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Ref No: RP/ FY 17 – 18/ 062/ 481

Dated: March 28, 2018

To,
Shri. Asit Kadayan,
Advisor (QoS),
Telecom Regulatory Authority of India,
Mahanagar Door Sanchar Bhawan,
Jawahar Lal Nehru Marg, Old Minto Road,
New Delhi – 110002.

Subject: Consultation Paper on 'Voice Services to LTE users (Including VoLTE and CS Fallback).'

Dear Sir,

This is with reference to your above mentioned consultation paper. In this regard, please find enclosed our response for your kind consideration.

Thanking You,

Yours' Sincerely

For **Bharti Airtel Limited**

A handwritten signature in blue ink, appearing to read 'Ravi P. Gandhi', is written over a horizontal line.

Ravi P. Gandhi
Chief Regulatory Officer

Encl: a.a.

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

We are grateful to the Authority for providing us with the opportunity to submit our response to the consultation paper on “Voice Services to LTE users (including VoLTE and CS Fallback).”

We would like to highlight the fundamental difference between the way a call is processed and handled on VoLTE as against circuit switched 2G/ 3G call (CSFB would technically be a 2G/ 3G call only). Since, the existing QoS norms cover only circuit switched voice, a new QoS regime is required for non-CS calls i.e. VoLTE. This is all the more important due to lack of dedicated resources for real-time voice in case of voice over LTE and thereby, rendering the connection open to severe vulnerabilities of a packet connection, which is on Best Effort basis by design itself.

As it is evident from the underlying transport structure of a packet network, the dependence on quality of flow control, error management and multi-path latency issues are critical for a voice call and minor errors in their functioning may severely impact the voice experience. In a circuit switched scenario for 2G, 3G and CSFB, the dedicated nature of connection pre-empts any such vulnerabilities in the call quality.

As the mobile IP voice blends mobility, IP and real-time communication, it begins with a subscriber’s smartphone. And then, relies on integration with all three layers of the LTE network—access, packet core and IMS—in order to deliver the requisite service. To ensure high service quality to the end users, VoLTE relies on specialized support from both end devices and network architecture, as well as dedicated radio resources during the voice call, to achieve the goal of replacing the legacy call in the long term.

The implementation of VoLTE varies across different devices depending on the OEM’s implementation of VoLTE profiles as compared to standard circuit switched voice for 3G/2G implementation, which eventually impacts the user experience. Parameters & impacts due to these implementations need to be addressed in the VoLTE quality guidelines. TEC, in their whitepaper, published the device-related interoperability & challenges. For details, refer to **Annex A**.

We further state that VoLTE deployment and call reliability are still nowhere in competition with the legacy CS calls. Call failure ratio, setup failure and number of unintended call drops are also higher than that of the legacy CS calls, which is a common problem seen in almost all networks where VoLTE has been launched. This has inevitably resulted into the impairment of user experience. A recent industry report (<http://solutions.amdocs.com/2016SOTR>) states that the frequency of call drops on VoLTE is 4-5 times higher than that of 2G and 3G calls, which is a significant variation given its ability to handle a higher density of calls.

In the backdrop of the above submissions, our detailed response is as follows:

Q.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.

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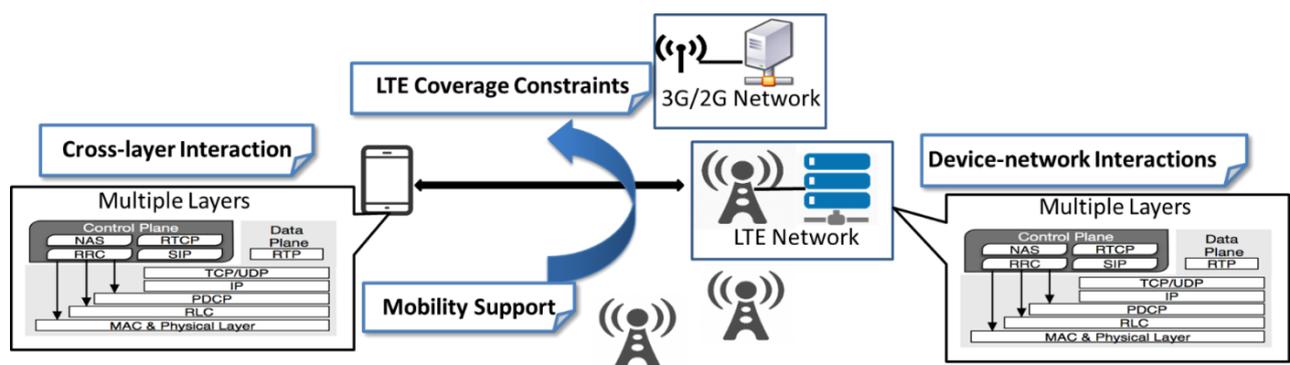
Q.2 If existing QoS parameters are not sufficient to monitor QoS of VoLTE/CSFB calls, then what new parameters can be introduced? Please provide details with justifications.

Bharti Airtel’s Response:

As highlighted above, the existing QoS regulations do not cover the performance of voice calls on a “continuous basis” in a packet switched network. The current QoS parameters only deal with call set up and tear down & hand over performances, which make the framework completely oblivious to the fact that in a NON-DEDICATED, BEST EFFORT connection, every packet during the call phase may render a significantly degraded experience. These indices are currently not covered in the QoS regulation and thereby, dilute the effectiveness of QoS for any voice over packet scenario.

Moreover, the un-acknowledged mode of packet transmission, re-transmissions, jitter and link sanity over radio and backhaul links lead to experiences like call muting, garbled voice, media gaps, packet loss, etc.

Some of these call drops may occur due to a failure in handover between LTE cells, or a failure to correctly implement fallback where LTE coverage becomes unavailable or there is a lack of coordination between device-originated and network-originated events.



Considering the above, the quality of VoLTE calls needs additional support from factors as stated below:

- **IP Packet Loss (uplink and downlink)** - ITU-T Rec. Y.1540 defines IP packet loss as “ratio of total number of lost packets to the total number of transmitted IP packets in a given measurement”. As VoLTE is the BEST EFFORT, we need to take the IP packet loss

rate into consideration in order to analyse the quality of the services provided by any TSP.

- **Block Error Rate (uplink & downlink)** - 3GPP TS 34.121 defines block error rate as “erroneous transport block over total received transport blocks.” It should be measured both for uplink and downlink links for user plane traffic.
- **Garbled Voice Packets** - Out of sequence or erroneous packets may form in VoLTE technology as it is packet-oriented. Hence, KPI should be formulated keeping these Garbled voice packets in mind.
- **Jitter** - As per ITU-T Rec Y.1540, Jitter is defined as the difference between the one-way delay of IP packet and reference IP packet transfer delay (e.g. average IP time Delay as a reference delay). For real-time services, delay jitter poses a significant threat to the service quality and also, the conversational quality. Hence, it is necessary to incorporate the same for quality measurement.
- **Mute calls** - A call is said to be a mute call whenever there is consecutive loss of RTP packets for a period of N seconds.
- **Mean Opinion Score (MOS)** - As per ITU- T P.800, it is a subjective measurement of voice speech quality on a scale of 1 to 5, where 1 corresponds to the lowest quality and 5 is the highest quality experienced by the end user.
- **Dropped call** - No. of abnormally terminated calls

Apart from the aforesaid KPIs for VoLTE call quality assessment, the following additional KPIs should be measured for end-to-end VoLTE user experience assessment.

- **Call setup success rate** - Ratio of successful calls established to the total call attempts for VoLTE bearer QCI 1. RRC congestion based on VOLTE QCI cannot be measured separately for VoLTE calls. In that case, the E-RAB establishment success rate can be taken into account for measuring the accessibility of VoLTE.
- **Call setup time** - Time interval from the moment a VoLTE user initiates the call until the user receives notification that the called party has been alerted
- **Post dial delay** - SIP session setup time which is the time interval between the INVITE and the ACK messages by the originator side
- **Registration success rate** - No. of successfully performed initial registration procedures of S-CSCF to the no. of attempted initial registration procedures

- **Registration delay** - Consists of SIP messages transfer delay, queuing delay in IMS and authentication delay of interaction of HSS with I-CSCF and S-CSCF
- **S-CSCF identification success rate** - Defined as the probability of being able to successfully register into S-CSCF initially
- **S-CSCF identification delay** - Time taken for the successful registration into S-CSCF
- **Location information request success ratio**- Ratio of successful location -info-answers (LIA) received from HSS to the total location-info-requests (LIR)
- **Number of location information requests** - No. of location-info-requests (LIR)
- **TAS INVITE success rate** - No. of successfully performed invite procedures to TAS(3rd party Application server) to the no. of attempted invite procedures
- **Number of INVITES at TAS** - No. of attempted invite procedures to the TAS
- **TAS UPDATE success rate** - No. of successfully performed invite procedures to TAS (3rd party Application server) to the no. of attempted invite procedures
- **Number of INVITES at P-CSCF, I-CSCF, S-CSCF** - No. of attempted invite procedure to the P-CSCF, I-CSCF and S-CSCF
- **Number of updates at P-CSCF**- No. of attempted update procedures to the P-CSCF
- **Number of concurrent RTP Streams** - Device to send multiple RTP streams simultaneously in a single RTP session
- **H.248 number of input octets per call** - It defines an event to detect the presence of voice/data for VoLTE session for compression.
- **IMS- number of UPDATES at S-CSCF** - No. of attempted update procedures to the S-CSCF
- **IMS- number of UPDATES at TAS** - No. of attempted update procedures to the TAS
- **Out-of-sequence packets**- No. of out of sequence RTP packets

QoS measurement for CSFB calls

The CSFB calls are same as 2G/3G calls. Hence, all parameters are captured in legacy technologies like 2G/ 3G and no separate KPIs for CSFB are needed.

Q.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.

Bharti Airtel's Response:

As defined earlier, VoLTE is a packet switched technology compared to circuit switched voice in 3G/2G. Hence, factors like silence / mute / garbled voice affect its user experience critically. Discontinuous nature of packet transmission across radio, transport & core network leads to higher packet loss/drops, thus impairing the voice quality. Therefore, the biggest impact on voice over the packet scenario would not come from drops but from the fact that during the conversation phase, the fidelity of call quality would be of the foremost importance.

Users experiencing lower voice quality due to these parameters would disconnect the calls. Such calls show up in the system as "*normal call release*", rather than "*Call drops*".

A call is said to be a mute call whenever there is a consecutive loss of RTP packets for a period of N seconds. Globally, a mute call for voice over IP network is designated as consecutive packet loss for a period of 2 secs or higher.

Countries like Austria use application probes for Jitter measurement. In France, Voice Quality is measured by the TSP and reported to the authority. Many countries like Austria, Germany, Mexico and Norway have built regulations on probe-based methodologies to measure latency and jitter in various access networks.

Globally, the following parameters are measured to assess the call quality in an IP network:

- IP packet loss rate (uplink and downlink)
- BLER (uplink & downlink)
- Garbled voice packets
- Jitter
- Mute call
- Mean Opinion Score

Q.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.

Bharti Airtel's Response:

An active measurement of Silent or Mute calls can be done by using the network probes on multiple network interfaces including (but not limited to) the following:

- S1-U, S1-MME, Sv
- Mw / Gm/Mg/Mj/Mr
- Rx/Isc/Cx/Iq/Sh/Ut
- IP transport network interfaces
- OSS subsystem parameters for call drop measurement

Due to lack of standard profile for VoLTE devices, there are device related issues, which need to be segregated in order to capture the impact on the network KPI's performance and end user experience.

Q.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.

Bharti Airtel's Response:

The Media Inactivity Timer (MIT) is the critical timer that influences the dropped call rate in VoLTE. Unlike, RLT in 2G/3G, MIT is typically not configured in the system and at best, it is implemented based on the device. Different devices have different MIT values and hence, call drop would depend on the device implementation in VoLTE instead on the network. Therefore, a higher value of MIT would lead to users disconnecting the call and hence, such calls would be pegged as "regular calls."

Current implementation of MIT as measured in selected sample of devices varies from 10 secs to 35 secs. This is significantly higher than the RLT in 3G/2G network, which varies from 2secs-22secs, with TRAI's guidelines of implementation of <22 secs for the circuit switched calls in 3G/2G networks. This puts VoLTE in contrast as device & network would not release the call even up to 35 secs in case of no media activity, and the call would most likely be terminated by the end user. This again would be pegged as "normal call release" as compared to "drop call."

In view of the above, it is recommended that critical parameters like MIT should be standardized in device and network implementation. Such implementation should be governed through device certification program (as the minimum set of testing proposed for deployment in India).

Q.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.

Bharti Airtel's Response:

For VoLTE calls, Post Dialing Delay (PDD) is equivalent to the SIP session setup time, which is the time interval between INVITE and ACK messages by the originator side. It is equivalent to the call setup time as defined in ITU-T E.800. It can be measured either using probes or from IMS.

But this KPI is only one KPI and the exhaustive list is enumerated in our responses above, which impacts Customer experience. The global report (as shared with earlier evidence) states that frequency of call drops on VoLTE is 4-5 times higher than traditional CS voice call in 2G/3G.

Q.7 Any other issue which is relevant to this subject?**Bharti Airtel's Response:**

There are few aspects that the authority needs to look after while measuring VoLTE's performance indicators.

A decrease in the Average Call Duration (also known as Average Length of Call - ALOC) or the increase in the number or percentage of RTP streams with 1-way audio may indicate poor voice quality because callers are hanging up and redialing in hopes of securing a better voice quality. It is therefore important to cross reference signaling KPIs with the corresponding media KPIs for the same subscriber sessions.

Another aspect is the way that small cells are deployed especially for VoLTE. In these environments, small cells are densely deployed, creating border interference that results in increased delay and jitter (the variance in delay). Dense small cell deployment also creates frequent handovers as users move about within a building and generate signaling load on the mobile core network. When a user on a VoLTE call moves along a cell border, ping-ponging may occur, causing handovers to be delayed, which results in even more interference, further degrading the voice quality. The KPIs for small cells need to be defined separately.

Also, the VoLTE traffic is dependent on the device readiness & software upgrade by OEMs for enabling VoLTE. The "4G only" service provider would have 100% of the calls on VoLTE, while for multi-technology operators, percentage traffic on VoLTE is low. As a consequence, the low call volume cells are present predominantly in VoLTE cells. Hence, we would request the Authority to allow exclusion of all Low Call Volume cells (having less than 150 calls in a busy hour) from any regulation mandated for VoLTE measurement.