Submission to the Telecom Regulatory Authority of India (TRAI):

Comments on Consultation Paper on Voice Services to LTE users (including VoLTE and CS Fallback)

MARCH 28TH, 2018

SAMSUNG

Table of Contents

SAMSUNG

1. Executive Summary

Samsung Electronics Co., Ltd (hereinafter Samsung) is pleased to submit a response to the TRAI consultation paper on "Voice Services to LTE users (including VoLTE and CS Fallback)"[1]. Samsung is grateful for the opportunity to work with the TRAI for defining the Quality of Service (QoS) Key Performance Indicator (KPI) parameters for LTE voice call services that can be used to benchmark and assess the performance of the service providers network. The TRAI consultation paper has captured the existing QoS parameters quite comprehensively.

In section 2, Samsung provides comments and suggestions on the issue for questions listed in the consultation paper. Samsung fully supports the consultation by the TRAI in defining the additional QoS parameters required while offering satisfactory Voice Services to LTE users by the service providers.

Samsung has proposed new parameters (in addition to existing ones) related to Session Initiation Protocol (SIP) & LTE protocol and made an attempt to define the silence or voice mute. In addition to these, TRAI may also consider studying and adopting GSMA's Modem Diagnostic Monitoring Interface (MDMI) specification and 3rd party metric collection tools used by the prominent service providers in United States to measure the network performance.

Samsung has demonstrated strong leadership in LTE technology through active participation in development of global standards and experience in successful implementation of VoLTE service across the globe, with leading service providers in the United States, Korea, Europe, China, Japan & India.

Finally, Samsung would like to thank the TRAI for the opportunity to comment on the consultation, and look forward to working closely with the TRAI in a continuous manner for evolution of satisfactory voice service to LTE users.

2. Comments and Suggestions

In this section, Samsung provides detailed comments and suggestions for questions listed in Chapter 4 of the consultation paper.

Q1. Whether prescribed QoS parameters, as per existing QoS Regulations, are sufficient to effectively monitor QoS of VoLTE/CSFB calls? Please provide suggestions with justifications.

In our opinion, the existing QoS Regulations are not sufficient to effectively monitor QoS of VoLTE/CSFB calls. Hence we request considering more use cases that are relevant to voice services e.g., Interoperability with Circuit Switched network, Emergency Calls, Supplementary Services as detailed below:

1) Interworking with non-VoLTE network: As some Operators may take time to deploy VoLTE services, if a VoLTE Subscriber is calling a non-VoLTE subscriber, additional considerations for QoS is necessary as the call involves interworking with Gateway and CS Network elements. Different QoS may be applicable even in the case of VoLTE to VoWIFI calls and standalone VoWIFI calls.

---

[1] http://trai.gov.in/sites/default/files/Consultation_Paper_VoLTE_CSFB_26022018.pdf

2) Consideration for Emergency Registration and Calls: Emergency Registration time, Call setup time can be considered.
3) SIP Timers for VoLTE Calls: TIMER A, B, C, Ringing timer, Ring timer, Answer Timer, Request Timer, Ack Timer, DBR timer, CSFB fallback timer, Call Forward timer.
4) Enhanced Voice Services (EVS) is being considered actively to improve Voice Quality in VoLTE Calls.

Q2. 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.

QoS problem linked to call session performance and the call drop possible reasons can be further improved by introducing the below additional parameters that are specified in RFC 3261, "SIP: Session Initiation Protocol" section 17.2.1.

1) SIP TIMER H - Wait time for ACK receipt

SIP Timer	Start	Stop	Expiry
TIMER H	Timer H is set to fire when Server transaction completed state is entered.	ACK received in server transaction completed state.	ACK not received in completed state.

Call Setup procedure (INCOMING CALL):

- Step 1 - UE receives SIP INVITE Packet from NWK for call connection.
- Step 2 - UE responds with 200 OK on user accepting the call connection.
- Step 3 - UE waits for the acknowledgement (ACK) from NWK.
- Step 4 - Call Setup procedure is complete when UE receives ACK.

TIMER H is started is during call setup procedure in Step 3. And if ACK is not received TIMER H gets fired and call drop will occur.

2) SIP Session refresh timeout

SIP Timer	Start	Restart	Expiry
Session refresh , RFC 4028	Session refresh timer started on call setup procedure complete.	Restart on periodic session refresh mechanism.	Failure in session refresh mechanism.

Session Refresh (client) procedure:

- Step 1 – UE completes call setup procedure.
- Step 2 - UE starts Session refresh timer on successful call setup.
- Step 3 - UE send periodic re-INVITE or UPDATE requests to keep the session alive.
- Step 4 - Failure in periodic session refresh request, UE will drop the call on session expiry.

Session Refresh is started on successful completion of call setup. Failure in session refresh mechanism leads to call drop.

3) SIP TIMER B – INVITE transaction timeout timer

SIP Timer	Start	Stop	Expiry
TIMER B, RFC 3261, 17.1.2	Timer B is started on INVITE request.	On receiving Reliable response in calling state.	Failure in receiving response in calling state.

Call Setup procedure:

- Step 1 - UE sends SIP INVITE Packet to NWK for call connection.
- Step 2 - UE starts TIMER B for reliable response from NWK.
- Step 3 – Failure in reliable response to UE leads to call setup failure.

TIMER B is started during call setup procedure in Step 2. Failure in reliable response from NWK leads to call setup failure.

Q3. 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.

Definition for instance of silence/mute and other parameters that can be factored to identify or measure silence/voice mute are suggested below:

Definitions:

1) Speech: Sound which is audible in the human perceivable audio frequency range.
2) Mute : Is a complete silence where comfort noise is also not played
3) Silence: Is the absence of human speech, but comfort noise is played.

Speech Quality:

In Voice call, the speech quality may be degraded due to mutations like Mute, Silence, and distortion and the similarity of the speech quality w.r.t to original source is quantified using Mean Opinion Scores.(POLQA or PESQ). If MOS is high, it means Speech quality is good and resembles the original signal. While, it being poor, implies the speech is distorted.

Interpretation:

1) Mute:

At Uplink, Mute could be due to “microphone off” trigger by the User and thus could be a normal operation and it may result in downlink (listener) receiving silence packets.
While, at the downlink, mute, could be due to absence of RTP packets from the network, device playback issues (eg: Drop in jitter buffers due to late arrival, decoding errors, etc.). If the mute is a resultant of packet unavailability for a continuous period of time (as defined by RTP timeout), the device would terminate the call with reason RTP Timeout/

2) Silence:

At the downlink side, the silence generation could be due to incoming silence frames or due to conceal network jitter, the device introduces silence

Measurements :

1) External Equipment:

MOS Scores can be evaluated using tools from various vendors. They provide an easy way to measure and quantify the device speech quality. However, they cannot identify the source of the problem i.e. if the Mute or silence is due to device or network. Apart, these equipment's being expensive, do no scale out for a mass testing in live Service provider network.

2) Network Assisted:

Enhancing the current RTP feedback mechanism from RTCP to RTCP-Extended Reports (RTCP-XR RFC: 3611) already includes a large number of metrics like Packet loss, Jitter buffer drops, Bursty traffic reports etc). RTCP-XR is employed by User Tier-1 Operators as a mechanism to improve their network & device performance. RTCP-XR is not a requirement with Service Providers in India.

3) Device Assisted:

Certain US Service Providers employ an on-device mechanism to collect the metrics. The device periodically updates there parameters to a Metrics Server which can either be hosted by Service Providers or TRAI. However, such uploads are performance intensive and can impact the device behaviour apart from having privacy implications to the User. Thus, these mechanisms are generally employed on in controlled environments. TRAI may encourage volunteers to download appropriate software and upload relevant parameters on TRAI Matrix server for monitoring.

Q4. 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.

Voice continuity test (constant, continuous signal - for example, Flute play) can be considered for detecting mutes. Voice capture tools can be used for post-test evaluation.

Mechanisms as explained in our response to Q3 may be helpful in detecting intentional (user initiated) or unintentional (device initiated) mute issues.

Q5. 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.

Timers that impact the VoLTE call performance are as below:

1) PDCP discard timer

Timer	Start	Stop	At expiry
PDCP discard timer	Configured by the Network and is started for every PDCP SDU that is received from upper	At PDCP PDU transmission	PDCP PDU discard.

	layers (unless it is set to infinity)		

PDCP Discard timer configured by the Network and is started for every PDCP SDU that is received from upper layers (unless it is set to infinity). If the PDCP packet could not be transmitted before this timer expiry, it will be discarded. If the value is set less, this can lead to frequent RTP/SIP packet losses and resulting in Mute, RTP Timeout. While, if the value is set too high, it will lead to delays and thereby affecting Jitters/latencies of packets end-to-end.

Recommended Range: 150-300 msec

Configurations that impact the VoLTE call performance are as below:

1) CDRX configuration

Considering 3GPP specification 36.331 and 36.321, C-DRX is connected mode DRX parameters configured by NW mostly with the aim of power saving feature. Having large value of C-DRX cycle leads to packet aggregation at modem and delayed transmission (assuming a small value of inactivity timer). This will again lead to increase in packet latency and jitter. Therefore, the overall delay from mouth to speaker is increased even more due to packets waiting for transmission opportunity.

Recommended Configuration:

OnDuration Timer - 2 to 8 sub frames

Inactivity Timer - 2 to 8 sub frames (Range of values provided in order to manage load due to active users and resource scheduling)

DRX Cycle length - 40 ms (Value recommended such that the layer 2 buffering delay and jitter is kept to a minimal value. The speech packet generation happens every 20 ms)

2) HARQ max retransmissions

Considering 3GPP specification 36.331 and 36.321, specifications allows configuration of HARQ max transmission up to 28 times. With very high values, it can lead to delays in voice packets. But if it is too less, it may result in more losses especially in poor radio conditions. Therefore, it is recommended that the maximum allowed number of HARQ max re-transmission of a packet belonging to HARQ process should be less than the 150 ms even in worst radio conditions.

Recommended Configuration:

HARQ max transmission- 5 to 8 counts (value selected such that sufficient retransmissions are accommodated and RLC reordering delay due to missing packets are kept to minimal value)

3) LTE MAC logical Channel configurations

Resource filling or transport block filling by MAC is performed in decreasing order of priority of logical channels. Therefore, it is imperative that the logical channel configuration (eg: logical channel priority, logical channel group etc) of a bearer configured for VoLTE are assigned priority that are higher than that of the ones allocated for other user traffic (typically non-GBR related services)

Q6. What parameters like Post Dialling 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.

No Comments.

Q7. Any other issue which is relevant to this subject?

TRAI may consider study of the standards & tools available for QoS metrics collection. Also to consider LTE-Advanced features for better QoS.

GSMA TS.31 standard Diagnostic Logging describes a standardised method to log modem diagnostic data on a device eliminating the need of physical tool logging. The primary user of the tool is expected to be MNO (mobile network operators).

The specification describes API (application programming interfaces) to retrieve the modem and other components data as part of MDMI (Modem Diagnostic Monitoring Interface). Using the MDMI, the diagnostic application will be able to retrieve information from the components being logged, such as a KPI or a protocol message.

MDMI is not yet implemented or featured as mandatory requirement in in any country. Different flavors of this are commercialized by different carriers across the world for similar intent e.g. North America region is used to collect few of modem related RF parameters like MDMI.

If this is mandated to be implemented in all commercial mobile phones, TRAI can look to get Call drop specific KPI from this interface for report on VoLTE call statistic (e.g. Delay, Jitter and Packet Loss) to aid in VoLTE analysis.

Possible use cases are listed below:

- Report the geolocation of the device and key RF parameters (RSRP, RSSI, SINR etc) to determine network coverage
- Present geo-located events on maps to allow better call drop analysis
- Capture handover statistics to debug handover issues
- Report VoLTE (Voice over LTE) call statistics (e.g. Delay, Jitter and Packet Loss) to aid in VoLTE analysis
- Real time reporting on the device
    - Tx power
    - Cell selection
    - RF parameters like RSRP, PUSCH Tx power etc.

3. Acronyms and Abbreviation

3GPP	3rd Generation Partnership Project
E2E	End to End
LTE	Long-term evolution
MBMS	Multimedia Broadcast Multicast Service
MC	Mission Critical
MDMI	Modem Diagnostic Monitoring Interface
PDN	Packet Data Network
P-GW	PDN Gateway

PLMN	Public land mobile network
PPDR	Public Protection and Disaster Relief
RAN	Radio Access Network
TRAI	Telecom Regulatory Authority of India
TSDSI	Telecommunications Standards Development Society, India
US	United States of America
VPN	Virtual Private Network

4. Contacts

Samsung R&D Institute India - Bangalore

Suresh CHITTURI (s.chitturi@samsung.com)

Samsung R&D Institute India - Bangalore

Srinidhi N (srinidhi.n@samsung.com)