Recommendation on Consultation Paper on Voice Services to LTE users

(including VoLTE and CS Fallback)

1. WHETHER PRESCRIBED QOS PARAMETERS, AS PER EXISTING QOS REGULATIONS, ARE SUFFICIENT TO EFFECTIVELY MONITOR QOS OF VOLTE/CSFB CALLS? PLEASE PROVIDE SUGGESTIONS WITH JUSTIFI-CATIONS.

We feel the present prescribed QoS parameters do not comprehensively cover VoLTE and CSFB calls. There are sufficient QOS parameters and regulations by ITU-T and GSMA to measure QOS of VoLTE/CSFB calls, as also mentioned in the consultation paper, but they do not relate directly to consumer experience.

In VoLTE i.e. voice over packet network, the radio link undergoes real time changes in link quality, resulting in issues which were not witnessed earlier like clipping, call muting. The present QoS parameter are more relevant to CS voice. The CS voice services, where measure of the signaling link gave a good indication of the overall quality of dedicated connection, is not necessarily true in case of VoLTE and CSFB calls. VoLTE call on 4G can move to legacy 3G/ 2G networks by means of SRVCC. CSFB i.e. providing voice to 4G subscriber via CS fallback, all put together complicates the overall scenario.

There are KPIs and parameters to monitor call in different stages, prescribed by ITU-T and GSMA like Registration success rate (KPI related to IMS), Call Setup Time, Post-Dialing Delay (PDD), Success rate, Handover time etc.

Voice is a service layer in the present Packet core Network. The present QoS would need to differ in different service layers and treat each service layer separately to arrive at the QoS for different service layers.

Quality of Experience (QOE) apart from Quality of Service needs to be measured to more closely relate to user experience and shall be ubiquitous across all technologies.

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

VoLTE calls involve different IP-based subsystems on IMS/EPC along with the Radio Networks. Network parameters like RTP packet loss, Speech path delay, Jitter can provide insights into the VoLTE/CSFB calls, and provide insight on the QoS for only a limited portion of the overall system.

We therefore advocate the use of actual voice samples to be collected to calculate the final QoE instead of measuring the QoS of different sub-systems. Given the increase in use of smartphones and subsequent increment in the volume of VoLTE, new mechanism is required to measure the reliability i.e. Voice Quality Experience of the actual connection.

Voice service to 4G subscribers is either being provided via VoLTE of via CS fallback to legacy network. KPIs and parameters recommended by ITU-T and GSMA like Registration success rate (KPI related to IMS), Post-Dialing Delay (PDD), CSFB are Setup Time Telephony (equivalent to PDD), CSFB Return to LTE Success Ratio etc. will give an indication on the overall service but not the Quality of Experience. The complexities of calls moving between different systems and layers needs a simple mechanism to capture actual voice quality experience.

As mentioned in point 1, service level measurements at network level is required. Voice quality experience, across all technologies including VoIP and VoWi-Fi should be measured to capture the experience of the end-user.

A simple methodology which allows to capture real voice samples directly from the user phone to evaluate the Voice Quality Experience is suggested further in the response. This process will allow to capture the actual voice samples of the end-customer and give insights into the voice quality experience for not just VoLTE and CSFB services, but also for voice services like VoIP, VoWi-Fi services in near future.

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.

While there are various network parameters captured at the Network Level like packet loss, jitter, latency, delay etc, they have no direct correlation with the voice silence/ muting as perceived by the end user. QoS issue like silence/voice mute cannot be captured for the VoLTE users or users served by CSFB and have little correlation with the DCR statistics available from the networks contrary to circuit based technologies, where the counter or timer values clearly indicate the quality of RF condition, since the signaling and speech paths are always on connections. It is important to realize that in India though there is network which is VoLTE only, there would also be Networks where VoLTE calls will move to the lower circuit based networks, adding to the complexity.

Certain engineering phones used in drive test can capture parameter like RTP packet loss (packet not received by the mobile), Speech Path Delay, No Speech or Silence, but have proprietary methods to define them. While some measure on msec levels, other look at every sec and classify them. There are no standard definitions in ITU on Mute and Silence either. All the above complications along with interoperability issues do not provide a common methodology to measure the voice quality as perceived by the consumer, who is unaware of the calls moving between technologies and various layers.

Drive test is limited by logistics and collection process, also these engineering phones are few, need expensive licenses and cannot be used across commercial phones. Hence introducing technology specific parameters or other methods may not help in measuring the voice quality experience of an end user.

Thus it is suggested to have a cost effective and easy solution to measure Voice Quality Experience and which can be implemented across most of the commercial phones, where actual speech samples are captured. The real speech samples can confirm the actual duration of mutes or silence. The method proposed can capture these silence for any voice call, independent of technology. The mutes can be captured for a period of 5, 8 or 10 secs and TRAI can arrive at the correct value through further consultations.

The mutes can be referred to as PDU (Proactive Disconnect by User) where lack of speech or mutes, may result in user disconnecting the call.

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.

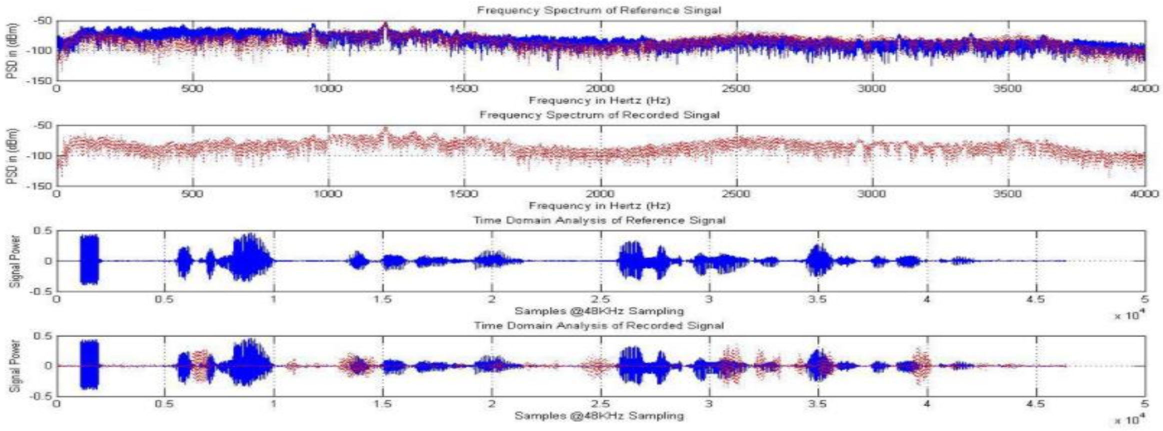
ITU-T recommends standards, POLQA also known as ITU-T Rec. P.863 for capturing the actual voice quality for each call. However the complexities highlighted in point 3 along with other OTT services like VoIP and newer VoWi-Fi makes it difficult to have a common methodology to measure and report silence/ mute at network level.

We suggest a cost effective, unique approach to capture the actual speech samples and measure the voice quality experience in terms of silence/ mute using the proprietary algorithm; which shall capture the actual speech form the test numbers (where predefined speech is played). The approach is independent of all the issues highlighted above and also takes care of other services like VoIP and VoWi-Fi. The suggested alternative process is fool-proof and will have actual samples recorded from the actual consumers against each location.

The approach is to measure the, actual live recorded samples on the smartphones by dialing onto a fixed predefined no. and then processing the results of the recorded speech to check the quality of the call.

The algorithm takes into account different parameters like Amplitude Delay, Coherence, Distortion, to capture the SNR and Mutes. The received or recorded signal is correlated to the reference signal.

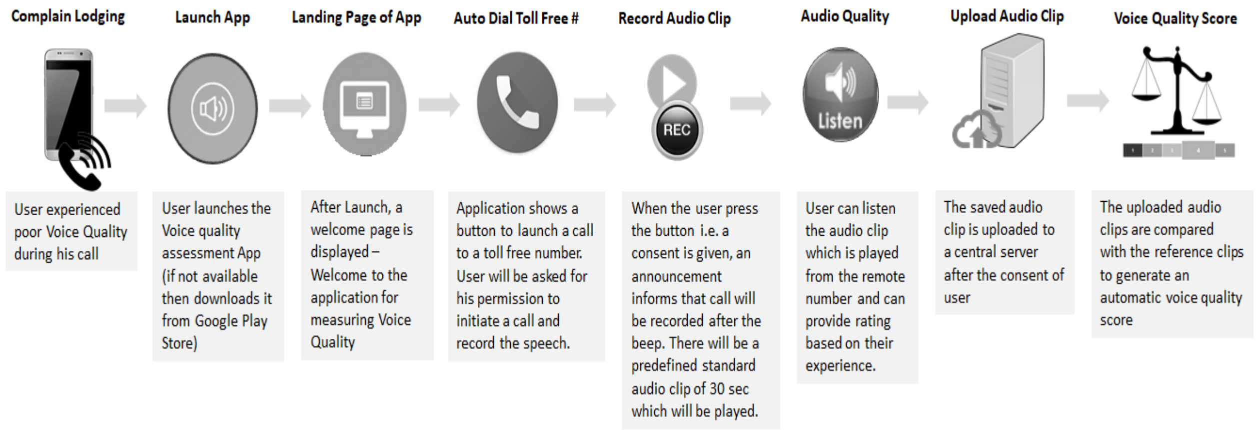
Below is an example of the speech sample which is compared with the reference signal and shows the actual experience captured from the phone.



Spectrograph of recorded and reference sample

Samples which reported mutes are captured along with duration to arrive at the Voice Quality Experience Score.

All these are available in an application which a user can measure from his phone. An end consumer who faces poor voice quality at various locations or while on the go, can get the score of the recorded audio sample which shall indicate the voice quality of their service provider.



Application based process to capture Voice Quality Experience

The above method can be used to capture the actual experience of the user and ensure that the speech samples are recorded and saved in case further analysis is to be conducted.

It allows to capture the actual speech recordings from the live recorded samples and provide the actual user perception of the voice quality i.e. the entire process is "VSMART"

(V) Verifiable – recorded samples are available, and results can be verified whenever required
(S) Specific – muting as observed in the clip is specifically highlighted
(M) Measurable – silence/ mutes in the recorded sample can be measured precisely
(A) Attainable – 100% results can be attained from the recorded samples
(R) Repeatable – every time the samples are checked, same results will be repeated

(T) Time Bound – the duration for which the clip is played is fixed, TRAI can however decide the duration before implementation

The key advantage to the process is:

- Simple and easy to use by consumers
- Solution with Indian IPR hence cost-effective with no license fees.
- A closer simulation of an individual experience of voice quality.
- It is a universal solution across technologies – 2G, 3G, VoLTE and VoIP
- Works on most of the commercial phones and gives a broader base.
- In some instances, it can also help eliminate, in case there are issues with the handsets.

The above process fulfills all the requirements and also takes care of all the complexities introduced by different services and technology.

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.

There are ITU-T and GSMA recommended KPIs on QOS for VoLTE like Registration Success Rate, Post Dialing Delay, Voice Quality and IMS Registration Success Ratio, VoLTE MO/ MT Session Setup Time, VoLTE Speech Quality, SRVCC quality parameters respectively which reflects Accessibility, Retainability, Mobility and Integrity of a call. Appropriate values of certain timers which are associated with Call Accessibility like T300, T301, T310, T311 and constant N310 and N311 also needs to be defined as they affect user perception of a VoLTE call. Inappropriate setting of these values may result in longer wait before a next call can be initiated by a user in case of failure.

Setting of various timers and constants as defined by 3GPP in their documents (3GPP TS 36.331) is network dependent. Optimization of these shall be left to TSPs depending on their network architecture as there are multiple subsystem from Radio network to application server and other technology layers and subsystems involved.

However, all of the above doesn't help in identifying the muting on network level nor they directly correlate with the silence/ muting observed on a VoLTE network.

ITU-T recommended P.863 voice quality testing standard POLQA designed for HD voice and VoLTE and VoIP are not a cost effective solutions nor can be used across various commercial handsets in the market.

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.

GSMA has defined QOS parameters for CSFB like Setup Time Telephony, CSFB return to LTE success ratio, Speech Quality, Call completion ratio circuit switched telephony. Also, for CSFB, the call is redirected to legacy network ones initiated i.e. time taken from ESR (Extended Service Request) to RRC request (on 3G). These shall also be monitored as longer duration of this will hamper user experience of a call.

Monitoring and reporting these at critical network points might help service provider to troubleshoot, however measuring QOE (Quality of Experience) using Voice Listening Quality is equally important to more closely relate to the user experience and troubleshoot the issues related to customer experience.

7. ANY OTHER ISSUE WHICH IS RELEVANT TO THIS SUBJECT?

None