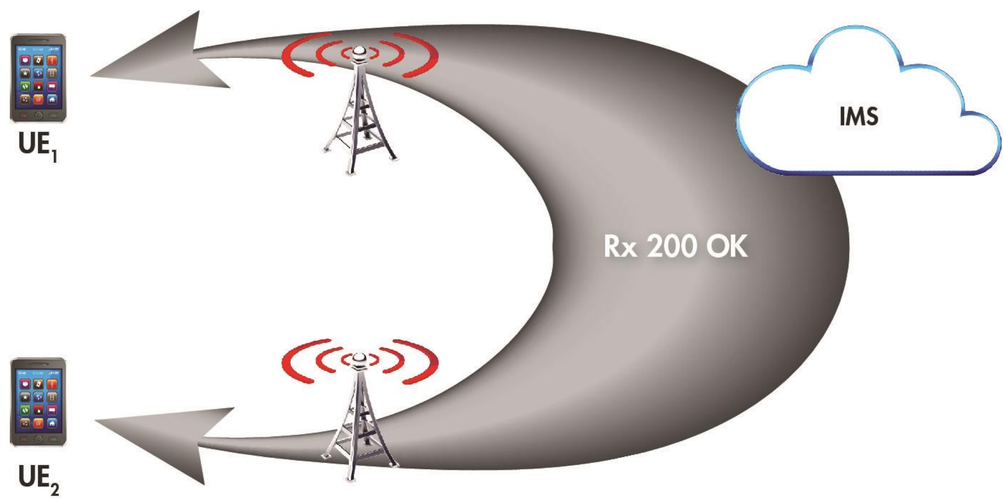
3) UE_3 receives 183 Session Progress with SDP Answer:

```
audio=AMR-WB, AMR
```

Test Case 31: Video Call Accepted as Voice Only

Description

This scenario describes a situation when a call initiated with video sharing is accepted as a voice only call.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) UE_1 and UE_2 are VoLTE Registered and available

Test Steps

1) UE_1 sends INVITE to UE_2 with SDP offer:

```
INVITE SDP
video=H.264
audio=AMR-WB, AMR
```

2) UE_2 displays incoming call notification as “Video Call” and prompts user with options to “Accept Video Call”, “Accept Voice-Only”, “Ignore”

3) UE_2 responds with 180 Ringing, which is propagated back through IMS network to UE_1

4) UE_1 sends PRACK to UE_2 for reliability

5) UE_2 user selects “Accept Voice-Only” and UE_2 sends 200 OK to UE_1 with HD voice only SDP answer (no video support):

```
200 SDP
video=port 0
audio=AMR-WB
```

6) UE_1 receives 200 OK with the SDP Answer and responds with ACK. SIP call is now established, RTP voice streaming starts with AMR-WB codec; voice Dedicated Bearer is created at this stage and follows the same procedure as in Test Case 6

7) Call is maintained for 3 minutes

8) UE_1 on hangup initiates a BYE

9) UE_2 responds with 200 OK, which reaches UE_1

Results

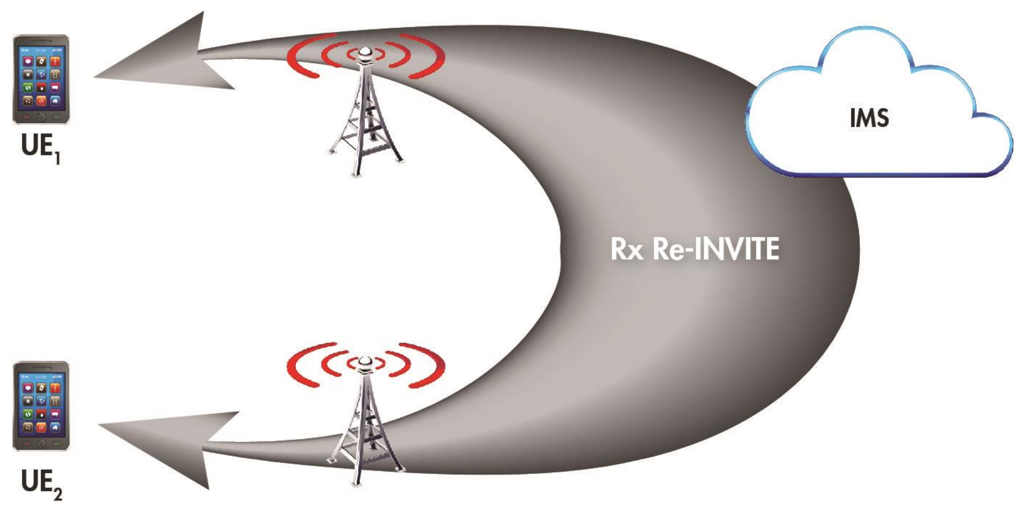
1) UE_2 sends 200 OK for voice only

2) UE_1 receives 200 OK to Invite process assigned for voice only

Test Case 32: Voice Call Transition to Video Call Accepted

Description

This scenario describes a voice call that is upgraded to a video call.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) A VoLTE voice call has been setup between UE_1 and UE_2 and RTP voice streams are exchanged between user devices on a dedicated bearer (QCI=1 and GBR)

Test Steps

1) UE_2 sends Re-INVITE to UE_1 with voice and video SDP offer:

```
Re-INVITE SDP
video=H.264
audio=AMR-WB
```

2) UE_1 accepts request to upgrade the call with video support and sends 200 OK:

```
200 OK SDP
video=H.264
audio=AMR-WB
```

3) RTP video streaming starts with H.264 codec; a new dedicated bearer for video streaming is configured at this stage (QCI=2 with GBR) and follows the same procedure as voice dedicated bearer in Test Case 6

4) Voice call continues and is maintained for 3 minutes

5) UE_1 on hangup initiates a BYE

6) UE_2 responds with 200 OK, which reaches UE_1

Results

1) UE_1 receives Re-INVITE

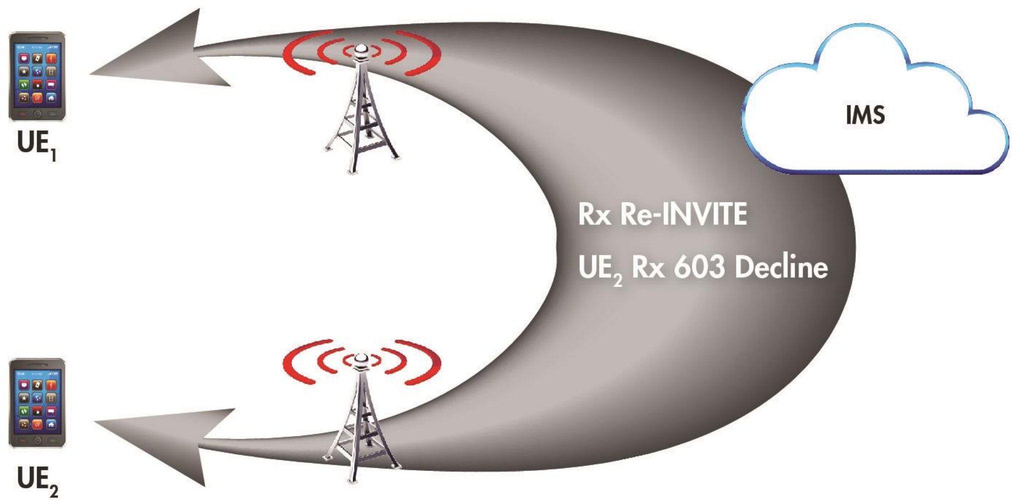
2) UE_2 receives 200 OK with SDP Answer containing H.264 video support

3) Successful initiation of voice and video streams, with correct properties, on a dedicated bearer (QCI=1, QCI=2)

Test Case 33: Voice Call Transition to Video Call Ignored/Time-out

Description

This scenario describes a situation where a request to upgrade voice call to video call is ignored.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) A VoLTE voice call has been setup between UE_1 and UE_2 and RTP voice streams are exchanged on dedicated bearer (QCI=1 and GBR) between these users

Test Steps

1) UE_2 sends Re-INVITE to UE_1 with voice and video SDP offer:

```
Re-INVITE SDP
video=H.264
audio=AMR-WB
```

2) UE_1 ignores display message to upgrade the call with video

3) Call upgrade times out and TAS sends CANCEL to UE_1

4) UE_1 responds with 200 OK

5) UE_1 sends 487 Request Terminated to P-CSCF, which forwards it to TAS through S-CSCF

6) TAS responds with 603 Decline, which is propagated to UE_2

7) Voice call continues and is maintained for 3 minutes

8) UE_1 on hangup initiates a BYE

9) UE_2 responds with 200 OK, which reaches UE_1

Results

1) UE_1 receives Re-INVITE

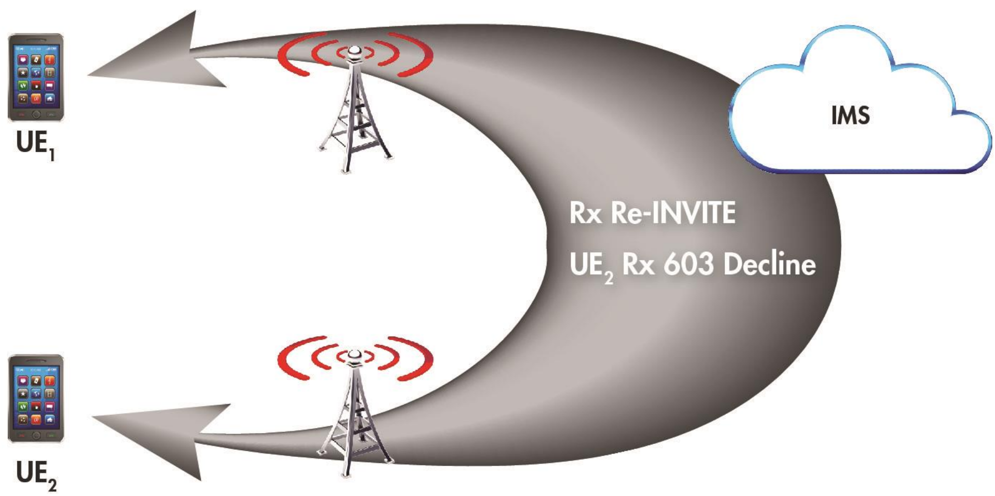
2) UE_1 sends 487 Request Terminated to P-CSCF

3) UE_2 receives 603 Decline

Test Case 34: Voice Call Transition to Video Call Rejected

Description

This scenario describes a situation where a request to upgrade voice call to video call is rejected.



Initial State

1) UE_1 and UE_2 are EPS attached and have the default bearer activated on VoLTE APN

2) A VoLTE voice call has been setup between UE_1 and UE_2 and RTP voice streams are exchanged between user devices on a dedicated bearer (QCI=1 and GBR)

Test Steps

1) UE_2 sends Re-INVITE to UE_1 with voice and video SDP offer:

```
Re-INVITE SDP
video=H.264
audio=AMR-WB
```

2) UE_1 rejects option to upgrade to a video call and sends 603 Decline, which is propagated back to UE_2

3) Voice only call continues for 3 minutes

4) UE_1 on hangup initiates a BYE

5) UE_2 responds with 200 OK, which reaches UE_1

Results

1) UE_1 receives Re-INVITE

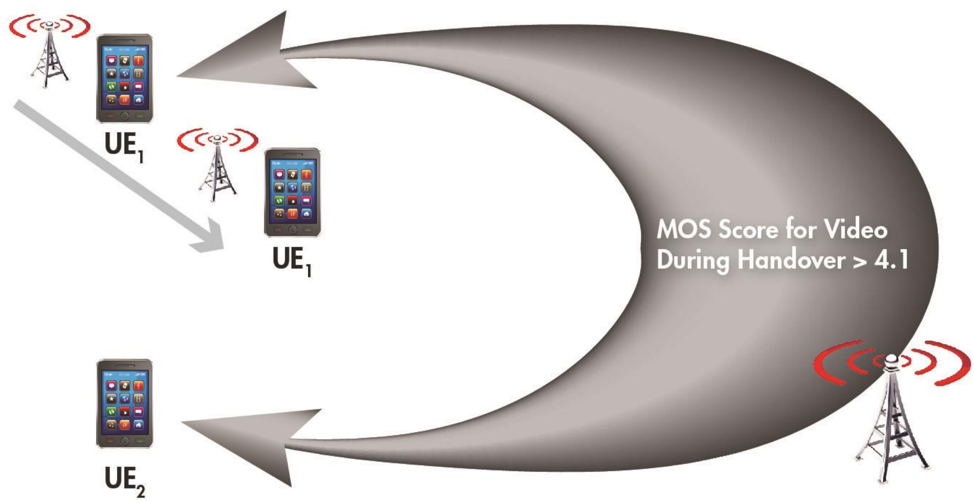
2) UE_2 receives 603 Decline

Test Case 35: Handover During VoLTE Video Call

Description

This test case explores user VoLTE video call quality under a handover situation.

This handover may be Intra-eNodeB, Inter-eNodeB handover across frequencies, bands or duplexing (FDD/TDD). This may invoke an X2, S11 or S1 with MME/S-GW reallocation.



Initial State

1) UE_1 and UE_2 are VoLTE Registered and available.

Test Steps

1) UE_1 calls UE_2 successfully establishing a 2-way video and audio call. RTP packets are sent and received on Dedicated Bearer (QCI=1, GBR for voice and QCI=2, GBR for video)

2) After 90 seconds UE_1 provides RRC Measurement Reports that detail better reception of Sector C than Sector A

3) The eNodeB then directs UE_1 to perform a handover across these sectors by sending RRC Connection Reconfiguration (Handover Command)

4) UE_1 performs a RACH procedure on the target sector/eNodeB

5) UE_1 sends a RRC Connection Reconfiguration Complete message to the target eNodeB on success

6) A DRB is established successfully between UE_1 and target eNodeB

7) Call continues for 30-60 seconds more

8) UE_1 ends the call by sending BYE. UE_2 responds with 200 OK

Results

1) UE_1 receives RRC Connection Reconfiguration (handover Command)

2) Handover signaling is successful

3) New Radio Bearer to target sector is setup successfully

4) RTP packets are using Dedicated Bearers (QCI=1 for voice and QCI=2 for video)

5) QoE assessment of MOS and PESQ for voice quality and MDI (Media Delivery Index) with DF (Delay Factor) and MLR (Media Loss Rate) components.

   a) MOS is expected to have a value > 4.1

   b) Maximum acceptable DF is between 9 and 50ms

   c) Maximum acceptable MLR = 0.0005

6) Voice stream Packet Loss < 0.001

7) Voice stream Jitter unaffected

Test Case 36: Video Call Put on Hold

Description

Here an existing video call is put on hold to initiate a new video call.

Initial State

1) UE_1, UE_2, and UE_3 are VoLTE Registered

2) A video call is established between UE_1 and UE_2 and is ongoing

3) RTP media packets are using dedicated bearer (QCI=1, GBR for voice and QCI=2, GBR for video)

Test Steps

1) UE_3 initiates a video call to UE_1 and sends INVITE with SDP Offer:

```
INVITE SDP Offer
audio=AMR-WB, AMR
video=H.264
```

2) A "Call-Waiting" screen indication is presented on UE_1, which sends 180 Ringing to UE_3

3) UE_3 responds with PRACK for reliability

4) UE_1 responds with 200 OK

5) UE_1 accepts the video call from UE_3 and sends Re-INVITE with SDP Offer to UE_2, which is put on “Hold”:

```
Re-INVITE SDP
audio=sendonly
```

6) UE_2 sends 200 OK with SDP Answer to UE_1:

```
200 OK SDP
audio=recvonly
```

7) UE_1 responds with ACK

8) Audio streams between UE_1 and UE_2 are put on Hold

9) UE_1 sends 200 OK with SDP Answer to UE_3:

```
200 OK SDP
audio=AMR-WB
```

10) At this point RTP voice and video streams are exchanged between UE_1 and UE_3 on Dedicated Bearers (QCI=1, GBR for voice and QCI=2, GBR for video); video call between UE_1 and UE_3 takes 30 seconds

11) UE_1 hangs-up video call with UE_3 and sends BYE to UE_3

12) UE_3 responds with 200 OK

13) UE_1 switches back to the video call with UE_2 by sending UE_2 RE-INVITE with SDP offer to UE_2:

```
RE-INVITE SDP
audio=sendrecv
video=sendrecv
```

14) UE_2 responds with 200 OK containing SDP Answer:

```
200 OK SDP
audio=sendrecv
video=sendrecv
```

15) At this point RTP voice and video streams are exchanged again between UE_1 and UE_2 on Dedicated Bearers (QCI=1, GBR for voice and QCI=2, GBR for video); the call takes 60 seconds

16) UE_1 on hangup initiates a BYE

17) UE_2 responds with 200 OK

Results

1) UE_1 receives the INVITE from UE_3
2) UE_1 receives the 200 OK with SDP Answer from UE_2

Test Case 37: Video Call Switch Hold

Description

A user puts an existing call on hold for incoming video call. The user then switches hold back to the original call.

Initial State

1) UE_1, UE_2, and UE_3 are VoLTE Registered

2) A video call is established between UE_1 and UE_2 and is ongoing

3) RTP media packets are using Dedicated Bearers (QCI=1 and GBR for voice, QCI=2 and GBR for video)

Test Steps

1) UE_3 initiates a video call to UE_1 and sends INVITE with SDP Offer:

```
INVITE: SDP Offer
video=H.264
audio=AMR-WB, AMR
```

2) A “Call-Waiting” screen indication is presented on UE_1, which sends 180 Ringing to UE_3

3) UE_3 responds with PRACK for reliability

4) UE_1 responds to PRACK with 200 OK

5) UE_1 accepts the video call from UE_3 and sends Re-INVITE with SDP Offer to UE_2, which is put on “Hold”:

```
Re-INVITE: SDP
audio=sendonly
video=sendonly
```

6) UE_2 sends 200 OK with SDP Answer to UE_1:

```
200 OK: SDP Answer: SDP:
audio=recvonly
video=recvonly
```

7) UE_1 responds with ACK

8) Audio and video streams between UE_1 and UE_2 are put on Hold

9) UE_1 sends 200 OK with SDP Answer to UE_3:

```
200 OK SDP
video=H.264
audio=AMR-WB
```

10) At this point RTP voice streams are exchanged between UE_1 and UE_3 on a dedicated bearer (QCI=1 and GBR) establishing an ongoing video call

11) UE_1 hangs-up video call with UE_3 and sends BYE to UE_3

12) UE_3 responds with 200 OK

13) UE_1 switches back to the video call with UE_2 by sending RE-INVITE with the SDP offer:

```
RE-INVITE SDP
audio=sendonly
video=sendonly;
```

14) UE_2 responds with 200 OK containing SDP Answer:

```
200 OK SDP
audio=recvonly
video=recvonly
```

15) At this point RTP voice streams are exchanged again between UE_1 and UE_2 on a dedicated bearer (QCI=1 and GBR); video call takes 60 seconds

16) UE_1 on hangup initiates a BYE

17) UE_2 responds with 200 OK, which reaches UE_1

Results

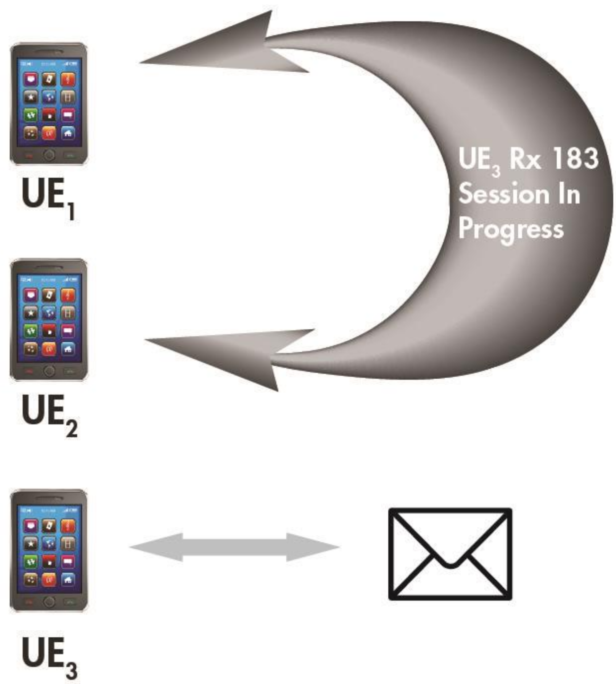
1) UE_1 receives INVITE with SDP media attributes

2) UE_3 receives INVITE with SDP media attributes

Test Case 38: Video Third Call Redirect to Voice Mail

Description

This scenario describes a user allowing an incoming video call to go to Voice Mail.



Initial State

1) UE_1, UE_2, and UE_3 are VoLTE Registered

2) A video call is established between UE_1 and UE_2 and is ongoing

3) RTP voice and video media packets are using Dedicated Bearers (QCI=1 and GBR for voice and QCI=1, GBR for video)

Test Steps

1) UE_3 initiates a video call to UE_1 and sends INVITE with SDP Offer:

```
INVITE SDP
audio=AMR-WB, AMR
video=H.264
```

2) A "Call-Waiting" screen indication is presented on UE_1, which sends 180 Ringing to UE_3

3) UE_1 selects "Ignore" action

4) UE_1 sends 486 Busy Here, which propagates to TAS; TAS initiates INVITE, which is routed through S-CSCF to voice mail system

5) UE_3 receives 183 Session Progress with SDP Answer:

```
183 Session Progress SDP
audio=AMR-WB, AMR
```

6) UE_3 responds with PRACK for reliability

7) A voice prompt is played to UE_3, which can deposit a voice message after the signal

8) At this stage, RTP voice packets are sent both ways between UE_3 and voice mail system; RTP voice media packets are using dedicated bearer (QCI=1 and GBR)

9) On hangup UE_2 receives BYE and UE_1 receives 200 OK

Results

1) UE_1 receives INVITE with SDP Offer:

```
audio=AMR-WB, AMR
video=H.264
```

2) UE_1 sends 486 Busy Here

3) UE_3 receives 183 Session Progress with SDP Answer:

```
audio=AMR-WB, AMR
```

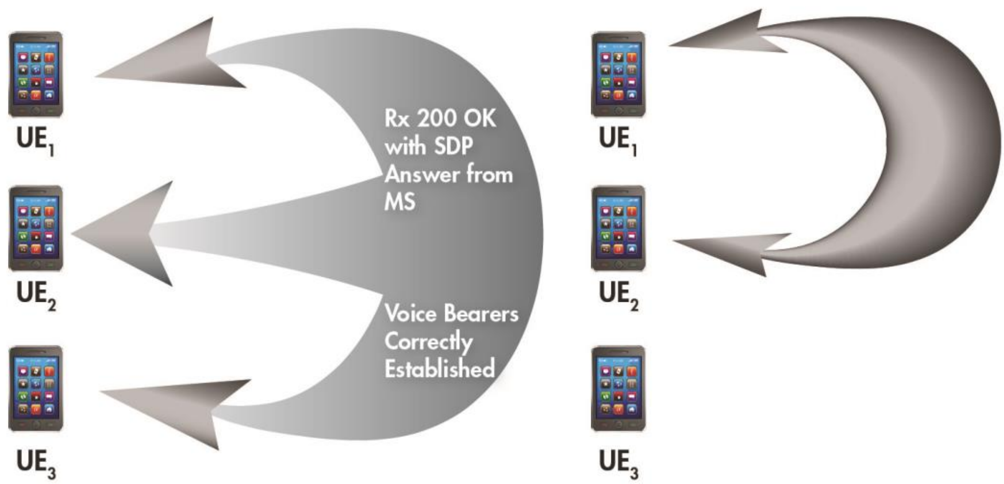
Section 5: Advanced VoLTE Testing

This section details advanced VoLTE testing scenarios, including multi-party calling and multi-tasking.

Test Case 39: Voice Ad-Hoc Multi-Party Conference

Description

Here a user in a call brings in a called third user for a multi-party conference.



Initial State

1) UE_1, UE_2 and UE_3 are VoLTE Registered

2) A voice call is established between UE_1 and UE_2; RTP voice packets are sent both ways

Test Steps

1) UE_1 puts UE_2 on hold by sending Re-INVITE with SDP Offer:

```
Re-INVITE
m=audio
a=sendonly
```

2) UE_2 responds with 200 OK with SDP Answer:

```
m=audio
a=recvonly
```

3) UE_1 sends ACK to UE_2

4) UE_1 establishes a voice call with UE_3

5) UE_1 puts UE_3 on hold by sending Re-INVITE with SDP Offer:

```
Re-INVITE
m=audio
a=sendonly
```

6) UE_3 responds with 200 OK with SDP Answer

7) UE_1 sends ACK to UE_3

8) UE_1 creates a conference by sending INVITE to conference URI, with SDP Offer:

```
INVITE
m=audio
a=sendrecv
```

9) UE_1 receives 200 OK with SDP Answer from the Media Server

10) UE_1 sends ACK

11) At this moment an active RTP path for voice is created between UE_1 and Media Server

12) UE_1 sends SUBSCRIBE to TAS

13) UE_1 receives 200 OK from TAS

14) TAS sends NOTIFY to UE_1 for current conference events

15) UE_1 invites UE_2 to conference by sending REFER request

16) The REFER request contains REFER TO header that indicates UE_2 and REPLACE header indicating that this REFER request will replace UE_1 to UE_2 call leg

17) UE_1 receives 202 Accepted to acknowledge REFER request

18) UE_1 receives NOTIFY with the status of ‘REFER’

19) UE_1 sends 200 OK

20) UE_2 receives INVITE without SDP

21) UE_2 answers with 200 OK with SDP Offer that includes media codec set that was utilized in the initial call with UE_1

22) UE_2 receives ACK with SDP Answer

23) At this moment an active RTP path for voice is created between UE_2 and Media Server

24) UE_1 receives NOTIFY about successfully performed joining to conference

25) UE_1 sends 200 OK

26) UE_1 receives NOTIFY about UE_2 successfully joined the conference

27) UE_1 sends 200 OK

28) UE_1 disconnects the dialog between UE_1 and UE_2 by sending BYE

29) UE_1 invites UE_3 to conference by sending REFER request

30) The REFER request contains REFER TO header that indicates UE_3 and REPLACE header indicating that this REFER request will replace UE_1 to UE_3 call leg

31) UE_1 receives 202 Accepted to acknowledge REFER request

32) UE_1 receives NOTIFY with the status of 'REFER'

33) UE_1 sends 200 OK

34) UE_3 receives INVITE without SDP

35) UE_3 answers with 200 OK with SDP Offer that includes media codec set that was utilized in the initial call with UE_1

36) UE_3 receives ACK with SDP Answer

37) At this moment an active RTP path for voice is created between UE_3 and Media Server

38) UE_1 receives NOTIFY about successfully performed joining to conference

39) UE_1 sends 200 OK

40) UE_1 receives NOTIFY about UE_3 successfully joined the conference

41) UE_1 sends 200 OK

42) UE_1 disconnects the dialog between UE_1 and UE_3 by sending BYE

43) RTP voice media packets are exchanged in both ways between each UE and the Media Server

44) RTP voice packets are using Dedicated Bearers (QCI=1, GBR)

45) UE_1 ends the conference call by sending BYE

46) UE_2 and UE_3 receive BYE and respond with 200 OK

47) UE_1 receives 200 OK

Results

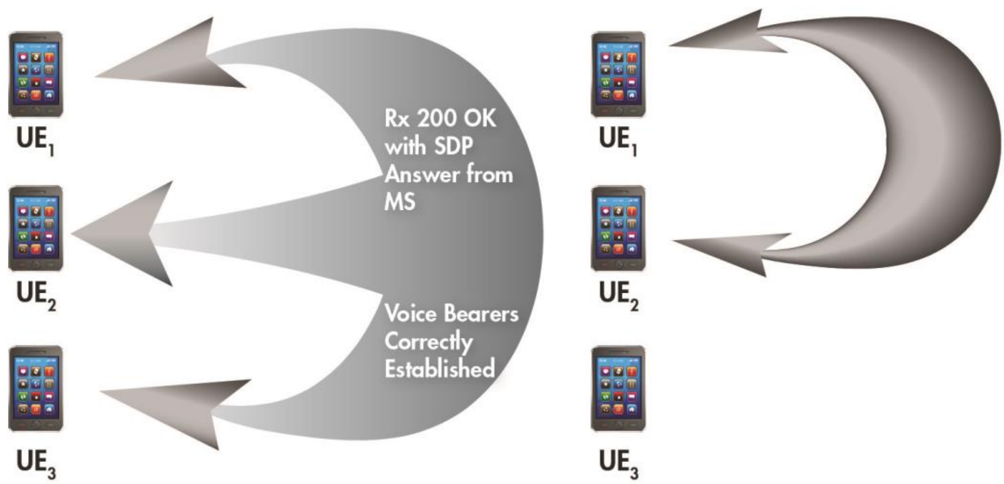
1) UE_1 receives 200 OK with SDP Answer from the Media Server after starting the multi-party conference

2) Voice Bearers are correctly established for all 3 UEs

Test Case 40: Video Ad-Hoc Multi-Party Conference

Description

Here a user in a call brings in a called third user for a multi-party conference.



Initial State

1) UE_1, UE_2, and UE_3 are VoLTE Registered

2) A video call is established between UE_1 and UE_2; RTP voice and video packets are sent in both ways between UE_1 and UE_2

Test Steps

1) UE_1 puts UE_2 on hold by sending Re-INVITE with SDP Offer:

```
Re-INVITE
m=audio
a=sendonly
m=video
a=sendonly
```

2) UE_2 responds with 200 OK with SDP Answer:

```
m=audio
a=recvonly
m=video
a=recvonly
```

3) UE_1 sends ACK to UE_2

4) UE_1 establishes a voice call with UE_3

5) UE_1 puts UE_3 on hold by sending Re-INVITE with SDP Offer:

```
Re-INVITE
m=audio
a=sendonly
m=video
a=sendonly
```

6) UE_3 responds with 200 OK with SDP Answer

7) UE_1 sends ACK to UE_3

8) UE_1 creates a conference by sending INVITE to conference URI, with SDP Offer:

```
INVITE
m=audio
a=sendrecv
m=video
a=sendrecv
```

9) UE_1 receives 200 OK with SDP Answer from the Media Server

10) UE_1 sends ACK

11) At this moment an active RTP path for voice and video is created between UE_1 and Media Server

12) UE_1 sends SUBSCRIBE to TAS

13) UE_1 receives 200 OK from TAS

14) TAS sends NOTIFY to UE_1 for current conference events

15) UE_1 invites UE_2 to conference by sending REFER request

16) The REFER request contains REFER TO header that indicates UE_2 and REPLACE header indicating that this REFER request will replace UE_1 to UE_2 call leg

17) UE_1 receives 202 Accepted to acknowledge REFER request

18) UE_1 receives NOTIFY with the status of ‘REFER’

19) UE_1 sends 200 OK

20) UE_2 receives INVITE without SDP

21) UE_2 answers with 200 OK with SDP Offer that includes media codec set that was utilized in the initial call with UE_1

22) UE_2 receives ACK with SDP Answer

23) At this moment an active RTP path for voice is created between UE_2 and Media Server

24) UE_1 receives NOTIFY about successfully performed joining to conference

25) UE_1 sends 200 OK

26) UE_1 receives NOTIFY about UE_2 successfully joined the conference

27) UE_1 sends 200 OK

28) UE_1 disconnects the dialog between UE_1 and UE_2 by sending BYE

29) UE_1 invites UE_3 to conference by sending REFER request

30) The REFER request contains REFER TO header that indicates UE_3 and REPLACE header indicating that this REFER request will replace UE_1 to UE_3 call leg

31) UE_1 receives 202 Accepted to acknowledge REFER request

32) UE_1 receives NOTIFY with the status of ‘REFER’

33) UE_1 sends 200 OK

34) UE_3 receives INVITE without SDP

35) UE_3 answers with 200 OK with SDP Offer that includes media codecs set that was utilized in the initial call with UE_1

36) UE_3 receives ACK with SDP Answer

37) At this moment an active RTP path for voice and video is created between UE_3 and Media Server

38) UE_1 receives NOTIFY about successfully performed joining to conference

39) UE_1 sends 200 OK

40) UE_1 receives NOTIFY about UE_3 successfully joined the conference

41) UE_1 sends 200 OK

42) UE_1 disconnects the dialog between UE_1 and UE_3 by sending BYE

43) RTP voice and video media packets are exchanged in both ways between each UE and the Media Server

44) RTP voice packets are using Dedicated Bearers (QCI=1, GBR)

45) RTP video packets are using Dedicated Bearers (QCI=2, GBR)

46) UE_1 ends the conference call by sending BYE

47) UE_2 and UE_3 receive BYE and respond with 200 OK

48) UE_1 receives 200 OK

Results

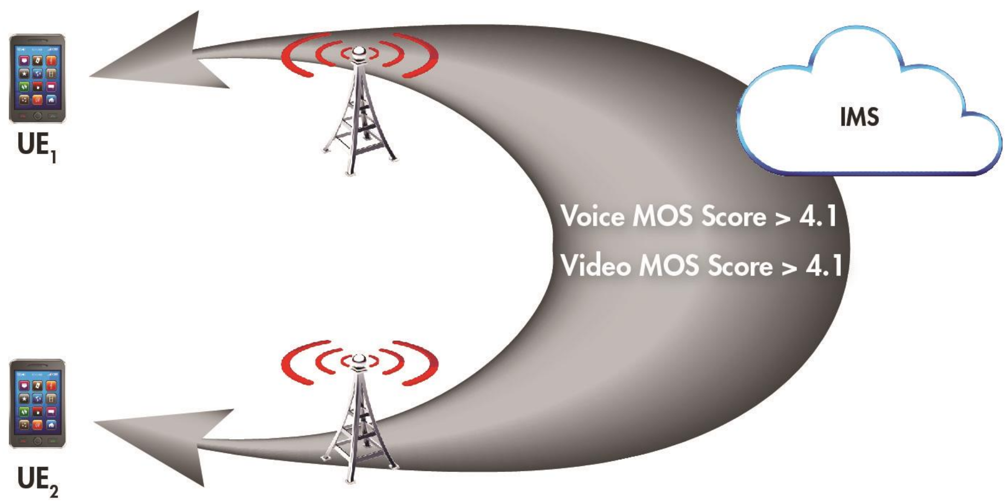
1) UE_1 receives 200 OK with SDP Answer from the Media Server after starting the Multi-party conference

2) Voice and video Bearers are correctly established for all 3 UEs

Test Case 41: Multitasking + Video Call

Description

This scenario examines the effect of other default bearer traffic activity on the video call quality.



Initial State

1) UE_1, UE_2, and UE_3 are VoLTE Registered

2) A video call is established between UE_1 and UE_2 and is ongoing

Test Steps

1) Another application is opened on UE_2; UE_2 sends a Re-INVITE to UE_1; SDP media description for video has attribute set to "sendonly":

```
m=video RTP/AVP 21
a=sendonly
```

2) UE_1 replies with 200 OK:

```
video=H.264
a=recvonly
audio=AMR-WB
```

3) The video stream is stopped from UE_1 to UE_2

4) UE_2 returns to VoLTE application and sends Re-INVITE to UE_1 with SDP Offer:

```
m=video RTP/AVP 21
a=sendrecv
```

5) UE_1 responds with 200 OK and SDP Answer:

```
video=H.264(sendrecv)
audio=AMR-WB
```

6) Both voice and video RTP packets are sent both ways between UE_1 and UE_2

Results

1) UE_1 receives Re-INVITE with "sendonly" attribute for video

2) UE_1 receives Re-INVITE with "sendrecv" attribute for video

3) RTP voice media packets are using Dedicated Bearers (QCI=1, GBR for voice and QCI=2, GBR for video)

4) QoE assessment of MOS and PESQ for voice quality and MDI (Media Delivery Index) with DF (Delay Factor) and MLR (Media Loss Rate) components:

    a) MOS is expected to have a value > 4.1

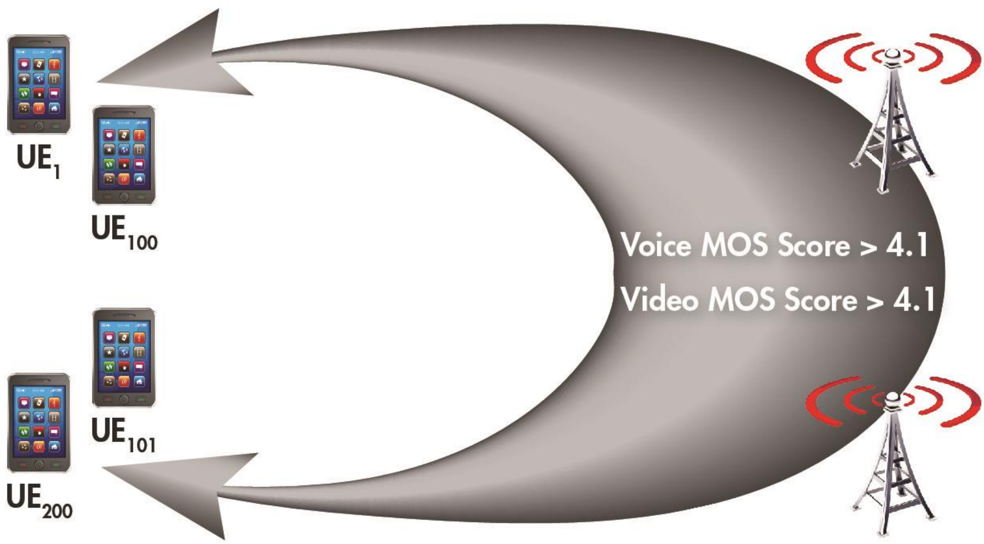
    b) Maximum acceptable DF is between 9 and 50ms

    c) Maximum acceptable MLR = 0.0005

Test Case 42: VoLTE Video Load Scenario

Description

This test case explores user VoLTE quality under load conditions. While UEs load the eNodeB the VoLTE quality should not decrease.



Initial State

1) UE_1... UE_{200} are VoLTE Registered

2) Video calls are established between UE_1 and UE_{101}... UE_{100} and UE_{200} and is ongoing

Test Steps

1) $UE_{1...100}$ initiate video calls to $UE_{101...200}$ following the process in Test Case 25

2) An Invite rate of 10 per second should be used

3) While ramping the voice and video, quality should be monitored to ensure quality remains excellent

Results

1) Monitor voice and video quality MOS score > 4.1

2) Media one-way delay under 150ms

Section 6: Real-World VoLTE Subscriber Modeling

To understand how your VoLTE network and services will perform while in actual use, testing needs to employ subscriber traffic and patterns that happen in real life. You may know how your network handles a particular type and constant load of traffic, but what happens to it when an airbus lands and most of the 879 passengers and crew turn on and use their phones all at once? Or when power is restored in a city?

The key to understanding the impact of these real-world events is to use real-world subscriber modeling that subjects an eNodeB to a challenging array of real-world scenarios. To target key performance metrics with real-world subscriber modeling, be sure your performance testing includes:

- Full-featured LTE UE emulation with FDD, TDD, DRX, etc.
- High capacity and rate, and multiple UE ranges, each with UE-specific properties
- Mobile application modeling with voice, video, and data traffic that includes QoE (MOS, PESQ) and QoS measurements
- Complex signaling operation including Attach, Detach, Handover, TAU, and Idle Mode
- LTE Inter- and Intra-eNodeB handover across all connected sectors
- Channel modeling that allows UE cell center/edge simulation with LTE DL Fast Fading emulations including Pedestrian, Vehicle, Urban, and High-Speed Train

Test Case 43: Airbus A380 Landing

Description

This scenario involves an Airbus A380 landing at Heathrow, passengers turning on cell phones with data roaming getting updates, and making VoLTE calls.

Initial State

1) UE_1 .. UE_{1000} are registered for the LTE network per sector

Test Steps

1) UEs initiate attaches at a rate of 50/s
2) 1000 UEs with VoLTE services commence IMS Registration
   a) Rate of 10 UE/s
3) 50 UEs initiate VoLTE connections to held messages
4) 50 UEs download stored SMSs

Results

1) RAN and core infrastructure handle Attaches at 50/s
2) 10 UE/s successfully complete IMS Registration
3) Successful download of stored voice messages
4) Successful download of stored SMS

Test Case 44: Power Outage Restored

Description

In this case, after a city regional power outage is restored, the network experiences an Attach Storm, with all succeeding UEs attempting IMS Registration.

Initial State

1) UE_1 .. UE_{1000} are registered for the LTE network per sector

Test Steps

1) UEs initiate attaches at a rate of 100/s

2) Successful UEs immediately start IMS Registration

Results

1) Successful handling of Attach Storm with UEs backing off

    a) Actual attach rate < 100/s

2) IMS Registrations sustained at Attach rate

Test Case 45: High-Density Financial District

Description

Extensive smart phone use and VoLTE operation occur in this model.

Initial State

1) UE_1 .. UE_{1000} are registered for the LTE network per sector
2) UEs initiate attaches at a rate of 10/s
3) Successful UEs immediately start IMS Registration

Test Steps

1) All UEs start HTTP downloads at 72kbps
2) Rotating 250 UEs initiate 3 minute VoLTE voice and video calls

Results

1) All UEs achieve 72Mbps aggregate HTTP download
2) Rotating 250 UEs successfully complete VoLTE calls
    a) Each VoLTE call has a voice and video MOS of "Excellent"

Test Case 46: Rush Hour Commuter Traffic

Description

This scenario emulates commuters making VoLTE calls that include extensive mobility.

Initial State

1) UE_1 .. UE_{1000} are registered for the LTE network per sector
2) UEs initiate attaches at a rate of 10/s
3) Successful UEs immediately start IMS Registration

Test Steps

1) Rotating 250 UEs initiate 10 minute VoLTE voice and video calls
2) All UEs eNodeB handover every 4 minutes
   a) Back and forth between 2 UEs or round patterns over multiple eNodeBs
3) Idle and connected mobility – HO (Inter/Intra) and TAU

Results

1) Rotating 250 UEs successfully complete 10 minute VoLTE calls
   a) Each VoLTE call has a voice MOS of "Excellent"
2) Each UE hands-over successfully without a failure

Test Case 47: Large University

Description

This test case employs savvy smart phone use with a triple-play mixture of traffic types.

Initial State

1) UE_1 .. UE_{1000} are registered for the LTE network per sector
2) UEs initiate attaches at a rate of 10/s
3) Successful UEs immediately start IMS Registration

Test Steps

1) Rotating 300 UEs initiate 1 minute HTTP download
2) Rotating 100 UEs initiate 10 minute FTP download
3) Rotating 250 UEs initiate 5 minute VoLTE voice and video calls
4) UEs will be entering and exiting Idle

Results

1) All UEs successfully run HTTP, FTP, and VoLTE sessions
2) Rotating 250 UEs successfully complete 5 minute VoLTE calls
   a) Each VoLTE call has a voice and video MOS of “Excellent”

Ixia VoLTE Test Solutions

Ixia's VoLTE test solutions measure voice quality end-to-end, from the UE through to the IMS core network. Operators can easily compare and contrast the voice quality of OTT service to operator-provided voice services. Using best-in-class industry algorithms, operators can measure the MOS of voice quality before going live. Additionally, Ixia's solutions test the functionality, scalability, and resiliency of LTE infrastructure components and new IMS networks that support VoLTE-based services.

Ixia's VoLTE test solutions offer:

- Full-featured VoIP SIP/IMS testing over wireless stacks
- Hardware-accelerated RTP over GTP that can scale up to more than 72,000 calls per load module
- Flexible functional and negative testing achieved with user control of SIP messaging and content
- Precise QoS measurements of voice quality under stress conditions
- Ability to model complex mobility scenarios, such as intra or inter eNodeB handovers, X2- or S1-based handovers, and S-GW relocations
- Control plane measurements such as call setup time and time to media when the network is under stress conditions
- Simultaneous generation of other types of stateful traffic such as HTTP and video
- MME/eNodeB and LTE Access simulations for testing access and core network simultaneously
- An Rx Diameter interface that is coordinated with the SIP/SDP for S-GW+P-GW and PCRF isolation testing

IxLoad Access

Ixia's LTE IxLoad Access test solution performs UE emulation that can achieve the test cases outlined in this book. It builds on the IxLoad platform that extends from easy to configure layers 4-7 voice, video, and data traffic into UE emulation. It provides the highest density UE emulation on the market, with1,000 connected active UEs simulated per board, high traffic, LTE feature rich, channel modeling, and mobility. This allows realistic subscriber modeling to replicate field issues and prove out capacity planning assumptions.

The following components are used to achieve comprehensive VoLTE testing:

Ixia Test Solution	Function
IxLoad	Software platform for full layer 4-7 testing, includes test creation, execution, analysis, and reporting
Wireless Triple-Play Bundle	Software package with stateful replay and emulation for video, VoIP, VoLTE, data, and peer-to-peer protocols
Xcellon-Ultra NP	Application traffic generation load module
XAir	LTE UE emulation load module
r10 Radio Head	LTE RF interface to LTE base stations, tunable to support all LTE bands
XM/XG Chassis	Ultra-high density and scalable chassis

IxLoad

IxLoad is a unified test solution for testing all aspects of wireless networks end-to-end, including LTE base stations, core network components, and the IMS subsystem. IxLoad supports a broad set of test applications measuring the scalability and capacity of data applications, the quality of rich media services, and the evaluation of security vulnerabilities.

Key Features

- Measures voice quality by a mean opinion score using E-model and perceptual estimation of speech quality (PESQ) algorithms
- High RTP performance supports more than one million concurrent calls with media per chassis
- Measures control plane and media latency, which can be used to understand call setup latency
- Support of the GSMA IR.92 specification
- Emulation of SIP endpoints to initiate and receive voice calls and SMS texts over eGTP
- Emulation of the IMS network (P-CSCF and MGW)
- AMR and AMR-WB codecs
- AKAv1 and AKAv2 authentication
- Default and dedicated bearers

Validating VoLTE

IxLoad's comprehensive VoLTE emulation includes:

Device Under Test	Emulated Nodes
eNodeB	UE, eNodeB, MME, S-GW
MME	HSS, eNodeB, S-GW, MME
S-GW	MME, eNodeB, PDN-GW
PDN-GW	S-GW, PCRF, IP Core
Network	UE, IP Core

More information:
http://www.ixiacom.com/pdfs/datasheets/ixload_volte.pdf

Wireless Triple-Play Bundle

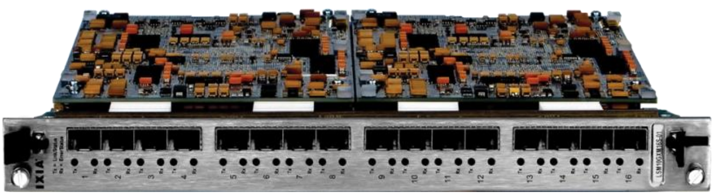
Ixia's IxLoad Wireless Triple-Play Bundle provides a comprehensive solution for testing wireless multiplay networks and services. IxLoad emulates hundreds of thousands of subscribers using the full range of voice, video, and data services. Its unique subscriber modeling provides mixes of wireless user traffic to be applied over time, allowing service providers to ensure quality of experience throughout the entire day. IxLoad is designed for use by NEMs in their R&D facilities and for use by service providers and enterprises in their pre-deployment labs. Ixia's chassis and load modules provide the scale needed to match the capacity of any device or system.

Available protocols include:

Service	Protocols
Internet	HTTP, P2P, FTP, SMTP, POP3, CIFS
Video	IGMP, RTSP, Adobe Flash Player™, Microsoft Silverlight™, Apple HLS, MPEG2, and H.264/AVC
Voice	SIP, MGCP, H.323, H.248, Cisco Skinny™, FAX over IP, video conferencing, and PSTN
Infrastructure	DNS, DHCP, LDAP, and AAA
Encapsulation/security	DHCP, IPsec, PPP/L2TP with integrated 802.1x and NAC authentication

Xcellon-Ultra NP Load Module

Ixia's Xcellon-Ultra NP load module is the highest-performing and most scalable application traffic generation solution available in the industry. They offer complete layer 2-7 packet generation, routing, and application testing functionality in a single XM load module. Xcellon-Ultra NP's CPU count and NP architecture create a powerful application traffic generation platform that emulates millions of real-world application flows, including voice, video, and massive amounts of data.



Key Features

- Flexible resource allocation for optimized L4-7 testing performance
- Wire-speed layer 2-3 traffic generation and analysis, high-performance routing/bridging protocol emulation, and true layer 4-7 application subscriber emulation and traffic generation
- Twelve 1GbE ports and one 10GbE port per module
- Compatible with XG12, XM12, or XM2 chassis

More information:
http://www.ixiacom.com/pdfs/library/quick_ref_sheets/xcellon-np-qrs.pdf

XAir Module

Ixia's XAir module is the next-generation hardware for LTE UE emulation. It delivers unparalleled LTE performance in the smallest footprint, providing the industry's highest UE density. This module allows LTE Advanced feature support. With Ixia's XM and XG platforms, you can easily support multiple sectors.

The XAir module supports complex subscriber modeling with:

- 1000 UEs per sector
- Voice (VoLTE), video, and data traffic support
- QoE analysis and scoring of each traffic stream
- Mobility over multiple sectors
- Channel modeling per UE

Each XAir board supports one sector, with 1GE ports connected to the IxLoad Xcellon-Ultra NP chassis. It supports a direct CPRI interface to an eNodeB or to Ixia's Remote Radio Head r10 units that cover all FDD and TDD frequency bands.

Key Features

- Highest density LTE UE emulation starting at 1000 connected active UEs per board
- Built-in high accuracy 10MHz clock for eNodeB synchronization
- Fully compatible with the Ixia XM and XG chassis and load modules for seamless testing with other Ixia hardware and test applications

More information:
http://www.ixiacom.com/pdfs/datasheets/xair-module.pdf

r10 Wideband Radio Head

Ixia's r10 Wideband Radio Head chassis for LTE and WiMAX testing supports a range of bandwidths and provides a user-friendly interface for simplified remote control of the test platform. A proven market leader as a base station or multiple UE emulator, the module is primarily aimed at LTE base station conformance testing.

Key Features

- Handles LTE UE radio modulation from 690 to 2690 MHz
- FDD or TDD LTE technologies
- Supports 5/10/15/20 MHz channel bandwidths

High-Performance Chassis

Ixia’s VoLTE test solutions are powered by three chassis options:

Chassis	Photo	Description
XG12		XG12 is Ixia's latest chassis technology, and is the test and measurement industry’s highest port density test system for Ethernet. • 12 Slots • Size: 19"w x 19.21”h x 27.2”d (48.26cm x 48.79cm x 69.09cm) • 97 lbs. (44.1 kg) average shipping weight • Supports XM Form Factor (XMFF) load modules • Allows full 12-slot use of the newest high-performance load modules, such as XAir
XM12		XM12 High Performance features the highest port density and performance in the industry for lab-based, rack-mounted operation. • 12 Slots • Size: 17.7"w (19.0"w including rack ears) x 17.5"h x 21.0"d (45.5cm x 44.5cm x 53.3cm) • 83 lbs. (37.65 kg) empty, 88 lbs. (39.92 kg) average shipping weight • Supports XM Form Factor (XMFF) load modules • UL and CE safety approval certifications; valid when the unit is operating from 200-240VAC mains

Chassis	Photo	Description
XM2		XM2 is ideal for desktop testing, smaller scale tests, and remote monitoring. • 2 Slots • Size: 14"w (19.0”w including rack ears) x 4.5”h x 19.25”d (35.6cm x 11.4cm x 48.9cm) with built-in carrying handle. • 25 lbs. (11.3 kg) empty, 30 lbs. (13.61 kg) average shipping weight • Supports XM Form Factor (XMFF) load modules

Key Features

- Ultra-high density and scalability
- Automation, unified APIs, and centralized management
- Multiuser operation
- Full range of interface support
- Hot swappable
- Backward/forward compatibility

More Information:

- Chassis Overview: http://www.ixiacom.com/pdfs/library/quick_ref_sheets/ixia-core-qrs.pdf
- XG12 Chassis: http://www.ixiacom.com/products/network_test/chassis/display?skey=ch_optixia_xg12
- XM12 Chassis: http://www.ixiacom.com/pdfs/datasheets/ch_optixia_xm12.pdf
- XM2 Chassis: http://www.ixiacom.com/pdfs/datasheets/ch_optixia_xm2.pdf

Ixia Hardware Configuration for VoLTE Testing

Carriers can configure Ixia's modular VoLTE test platforms to suit their current needs while offering a simple path to grow their system as services expand. Ixia provides the industry's highest density of UE emulation – up to 6000 UEs in one XG12 chassis. Ixia's platform performance supports very high rate operation, including up to 300 attach attempts per second. Following are examples of how the platform can be configured for testing one to six sectors.

One-Sector Configuration

This configuration of an XM2 chassis with one XAir load module, one Xcellon-Ultra NP load module, and one r10 Radio Head supports 1000 connected UEs.

Three-Sector Configuration

This configuration of an XM12 or XG12 chassis with three XAir load modules, three Xcellon-Ultra NP load modules, and three r10 radio head module supports 3000 connected UEs.

Six-Sector Configuration

This configuration of an XG12 chassis with six XAir load modules, six Xcellon-Ultra NP load modules, and six r10 Radio Head modules supports 6000 connected UEs.

References

Specification	Version	Name
GSMA IR.92	3.0, Dec 2010	IMS Profile for Voice and SMS
GSMA IR.94	3.0, July 2012	IMS Profile for Conversational Video Service
3GPP TS 23.228	9.4.0	IP Multimedia Subsystem (IMS); Stage 2
3GPP TS 24.229	9.60	IP Multimedia Call Control Protocol Based on SIP and SDP, Stage 3
3GPP TS 33.203	9.5.0	R9 Access Security for IP-based Services
RFC 3261	June 2002	SIP: Session Initiation Protocol
RFC 3550	July 2003	RTP: A Transport Protocol for Real-Time Applications
RFC 3551	July 2003	RTP Profile for Audio and Video Conferences with Minimal Control
RFC 4867	April 2007	RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs

Validating VoLTE

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