Telecom Regulatory Authority of India

Consultation Paper

on

Voice Services to LTE users (including VoLTE and CS Fallback)

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Chapter 1

Introduction

1.1 Background

1.1.1 Sub-Clause (v) of clause (b) of sub-section (1) of section 11 of Telecom Regulatory Authority of India Act, 1997 (24 of 1997) mandates the Authority to lay down the standards of quality of service to be provided by the service providers and ensure the quality of service and conduct the periodical survey of such service provided by the service providers so as to protect interest of the consumers of telecommunication services.

1.1.2 In exercise of its functions under the above provisions in the TRAI Act, the Authority had notified the Regulation on Quality of Services (QoS) of Basic and Cellular Mobile Telephone Services, 2000 vide Notification dated 5th of July, 2000. The objectives of these regulations were to create conditions for customer satisfaction by making known the quality of service which the service provider is required to provide and the user has a right to expect; measure the Quality of Service provided by the Service Providers from time to time and to compare them with the benchmarks so as to assess the level of performance; and to generally protect the interests of consumers of telecommunication services.

1.1.3 These regulations were subsequently reviewed and TRAI issued the revised QoS standards for these services in year 2009 as The Standards of Quality of Service of Basic Telephone Service (Wireline) And Cellular Mobile Telephone Service Regulations, 2009 (7 of 2009). This main regulation has been amended five times so far. Amendments made time to time included revising or redefining existing parameters for network QoS or Customer Service Quality. Sometimes, parameters were also added, if required. Changes in parameters were also required to include requirements of new radio access technologies introduced in the mobile networks.

1.1.4 Last amendment i.e. fifth amendment to the regulations was issued on 18th August 2017 which redefined QoS framework for Drop Call Rate (DCR) assessment. The new framework introduced DCR benchmarks on the basis of spatial and temporal distribution of DCR values. The change in assessment methodology was to move from average based approach to a percentile based approach. New QoS framework for DCR assessment is to measure and reflect a more realistic and granular picture of the networks performance. Spatial distribution and Temporal distribution of DCR provides insight into area-to-area variations of DCR and day-to-day variations of DCR. $Q_{SD}(90, 90) \leq 2\%$ and $Q_{TD}(97, 90) \leq 3\%$ were set as benchmarks for DCR parameters to represent performance of network for most of the cells in the network and on most of the days.

1.1.5 It was also decided that DCR benchmark should be technology agnostic. Accordingly, from a DCR benchmark and measurement perspective, a networks performance on all technologies (e.g. 2G, 3G or 4G) deployed by the TSP in a LSA and used for providing voice services are treated equally. After this amendment, the consolidated data of all the cells of the TSP is to be assessed as part of a single network, irrespective of the technology being used i.e. whether cells belongs to GSM, CDMA, WCDMA or LTE.

1.1.6 Similarly parameters other than DCR like Accessibility (Call Setup Success Rate and Session Establishment Success Rate, Channel/ Bearer Congestion), Retainability parameter of Connection with good voice quality are also computed in a technology agnostic manner and these parameters includes LTE radio access bearers, Voice over LTE (VoLTE) in addition to voice and network performance for 2G and 3G networks. While computation of QoS parameters for all RATs may be consolidated as a single data set, there may be technology specific terms and thresholds for considering samples as good or bad. For example, RxQual, Ec/No, RSRQ/ SINR parameters represent quality of Radio signal for GSM, WCDMA and LTE networks respectively. Corresponding values of parameters for different technologies may be computed in a technology specific manner but once computed they may become part of single data set. For example, DCR may be computed with counter values pegged at different network nodes and in scenarios which are technology specific but once DCR is computed it is part of single DCR matrix combined for all technologies. DCR benchmarks are applicable on this single data set.

1.1.7 For DCR assessment, VoLTE was also included in the revised QoS regulations and Explanatory Memorandum (EM) to fifth amendment referred to ITU-T Recommendation G.1028 which defines end-to-end quality of service for voice over 4G mobile network. However, detailed deliberations on various issues of QoS specific to voice services for LTE users were not done during earlier consultation. In August 2016, when consultation process was

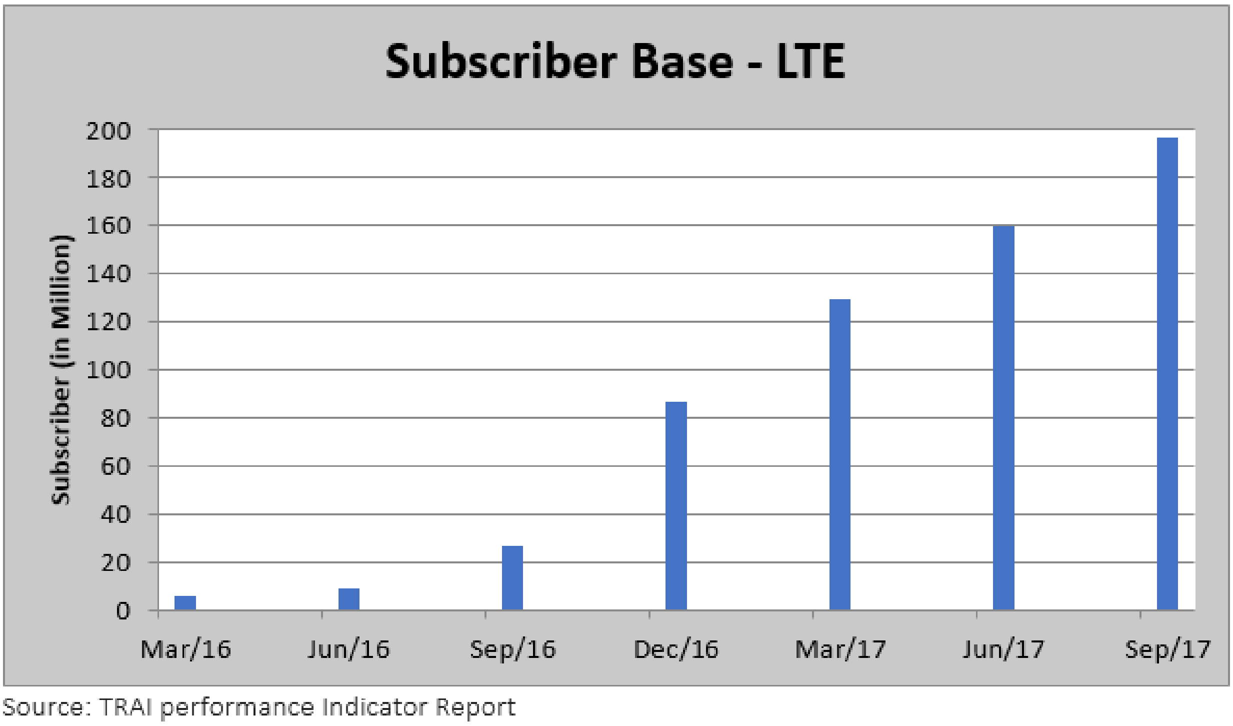

Figure 1.1: LTE Subscriber base in India

initiated to review QoS, LTE networks were not rolled out in many parts of the country and VoLTE was not offered by any of the TSP. LTE network penetration in India has increased significantly in last one year and LTE subscription has crossed 200 million figure. Geographical as well as population coverage of LTE network is increasing day by day, as TSPs are putting up more and more base stations of LTE. First VoLTE launch in India was in September 2016. Many operators providing LTE coverage have yet to offer VoLTE and may launch VoLTE in phased manner instead of pan India launch on day one e.g. launch on city to city basis or LSA to LSA basis. Almost all other operators having LTE networks in India have plans to launch VoLTE services soon.

1.2 Key Differences in Voice over LTE

1.2.1 To provide Voice services to LTE users, there are mainly two types of solutions, one is Voice over IP Multi-Media Sub-System (IMS) and other one is Circuit Switched Fall Back (CSFB). CSFB was introduced in 3GPP release 8 specifications to provide voice services to LTE users without IMS by redirecting user equipment (UE) to 2G or 3G networks in case of Mobile Originated (MO) or Mobile Terminated (MT) voice calls. Voice call using IMS required IMS or VoLTE client by default, either directly native on the chip or as a dedicated application in the UE to interact with network . CSFB also requires functional support on device as well as network side. Initial CSFB functionality as specified in 3GPP release 8 is available in all devices and networks. To improve performance of CSFB, enhancements have been done over subsequent releases. Support of enhanced CSFB features in device and network may be network deployment specific. These may vary from operator to operator and also area to area for same operator. Devices in the network may also be compliant to different 3GPP release.

1.2.2 VoLTE implementation may take different paths based on factors such as spectrum availability, operator voice strategies and market conditions. As operators determine their VoLTE deployment strategy, they may choose an evolutionary approach that starts with Circuit Switched Fallback (CSFB) and, when IMS is deployed, and introduce SR-VCC (Single Radio Voice Call Continuity) once VoLTE is launched prior to ubiquitous LTE coverage. Other operators may choose to achieve ubiquitous LTE coverage prior to launching VoLTE service. The operators choice may depend on local market, technology architecture, spectrum availability and competitive conditions along with business objectives. As a result, there may be some operators who have networks only with LTE and voice services may always be VoLTE based. While there may be other operators who are having multiple RATs in their networks e.g. GSM or CDMA, WCDMA, LTE and they may have option to provide voice services either using CSFB or VoLTE or combination of both.

1.2.3 LTE networks are IP based and it is common protocol for delivering voice and data. Voice and data have quite different characteristics, data in nature is asymmetric and comes in bursts while voice is symmetric and almost regular in periods. Moreover, voice is real time and improvements in error rate by retransmission of packets may not be employed in case of voice if such retransmissions are not completed within time limits applicable for voice services e.g. in less than 50 ms. IP overheads for voice services with vary small packet size may be very high and inefficient for radio technologies. To deliver Voice over LTE and meet stringent requirements of voice, a lot of features in radio and core networks have been developed like Robust Header Compression (RoHC), Transmission Time Interval (TTI) bundling, Semi Persistent Scheduling (SPS) state etc.

1.2.4 However, as in case of Circuit Switch based technologies, packet based technologies may not have dedicated path for voice packets. Voice is a real-time service with tight delay requirements thus it requires a robust underlying radio network to ensure an optimal user experience. VoLTE requires end-to-end QoS to be supported at all layers from device through the radio network up to the core network and including interaction with the IMS core. In case of VoLTE, variability or changes in the delay (jitter) becomes more critical issue than fixed delay. For VoLTE, framing and queuing is different than CS voice as there may be delay in transmission of speech. Audio is transmitted in a packet stream using packet switched network and affected by variations in transport networks. VoLTE receivers use jitter buffer and fill buffers before decoding. There is tradeoff with the size of jitter buffer, large jitter buffer deal with high varying delay but may add to absolute values of delay. On other hand, very short jiter buffer there is higher probability of buffer underrun and it may introduce gaps in audio decoding which may lead to silence or voice mute periods. VOLTE devices are also using sophisticated strategies to maintain audio quality during buffer underrun. VoLTE client can insert buffer underrun gaps during natural silence periods (extending speech pauses) or repeat packets if no natural silence periods are available (speech information stretched). This may affect spectral distribution and temporal structure of voice stream. So strategies to mask uncompensated delays by smart processing should be appropriate one. Time warping with pitch preservation may be acceptable to user. Inappropriate processing or no processing at all may also affect quality perceived by the user.

1.2.5 In case of packet swithced network, there is another difference from circuit switched network that error or loss free and timely delivery of voice packets by radio networks may not be enough to ensure end to end performance. In packet switched network, IP packets might get lost or delayed in core or transport networks. Even during handover phase, voice packets yet to be delivered might be required to be handled appropriately. Delay or variations in dealy in delivery of packets or loss of packets or errors in packets may result into poor QoS experience for VoLTE users. Such instances may result into silence period or voice mute observed by the users. Extent to which this is experienced by the user during a call before dropping of call may also be implementation specific. Need was felt to deliberate and consult on the issue to capture occurrence of dropped call appropriately.

1.2.6 2G snd 3G based radio technologies and devices came in market with voice capabilities from day one and voice services to users camped on these networks could have been provided by the same network on which they were camped on. Though, operators had choice to make service based handover for load balancing or serve particular type of service with preferred network. However, in case of LTE, situation is different and on networks side as well as device side, voice service may not be supported over LTE. For purpose of voice, user may be required to be pushed to other technologies i.e. fall back to Circuit Switched technologies (CSFB) which may be bit more time taking process than usual call set up time and may affect QoS experience of user. Actual impact may also be network deployment specific and support of latest features in the devices.

1.2.7 Readiness of all LTE devices to support interoperable VoLTE client with serving LTE network may take time. Interoperability issues may be operator specific and even if operator has launched VoLTE in the area, VoLTE may not be available to all types of LTE device users. LTE users having devices interoperability with IMS of that operator may be served using VoLTE while other LTE users may be served via CSFB for voice services. QoS for voice services assessed for a particular LTE network may vary for VoLTE and CSFB users. There might be some issues which may be observed more in cases of CSFB e.g. delay in call set up time, registering incoming call attempt as miss call by phone while there was no alert to user most probably due to improper behaviour of network or incomplete execution of call set up procedure involving mutiple types of radio networks.

1.3 Scope of Consultation

1.3.1 In view of above issues related to voice service to LTE users either via VoLTE or CSFB, need was felt to consult and deliberate on issues to identify QoS KPI or measurement procedure. This consultation is focussed and limited to QoS issues related to Voice services to LTE users e.g. silence or voice mute for VoLTE users and voice call related issues experienced by users served via CSFB and capture them appropriately in the DCR statistics available from the networks or during drive tests. DCR assessment methodology and other parameters would remain same as prescribed in the fifth amendment to the QoS regulations.

1.3.2 Issues of VoLTE and CSFB are deliberated in next three chapters of the consultation paper in more detail and raise specific issues for consultation:Chapter 2 deals with voice service for LTE users via VoLTE. It covers the issues or concerns observed related to VoLTE and attempt to analyse the probable reasons for the same. In the end, it discusses about possible Key Performance Indicator (KPIs) which can be measured and reported upon to improve the perceived quality of voice services by consumers.Chapter 3 deals with the issues or concerns related to voice service for LTE users via CSFB. It covers the issues or concerns observed related to CSFB and attempt to analyse the probable reasons for the same. In the end, it discusses about possible Key Performance Indicator (KPIs) which can be measured and reported upon to improve the perceived quality of voice services by consumers. Chapter 4 covers the issues for consultation and seek suggestions and inputs from all the stakeholders.

Chapter 2

Voice over LTE

2.1 VoLTE: Voice over IMS

2.1.1 VoLTE is not simply a voice service to serve users over packet switched networks instead of circuit switched networks, so that both voice and data can be provided by common IP based networks (such as LTE networks). It has much more capabilities to evolve with. It takes advantage of flexibility of packet based services, dynamic allocation of larger bandwidth by the radio networks, capabilities of Session Initiation Protocol (SIP) to negotiate the codec type, etc. VoLTE can also provide HD voice and enriched communication with AMR-WB codec.

2.1.2 To reduce complexity and specify a minimum feature set, GSMA defined profile for Voice over IMS (MMTel) solution, which was christened as VoLTE. This is documented in GSMA Permanent Reference Document (PRD) IR.92. This document also profiles capabilities to support the CS-IMS voice transition referred to as IMS Centralized Services (ICS) and provides service consistency for all mobile subscribers. The service continuity may also be fulfilled by Single Radio Voice Call Continuity (SR-VCC) or its enhanced version, eS-RVCC which provides service continuity for subscribers who are on a VoLTE call but move out of LTE coverage. VoLTE deployment standard has now become an industrial reference standard for VoLTE deployment.

2.1.3 3GPP introduced SRVCC feature to support seamless handovers between PS based voice calls and CS based voice calls. SRVCC combines optimized handovers defined between LTE and legacy 2G/3G networks and Voice Call Continuity defined in IMS core network. SRVCC requires UE to support the ability to transmit and receive on two networks (PS-based and CS-based) simultaneously. SRVCC specifications have evolved to reduce voice interruption time that impacts the user experience and enhanced version of SRVCC is known as eSRVCC and defined in 3GPP release 10. aSRVCC was introduced for PS to CS SRVCC access transfer of call in the Alerting phase and bSRVCC (release 12) for before Alerting phase. rSRVCC in release 11 is also introduced for Reverese SRVCC from GERAN/UTRAN to E-UTRAN.

2.1.4 GSMA PRD IR.92 provides the basis for the definition of other services such as GSMA PRD IR.94 as IMS Profile for Conversational Video Service, as well as GSMA PRD IR.58 as IMS Profile for Voice over HSPA (VoHSPA). Similar to GSMA PRD IR.92, PRD IR.94 defines a minimum mandatory set of features required to implement and guarantee an interoperable, high-quality IMS-based conversational video service over LTE, commonly referred as Video over LTE (ViLTE).

2.2 VoLTE compatibility issue among 4G devices

2.2.1 On one hand, VoLTE has all the advantages of flexibility and evolution to provide better and more enriched communication services using IP and SIP, on the other hand it may suffer from variable latency and possible IP impairments or interoperability issues. These impairments can come from any parts of the chain of sub-systems beginning from radio network to Application Servers including intermediate subs-systems like packet core network, IMS, transport networks etc. VoLTE interoperability issues might be related to EPS bearer set up, proprietary simplification of SIP call flows, operator specific IP headers, different security configurations IPsec, AKA etc.

2.2.2 Circuit Switched Voice tightly specified based on 3GPP Non Access Stratum (NAS) protocols for Call Control and Mobility Management and included state machines and timers. Errors and failures case were also well specified while in case of implementation of VoLTE, there may be set of issues in which specific behaviour may differ device to device. Even UICC interoperability may not be considered for all cases in VoLTE due to different VoLTE dialects not interoperable with each other. Operators may offer downloadable IMS client for devices but it may not be fully equivalent to embedded IMS client in all aspects.

2.3 Issues/concerns for voice service via VoLTE

2.3.1 In case of VoLTE, call set up delay may be estimated from SIP INVITE to SIP 180-Ringing messages. KPIs impacting VoLTE call set up latency may be involving stages of SIP INVITE or Session Start to Dedicated Bearer Set up, dedicated Bearer to Ringing, Conversion Delay after Call Answer, Total Call Set up Delay. Call set up delay in case of

VoLTE may also be dependent upon paging parameters and UE inactivity timers. Aggressive values may impact network resources, UE battery time while relaxed values may increase delay to set up call.

2.3.2 As in case of other mobile technologies, in LTE there may be several reasons of Voice call drop or session drop. In a particular RF situation, whether the session is to be dropped or not may be implementation specific. The drop may be caused by a UE message or by measurements carried out by the eNodeB. Most of the Voice Call drops may be pegged as Radio Induced Drops or Mobility Management Entity (MME) Induced Drops. Similar to other radio access technologies, there are many timers and constants which are used to keep radio link between UE and eNB in sync and retain the connection. Values of these timers and constants may determine the duration for which UE can wait to decode particular frames or maximum attempts it can make to connect or re-establish the radio connection without dropping the call or session. Even the counters which are pegged in instances of drop may be dependent upon the scenario. Differences in case of LTE may be in numbers of counters or timers or number of stages in the procedure. Inappropriate setting of these values may improve counter values for call or session drops or success rates while actual user experience may not be good.

2.4 Optimization of timers and constants in LTE

2.4.1 In LTE, Call drops may be pegged due to radio induced drops, MME induced drops or Handover failures. To conquer these problems, various timers (such as T304, T301, T311 etc) and constants (such as N310, N311 etc) are defined by 3GPP in its documents [3GPP TS 36.331]. Optimization of such timers and constants is very essential from the users experience prospective. some of the constants and timers whose values may impact QoS but improve DCR statistics or call set up success rate are mentioned as per Annexure I. Inappropriate value configuration of such timers and constants in the network may lead to situations like; frequent call drops, longer call set up delay, silence periods or voice muting. In case of voice muting, call may not getting dropped but customers experience a prolonged silence periods during a voice call.

2.4.2 An additional parameter Radio Link Timeout (RLT) for GSM based radio access network was defined in the fifth amendment of regulations. High RLT value i.e. 48 and higher configured in the network for more than three days is required to be recorded with proper justifications.

2.4.3 Similarly, for VoLTE call quality assessment, range of timer and constant values, which affect user experience but not properly reflected in QoS assessment, may be required to be discovered for general network conditions and prescribed appropriately. Inappropriate values may result into longer call set up delay, silence periods or voice muting etc. Certain cases which may require setting values beyond normally permissible limits for longer periods may need proper justification and to be recorded.

2.5 Indicator for VoLTE performance measurement

2.5.1 ITU-T Recommendation G.1028 regarding end-to-end quality of service for voice over 4G mobile network, has defined End-to-end quality indicators and corresponding network KPIs for Voice over 4G mobile network. Some of them are Registration success rate (KPI related to IMS), Post-Dialing Delay (PDD), voice quality (MOS-LQ) and call drop rate, which can reflect the poor customer experience for voice service. It also identified list of degradations encountered by end-users of VoLTE service and their potential causes, as per Annexure II. In other hand, GSMA also considered QoS parameters defined by ITU-T in its recommendation G.1028 and defined the quality of service (QoS) parameters based on field measurements and their computation in its document IR.42. Some of the parameters identified by GSMA for VoLTE are IMS Registration Success Ratio, IMS Third-party Registration Success Ratio, VoLTE MO/MT Session Setup Time, VoLTE speech quality (SpQ MOS-LQO and SpQ R-Factor-LQ), SRVCC (PS-CS) Quality Parameters etc.

2.5.2 Telephony speech quality is an indicator representing the quantification of the end-to-end speech transmission quality of the Mobile Telephony Service. As measurement of call quality is very subjective area, therefore Mean Opinion Score (MSO) used in telecommunications to access the human users opinion of call quality. The validation of the end-to-end quality is made using MOS-LQO scales. These scales describe the opinion of users with speech transmission and its troubles (noise, robot voice, echo, dropouts and so on), according to ITU-T Recommendation P.862/P.863.

2.5.3 ITU-T recommendation P.863 is a mobile voice quality testing standard and also known as Perceptual Objective Listening Quality Analysis (POLQA). It has been especially developed for super wideband (SWB) requirements of HD Voice, 3G, VoLTE (4G), VoHSPA and VoIP (Voice over Internet Protocol). It considers many new conditions which are there in emerging technologies like LTE e.g.,

- New types of speech codecs as used in 3G/LTE and audio codecs, e.g. AAC and MP3

- Voice Enhancement systems, Noise Reductions, Discontinuous transmission (DTX)
- Codecs that modify the audio bandwidth, e.g. SBR (Spectral Band Replication)
- Measurements on signals with very high background noise levels
- Correct modeling of effects caused by variable sound presentation levels
- Support of NB (300 to 3400 Hz) and SWB (50 to 14000 Hz) mode
- Handling of time-scaling and warping as seen in VoIP and 3G packet audio

2.5.4 Further, 3GGP also identified KPI parameters for E-UTRAN [3GPP TS 32.451] and IMS [3GPP TS 32.454] that are equivalent to parameters identified in ITU-T Recommendation G.1028, like Initial Registration Success Rate of S-CSCF, Third Party Registration Success Rate, Call Drop Rate of IMS Sessions, E-RAB Retainability (reflect abnormal releases of the service) etc. QoS monitoring and reporting at critical network points (end-user part/ access part/ core part) can help regulator and service providers to troubleshoot the concerns/issues related to customer experience. In India, LTE is still in evolution stage, but 4G LTE availability rate is growing at a faster pace, which poses the need to closely analyze the problems related to Quality of Services.

Chapter 3

Voice Services for LTE Users via CS Fall Back

3.1 Evolution of CSFB performance over different release of specifications

3.1.1 Circuit Switched Fallback (CSFB) was introduced in 3GPP release 8 specifications to provide voice services to LTE users without IMS by redirecting user equipment (UE) to 2G or 3G networks in case of Mobile Originated (MO) or Mobile Terminated (MT) voice calls. Initial CSFB functionality as specified in 3GPP release 8 is available in all devices and networks. To improve performance of CSFB, enhancements have been done over subsequent releases. Support of enhanced CSFB features in device and network may be network deployment specific. These may vary from operator to operator and also area to area for same operator. Devices in the network may also composed of different releases.

3.1.2 In 3GPP specifications mentions various methods for CSFB to handle voice calls from LTE to other Radio Access Technologies (RATs). First method is Redirection based and radio resources in the LTE networks assigned to UE are released and redirect UE to other Radio Access Technologies (RATs). In this case PS data session may be interrupted until the PS data session in re-initiated in new RAT. Another method is a PS handover-based method where voice call handling may initiate with target RAT and PS data session can remain till time handover is completed. 3GPP later releases than release 8 has number of improvements and flavours in the Redirection method. Summary of key enhancements are tabulated below[1]:

[1]White paper on Circuit- switched fallback by Qualcomm in collaboration with Ericsson 2012

CSFB performance	Release 8 (Release with Redirection-Basic)	Release 8 (Release with Redirection-Skipping)	Release 9 (Release with Redirection-SI tunnelling)
Features	Reads all the System Information message prior to assessing the target cell. Redirection based with SIB skipping Feature: Defer Measurement Control Reading (DMCR)	Only read mandatory System Information and Redirection based with Defer Measurement Control Reading (DMCR) and optimized SIB scheduling	System information is tunnelled from the target Radio Access Network (RAN) via the core network to the source RAN and Redirection based with DMCR and RRC release including a list of 3G cells may be included in the redirection message sent to the device. This can avoid reading any system information on the target cell
Outgoing Call setup time	63-65% more vs. legacy Some System information Block messages can be very large depending on the number of neighbours, which can result in segmentation	22% more vs. legacy	13-14% more vs. legacy
Incoming Call Setup time	96% more vs. legacy	33% more vs. legacy	20% more vs. legacy

3.1.3 Further enhancements include CSFB LTE to UMTS through PS Handover without measurement and with measurement. In such cases, target cell is prepared in advance and device can camp-on the cell directly in connected mode once the voice call is initiated.

3.2 Issues/concerns for voice service via CSFB

3.2.1 Call Set up delay can be derived from the time when Extended Service Request (ESR) message is sent until the Alerting message is received. KPIs impacting CSFB call set up latency may be involving stages of LTE ESR to Redirection of PSHO command and additional time to tune to UMTS, SIB read time including cell selection, UMTS RRC latency, Non-Access Stratum (NAS) end-to-end time and total call set up delay.

3.2.2 In case of CSFB, issues related to call set up delay, call set up success rate and how fast to camp back to LTE after voice call is released. Camping back to LTE after release of voice call may be network deployment specific e.g. cell re-selection procedure may be used or network based fast return to LTE may be deployed. CSFB requires additional steps to set up call and this adds extra delays.

3.2.3 For voice calls of LTE users via CSFB methods, call quality assessment may be required to be assessed and suitable parameters may need to be appropriately defined. For CSFB scenarios, range of parameter values which affect user experience but not properly reflected in QoS assessment may be required to be discovered for general network conditions and prescribed appropriately. Inappropriate values may result into longer call set up delay, silence periods or voice muting etc. Certain cases which may require setting values beyond normally permissible limits for longer periods may need proper justification and to be recorded.

3.3 Indicator for CSFB performance measurement

3.3.1 ITU-T Recommendation G.1028 regarding end-to-end quality of service for voice over 4G mobile network, has defined End-to-end quality indicators and corresponding network KPIs for Voice over 4G mobile network. Some of them are Registration success rate (KPI related to IMS), Post-Dialing Delay (PDD), voice quality (MOS-LQ) and call drop rate, which can reflect the poor customer experience for voice service. GSMA also considered QoS parameters defined by ITU-T in its recommendation G.1028 and defined the quality of service (QoS) parameters based on field measurements and their computation in its document IR.42. Some of the parameters identified by GSMA for CSFB are Setup Time Telephony (equivalent to PDD), CSFB Return to LTE Success Ratio, Speech Quality on Call Basis (SpQ), Call Completion Ratio Circuit Switched Telephony etc.

3.3.2 Post-Dialing Delay (PDD), voice quality (MOS-LQ) and call drop rate may be relevant parameters to identify and quantify quality of service degradation in voice services via CSFB. Post- Dialling Delay (PDD) are defined by ITU as "Time interval (in seconds) between the end of dialling by the caller and the reception back by him of the appropriate ringing tone or recorded announcement". QoS monitoring and reporting at critical network points (end-user part/ access part/ core part) can help regulator and service providers to troubleshoot the concerns/issues related to customer experience. LTE is a all-IP technology and providing voice services via LTE is a big challenge. LTE is still in evolution stage in India, but 4G LTE availability rate is growing at a faster pace, which poses the need to closely analyze the problems related to Quality of Services.

Chapter 4

Issues for Consultation

1. Whether prescribed QoS parameters, as per existing QoS Regulations, are sufficient to effectively monitor QoS of VoLTE/CSFB calls? Please provide suggestions with justifications.

2. If existing QoS parameters are not sufficient to monitor QoS of VoLTE/CSFB calls, then what new parameters can be introduced? Please provide details with justifications.

3. How to define instance of silence/voice mute? How many such instances may be accepted during voice call? Whether existing parameters like packet loss, jitter, latency, end-to-end delay are sufficient to identify or measure silence/voice mute or some other parameters are also need to be factored to measure it? Please provide details with justifications.

4. How to measure report and evaluate network or service from perspective of silence/voice mute problem? Which ITU measurement tools can be used to prepare framework for measurement of silence/voice mute problem? Please provide details with justifications.

5. Whether certain range of timers and constants are required to be prescribed which may affect VoLTE call quality assessment? If yes, which may be those timers and constants and what may be the suggested ranges of timers and constants? Please provide details with justifications.

6. What parameters like Post Dialing Delay (PDD) may be introduced to measure performance of users being served voice via CSFB? What may be the threshold? How to measure report and evaluate? Please provide details with justifications.

7. Any other issue which is relevant to this subject?

ANNEXURE-I

Some timers and constants defined for LTE [3GPP TS 36.331]

1. Timers

SN.	Timer	Start	Stop	At expiry
1	T300	Transmission of RRCConnectionRequest or RRCConnectionResumeRequest	Reception of RRCConnectionSetup, RRCConnectionReject or RRCConnectionResume message, cell reselection and upon abortion of connection establishment by upper layers	Reset MAC, release the MAC configuration and re-establish RLC for all RBs that are established.
2	T301	Transmission of RRCConnectionReestabilshmentRequest	Reception of RRCConnectionReestablishment or RRCConnectionReestablishmentReject message as well as when the selected cell becomes unsuitable	Go to RRC IDLE.
3	T304	Reception of RRCConnectionReconfiguration message including the MobilityControl Info or reception of MobilityFromEUTRACommand message including CellChangeOrder	Criterion for successful completion of handover within E-UTRA, handover to E-UTRA or cell change order is met (the criterion is specified in the target RAT in case of inter-RAT)	In case of cell change order from E-UTRA or intra E-UTRA handover, initiate the RRC connection re-establishment procedure; In case of handover to E-UTRA, perform the actions defined in the specifications applicable for the source RAT

SN.	Timer	Start	Stop	At expiry
4	T310	Upon detecting physical layer problems for the PCell i.e. upon receiving N310 consecutive out-of-sync indications from lower layers	Upon receiving N311 consecutive in-sync indications from lower layers for the PCell, upon triggering the handover procedure and upon initiating the connection re-establishment procedure	Rs. If security is not activated: go to RRC IDLE else: initiate the connection re-establishment procedure.
5	T311	Upon initiating the RRC connection re-establishment procedure	Selection of a suitable E-UTRA cell or a cell using another RAT.	Enter RRC IDLE.

2. Constants

SN.	Constants	Usage
1	N310	Maximum number of consecutive ”out-of-sync” or ”early-out-of-sync” indications for the PCell received from lower layers
2	N311	Maximum number of consecutive ”in-sync” or ”early-in-sync” indications for the PCell received from lower layers

ANNEXURE-II

List of degradations encountered by end-users of VoLTE service and their potential causes [ITU-T G.1028]

1. QoS problem linked to call session performance

SN.	Kind of degradation	Possible reasons	Location
1	Identification Failure	i) Problem with MME, HSS or PCRF	EPC
2	Unavailability of basic call	i) Error in scheduling ii) Radio resource control (RRC) connection set-up failure (reception of RRC connection reject, or expiry of timer T300, no RRC connection set-up complete sent after reception of RRC connection set-up)	eUTRAN
3	Unavailability of basic call	i) Not available due to load (serving gateway (SGW) or packet data network gateway (PGW)) ii) Failed negotiation iii) Reception of several SIP error codes iv) Reception of SIP CANCEL from IMS v) TD internal timer expired, causing a 'SessionSetupFailureTimeout'	EPC
4	High post dialling delay	i)Load ii) Interworking between systems iii) Use of SIP preconditions iv) CS fall back or IMS tromboning at call set-up	All
5	Link failure	i) Bad negotiation between two equipments of the network during call establishment (bad codec management)	eUTRAN/ EPC
6	White call	i)Terminal is not able to code or decode speech while the signalling is OK for the communication	Terminal

SN.	Kind of degradation	Possible reasons	Location
7	Call drop	i) Terminal bug, bad covered area, handover/SRVCC failures due to cells neighbourhood problem, etc ii) RRC connection drop (at reception of RRC connection re-establishment reject, or expiry of timer T301 or in case RRC connection release is received before new RRC connection set-up attempt)	Terminal/ eUTRAN
8	Call drop	i) Link failure: System failure, bad re-negotiation between two equipments of the network during call ii)Reception of SIP status code 500 (Server Internal Error) iii) No RTP packet received during a period longer than 'SessionDropTimeout' TD internal timer iv) No SIP 200 OK on BYE is received within the time measured by 'SessionHangupTimeout'TD internal timer	EPC

2. QoS problem linked to perceived speech quality

SN.	Kind of degradation	Possible reasons	Location
1	Noise	i) Disturbing comfort noise generation (CNG) due to bad noise reduction ii) Noise due to bad electronic implementation on terminal (e.g., analogue /digital conversion) iii) Disturbing residual noise due to bad noise reduction iv) Background noise (street, car, etc.) v) Additional noise due to eUTRAN configuration problem	Terminal
2	Echo	i) Bad performance of acoustic echo cancellation (AEC)/ No AEC. As reminder: Acoustic echo is the coupling between the loudspeaker and the microphone of the phone terminal	Terminal

SN.	Kind of degradation	Possible reasons	Location
3	Echo	i) Bad performance of electric echo cancellation (EEC)/No EEC . Reminder: Electrical echo is due to digital to analogue transformation for a call between mobile terminal and PSTN (No electrical echo for mobile to mobile call).	Networks
4	Low/high speech level	i) Bad performance of automatic gain control (AGC)/No AGC.	Terminal
5	Encoding / decoding issues	i) Narrowband instead of wideband speech quality ii) Lower WB-AMR bitrate/AMR leading to distortion on speech signal. iii) Many transcodings leading to distortion on speech signal iv) Rebuffering and time scaling causing distortion	Terminal/ eUTRAN
6	Terminal Acoustic	i) Although WB-AMR codec is supported, the acoustical performance of the terminal (on receiving and/or sending side) is not wideband compliant. ii) Not well-balanced acoustic terminal can lead to a sound which seems too aggressive, too muffled, etc. iii) Distortion due to transducers.	Terminal
7	Low/high speech level	i) Bad performance of automatic gain control (AGC)/No AGC.	Terminal
8	Encoding / decoding issues	i) Narrowband instead of wideband speech quality: Remote terminal not WB HO towards 2G Call with PSTN, 2G, mobile platforms, etc. where wideband is not deployed. Interworking with CS 3G not WB	Terminal/ eUTRAN
9	Chopped Conversation	i) Bad VAD/DTX/DRX implementation ii) Problem with voice quality enhancement (VQE) algorithm.	Terminal
10	Chopped Conversation	i) IP packet loss or jitter in network ii) Bad handling of IP packet loss and inter-arrival jitter by jitter buffers or packet loss concealment (PLC) inside terminals	All

SN.	Kind of degradation	Possible reasons	Location
11	RTP/IP packet loss	i) Network congestion (several causes : traffic load, distance from cell centre causing activation of TTI bundling, for instance) ii) Jitter buffers not adapted to actual jitter amount or packet size (can depend on use of RoHC or not)	EPC/ Terminal
12	E2E delay	i) Network load. Media handling (packet construction, jitter buffer management) ii) Speech processing in terminals iii) Random access channel (RACH) upon receiving handover command iv) RACH/contention procedure v) Additional RACH attempts vi)Dynamic scheduling, link adaptation vii) Radio link failure/re-establishment during handover (possibly different cell)	All
13	RTP/IP Desequencing	i) New route after a problem such as congestion	EPC
14	Network Delay Variation (Jitter)	i) Network congestion ii) Jitter buffers not adapted	EPC/ Terminal
15	Radio degradations	i) Limit of the cell coverage ii) Interference iii) Area not well covered (obstacle, etc.) iv) Bad radio optimization v) Radio loss profile vi) Bad radio scheduling vii) No or bad use of hybrid automatic-repeat-request (HARQ) mechanisms	eUTRAN
16	Handover latency	i) Latency due to new route after HO or SRVCC	EPC/CS network

List of Abbreviations

CDMA Code Division Multiple Access.

CSFB Circuit Switched Fallback.

DCR Drop Call Rate.

E-UTRAN Evolved Universal Terrestrial Radio Access Network.

GSM Global System for Mobile.

GSMA GSM Association.

HSPA High-Speed Packet Access.

LSA License Service Area.

LTE Long Term Evolution.

NAS Non-Access Stratum.

QoS Quality of Service.

RAT Radio Access Technology.

RLF Radio Link Failure.

RLT Radio Link Timeout.

RRC Radio Resource Control.

SIP Session Initiation Protocol.

SRVCC Single Radio Voice Call Continuity.

TRAI Telecom Regulatory Authority of India.

TSP Telecom Service Provider.

UTRAN Universal Terrestrial Radio Access Network.

ViLTE Video over LTE.

VoHSPA Voice over HSPA.

VoIP Voice over Internet Protocol.

VoLTE Voice over LTE.

WCDMA Wideband Code Division Multiple Access.