Experimental evaluation of WebRTC voice quality in LTE coverage tests

Najmeddine Majed[1,2], Stéphane Ragot[1], Xavier Lagrange[2], Alberto Blanc[2], Jérôme Dufour*, Guillaume Grao[1],
[1]Orange Labs, Lannion, France [2]IMT Atlantique, Rennes, France

E-mail:{najmeddine.majed, stephane.ragot}@orange.com, {xavier.lagrange, alberto.blanc}@imt-atlantique.fr

***Abstract*—In this paper, we present experimental results on WebRTC voice quality as a function of LTE radio coverage. Different LTE radio conditions were tested by varying the radio path loss (with fast fading) in controlled test lab conditions. Voice quality was evaluated at different speech coding bit rates in terms of POLQA score and mouth-to-ear delay. Results show that, with acknowledged mode (AM) data transmission over LTE, degraded radio coverage translate essentially into increased end-to-end delay and jitter, with virtually no packet loss up to the coverage limit. Voice quality is analyzed by studying the influence of the specific jitter buffer used in WebRTC endpoints.**

***Keywords*—WebRTC; LTE; MOS; delay; jitter buffer**

I. Introduction

Mobile operators are now deploying mobile telephony services over IP networks, such as voice over Long-Term Evolution (VoLTE) or voice over Wifi (VoWifi). Unlike mobile networks from previous generations that used circuit-switching (CS) for voice and packet-switching for data, these new voice services rely on Voice over IP (VoIP) in radio access and core networks. Moreover, mobile operators now face an increased competition from Over The Top (OTT) players that can take advantage from enhanced data networks (e.g. LTE) and service evolutions (e.g. alternative dialers replacing the default phone app in Android). In particular, new services are emerging based on the WebRTC (Web Real-Time Communications) technology. In this competitive environment, quality of experience (QoE) has become a major issue to ensure end users get the best possible call quality in various network conditions.

QoE is affected by many factors [1], [2], including speech quality, service availability, cost, security, etc. In this work we limit ourselves to the speech quality dimension, which is characterized by different metrics, such as Mean Opinion Score (MOS), mouth-to-ear delay, perceived loudness, frequency spectrum, etc. A complete review of quality metrics in VoIP can be found in ITU-T G.1020 [3] and G.1021 [4], including network, terminal and overall metrics. These specifications also deal with models of de-jitter buffer (or "jitter buffer" for short). The impact of jitter buffer adaptation to network condition has been studied, for instance, in [5]–[9].

The objective of this work is to study speech quality of WebRTC over LTE radio link. In LTE network quality of service (QoS) is applied to a set of bearers, which are virtual transport channels with a set of network configurations (bearer type, priority, packet delay, packet loss) to differentiate traffic classes. On one hand, VoLTE is transported over dedicated Guaranteed Bit-Rate (GBR) bearers with Unacknowledged Mode (UM) data transmission at the LTE radio link control (RLC) layer and specific QoS guarantees (for 98% of IP packets): packet delay budget for each LTE radio leg $\leq$ 80 ms and packet loss $\leq 10^{-2}$. On the other hand, WebRTC media is typically transported with the RLC Acknowledged Mode (AM) over the Non-GBR default bearer, which is used for data traffic over Internet and has the following QoS settings: packet delay budget for each LTE radio leg $\leq$ 300 ms, packet loss rate $\leq 10^{-6}$. To make it more tractable, this study is limited to the case of radio coverage tests in reproducible and controlled lab conditions (e.g. one mobile phone per radio cell). Aspects such as network congestion and multi-user environments or performance in live network conditions (drive tests) are not taken into account and out of scope. To be specific, we measure and analyze speech quality and mouth to ear delay for various LTE coverage conditions and codec bit rates.

Ideally, subjective tests (especially conversational tests) are required to evaluate QoE under different network conditions [10]. However, these types of tests are costly and difficult to conduct. For speech quality measurement, intrusive objective tools are typically used. In this work, Perceptual Objective Listening Quality Assessment (POLQA) [11] was used to assess voice quality in terms of MOS-LQO (Listening Quality Objective) together with a measurement of mouth-to-ear delay between mobile phones. A lot of efforts have been devoted to evaluate objectively VoIP quality, as in [12], [13]. Tests have often been conducted through simulations without a real VoIP system implementation. The evaluation of WebRTC voice quality has received less attention in the literature [9], [14]. In particular, an evaluation of WebRTC quality over LTE has been reported using simulations in [14]. To better represent the real usage context of end users and better evaluate QoE, it is preferable to rely on real system implementations (mobile phones and networks). In this work we report some metrics representing WebRTC speech quality over LTE radio link based on radio coverage tests.

This paper is organized as follows. In Section II, we review briefly WebRTC and the updates made to the Chromium/AppRTC source code. In Section III, we present the experimental setup used to measure voice quality. In Sections IV and V, we report results in good and degraded coverage conditions, respectively. Speech quality is analyzed in Section VI by studying the influence of the specific jitter buffer used in WebRTC end points, before concluding in Section VII.

*J. Dufour was a contractor at Orange when this work was done.

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II. WebRTC review and modifications to Chromium/AppRTC source code

WebRTC [15], [16] is a technology specified by the World Wide Web Consortium (W3C) and the Internet Engineering Task Force (IETF) to provide real-time communication capabilities to media-capable end points (e.g. browsers, native applications). It has been developed to enable communications with only few lines of JavaScript code, without any plugins, and it is supported in browsers such as Chrome, Firefox, Opera [17]. In particular, W3C has defined several JavaScript APIs: getUserMedia (MediaStream) to handle media capture/rendering, RTCPeerConnection which is the most complex API and handles peer network connections, RTCDataChannel for data exchange (other than audio/video streams). IETF has defined a set of protocols to exchange data (voice, video, text, etc.) in peer-to-peer mode, including NAT traversal protocols with ICE (Interactive Connectivity Establishment). A key issue is that the signalling protocol between end points is not fully specified and left to service providers. The only signalling constraint is to rely on JavaScript Session Establishment Protocol (JSEP) [18] which makes use of Session Description Protocol (SDP) to exchange media capabilities and other parameters (e.g. ICE candidates).

In this work, we used a native application called AppRTC, which is provided as a demo app in the open-source WebRTC Chromium project [19]. WebRTC defines two voice codecs that are mandatory-to-implement: OPUS and ITU-T G.711. Because this work has been conducted in a larger context to compare the coverage of VoLTE and WebRTC in LTE radio access, if possible with the same voice codec, the source code of the WebRTC library from Chromium (also used in AppRTC) has been modified to include the support of an additional codec in the audio coding module: Enhanced Voice Services (EVS) [20]. The EVS codec has been standardized by 3GPP in September 2014 to provide new functionalities and improvements for mobile communication, including:

- Enhanced quality and coding efficiency for narrowband (NB) and wideband (WB) speech services.
- Enhanced quality by the introduction of superwideband (SWB) and fullband (FB) speech.
- Enhanced quality for mixed content and music in conversational applications.
- Robustness to packet loss and delay jitter.
- Backward compatibility to the 3GPP AMR-WB codec.

The following modifications were applied to Chromium's WebRTC library. Encryption (DTLS/SRTP) was disabled to allow decoding captured RTP streams. The EVS encoder and decoder were integrated in WebRTC's audio coding module and added to the codec database by replicating the existing integration of the OPUS encoder and decoder; due to this replication, one may consider that the results presented hereafter reflect the audio processing that is also executed with OPUS, even if tests have been conducted with EVS. The audio path was sampled at 48 kHz; note that the EVS codec was constrained to operate in SWB mode at 9.6 kbit/s or above. The RTP payload format for EVS was implemented; experiments were conducted with specific SDP settings (`ptime=20, hf-only=1, cmr=1`), hence 2 header bytes were appended to speech data in each 20 ms RTP packet.

III. Experimental setup

In this work, we used an LTE/Evolved Packet Core (EPC) network that supports IP Multimedia Subsystem (IMS), with an access to the Internet to make sure that WebRTC voice calls could be properly established. The test platform used in this work is based on with commercial radio equipments found in live networks. Two mobile phones were connected to two different LTE base stations (called "eNodeBs") with LTE radio cell in the 2.6 GHz band (20 MHz bandwidth).

The mobile phones under test were radio isolated in separate RF shielded boxes, in which a radio antenna was placed to provide LTE radio access. One of the generated radio signals was attenuated using a fast fading generator as shown in Fig.1. In radio propagation, fast fading presents the effects of the rapid variation of radio channel characteristics in time compared with the duration of data symbol. The EPA (Extended pedestrian A) 3km/h multipath channel model was used. The radio signal level is defined in terms of reference signal received power (RSRP) [21] as the linear average over the power contributions of the resource elements that carry cell-specific reference signals within the considered measurement frequency bandwidth. RSRP measurement (in dBm) is used mainly to rank different candidate cells in accordance with their signal strength. The RSRP level is translated to radio pathloss level expressed in dB.

In this work we focused on the lowest three bit rates of the EVS codec in Super-Wideband (SWB) mode: 9.6, 13.2, and 24.4 kbit/s. A computer operating a voice quality measurement system (MultiDSLA) [22] was used to conduct testing, collect and analyze test data. Voice quality was measured using ITU-T P.863 (POLQA v2.4) [11]. The two phones were traced by a network sniffer called (QXDM) [23] to capture the incoming and outgoing data traffic for further analysis. Note that video was deactivated for all tests to evaluate speech-only communications. Voice transmission was only in one direction (from Mobile1 to Mobile2), silence was injected in the other direction.

An 8-second test sequence consisting of one 4-second female sentence and one 4-second male sentence in French language was used; this sequence is compliant with P.863.1 [24] and it has been selected for drive tests and verified to provide POLQA results close to average scores obtained over a larger number of male and female sentence pairs. Results are reported based on the average of 20 measurements in given pathloss conditions (i.e. 20 repeats of the 8s sentence pair).

IV. MOS and delay in good network condition

Tests were carried out under good LTE coverage conditions. The RSRP level was around -90 dBm corresponding toa pathloss of 114 dB. The measured quality (MOS-LQO$_s$) increases with bit rates, as shown in Table I. It should be noted that the mobile phones were used in headset mode (with an audio input / output from the electrical jack interface). The headset jack interface and other acoustic characteristics of the phones (e.g. the frequency response in sending / receiving) as well as the audio processing module in WebRTC can have some impact on POLQA scores. Hence, the reported scores do not reflect the POLQA scores that could be expected from the sole codec contribution (e.g. around 4.75 for EVS at 24.4

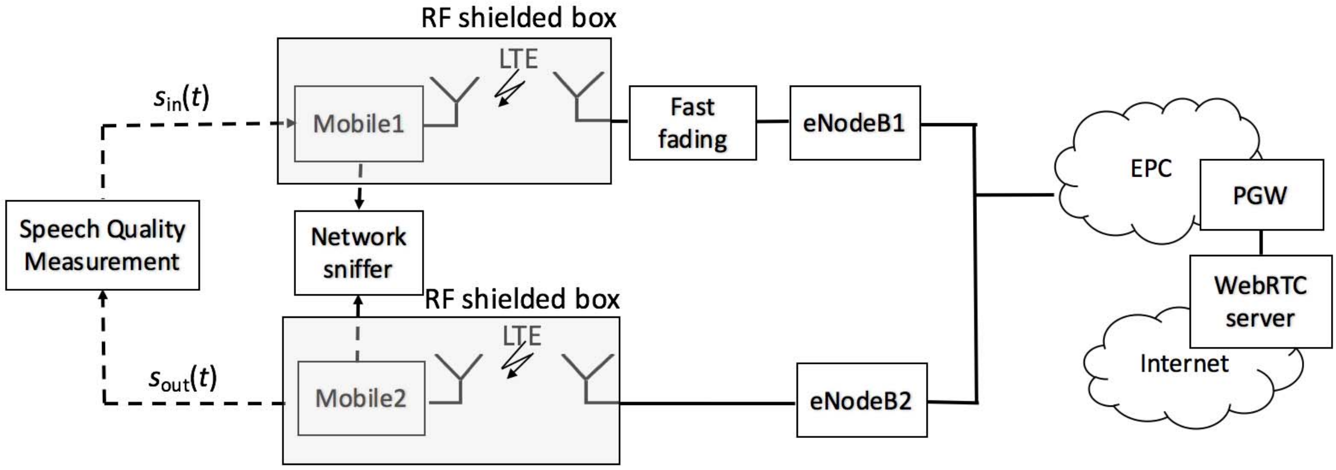


Fig. 1. Experimental setup.

kbit/s). For these reasons, POLQA scores depend on the phone models used for testing, and the values in Table I should not be taken as absolute values, they should only be understood as indicative examples.

Table II gives the corresponding measured mouth-to-ear delay. This delay is close to 300 ms in good coverage conditions, which is higher than the figure of 150ms recommended by ITU-T G.114 for good user satisfaction [25]. Similar to MOS, it has been noticed that different phone models can yield to significantly different delay values (e.g. up to 200-300 ms extra delay), therefore the specific values reported in Table II should only be considered as examples that are valid only for specific phones under test.

Note that in good network conditions, codec bit rate has a slight influence on MOS (i.e. 0.3 MOS difference between the lowest and highest used bit rate), while it does not have any effect on delay (as expected). It is also important to note that the two mobile phones used for testing had an IP address allocated by the EPC PGW (Packet Data Network Gateway) in the same local network, therefore speech packet streams did not get out of the EPC network; in other scenarios using a TURN (Traversal Using Relays around NAT) relay, mouth-to-ear delay could be expected to be higher.

TABLE I. MOS IN GOOD LTE COVERAGE

EVS (kbit/s)	Min. MOS	Avg. MOS	Max. MOS
9.6	3.5	3.84	3.97
13.2	3.97	4.09	4.15
24.4	4.06	4.11	4.15

TABLE II. M2E DELAY IN GOOD LTE COVERAGE

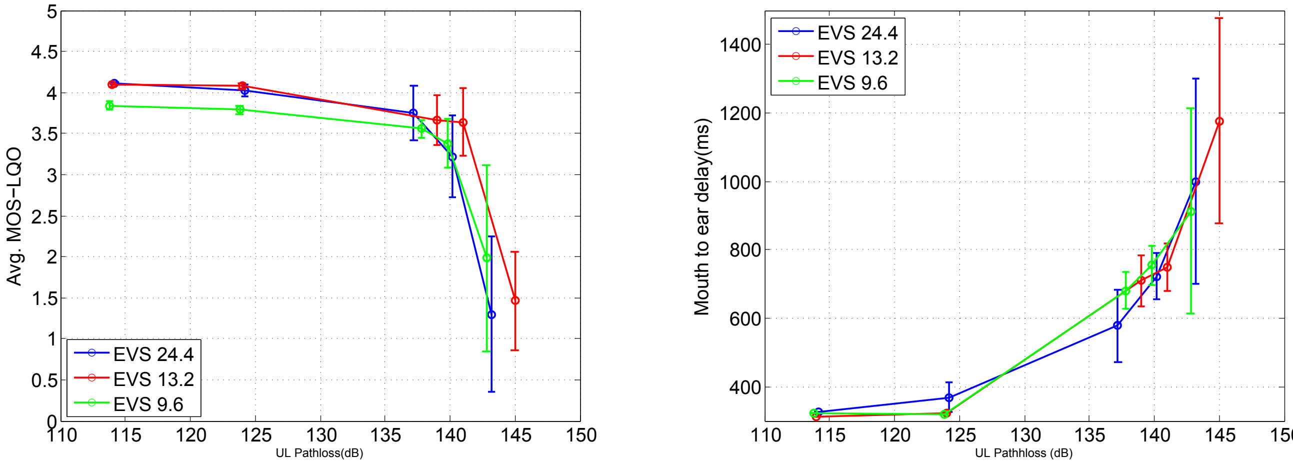
EVS (kbit/s)	Min. delay (ms)	Avg. delay (ms)	Max. delay (ms)
9.6	310	323	330
13.2	308	313	318
24.4	319	325	331

V. MOS AND DELAY AS FUNCTION OF LTE COVERAGE

Tests were conducted under degraded LTE coverage conditions using the fast fading generator to decrease the RSRP level and consequently increase radio pathloss until call drop. The same audio sequence was used (with 20 repeats of the same 8s sentence pair per pathloss condition). The uplink direction of Mobile1 was attenuated and Mobile2 was receiving in stable, good radio conditions. MOS and delay results are reported in terms of average and 95% confidence intervals.

Fig. 2 (a) shows the evolution of MOS as function of radio pathloss. The MOS value decreases in a similar way for the three tested codec bit rates and bit rate had a slight influence on coverage. It was expected that lower rates would enable to operate the codec in higher pathloss. This was verified by the trends shown for EVS at 13.2 and 24.4 kbit/s. The MOS curve at EVS 9.6 kbit/s was parallel to the MOS curve at 13.2 kbit/s with no cross-over; this may be explained by the lower instinsic codec quality at 9.6 kbit/s which did not seem to be compensated in more degraded network conditions, even close to the coverage limit. We highlight here that with the testbed used in this work there was an uncertainty of about 1 or 2 dB on actual pathloss due to fast fading (resulting in fluctuations in pathloss evaluation). Care should be therefore taken when intepreting the relative coverage of different bit rates. If the coverage limit was for instance set to a MOS threshold of 2.5, one would see that the corresponding path loss would be around 142, 143 and 141 dB at 9.6, 13.2 and 24.4 kbit/s; given the uncertainty on pathloss, coverage may be considered nearly equivalent for all tested bit rates. Note also that the confidence intervals indicate higher variability in higher pathloss and it would have been interesting to use more than 20 repeats of the 8s sentence pair to further reduce the size of confidence intervals (at the cost of increased test time).

Fig. 2 (b) shows that the average mouth-to-ear delay of WebRTC voice calls increases sharply with increasing pathloss. The same behavior can be noticed for the three different codec bit rates. This indicates that bit rate had no significant influence on the evolution of mouth-to-ear delay when degrading network condition. As discussed later, one can interpret the delay curves by the fact that data transmission



(a) Impact of LTE coverage on MOS score

(b) Impact of LTE coverage on M2E delay

Fig. 2. MOS and mouth-to-ear (M2E) delay results as a function of uplink (UL) pathloss.

over LTE was configured with acknowledged mode (AM); in this case, degrading radio coverage translated mainly into increasing end-to-end delay and jitter.

VI. Discussion

A. Analysis of RTP packet traces

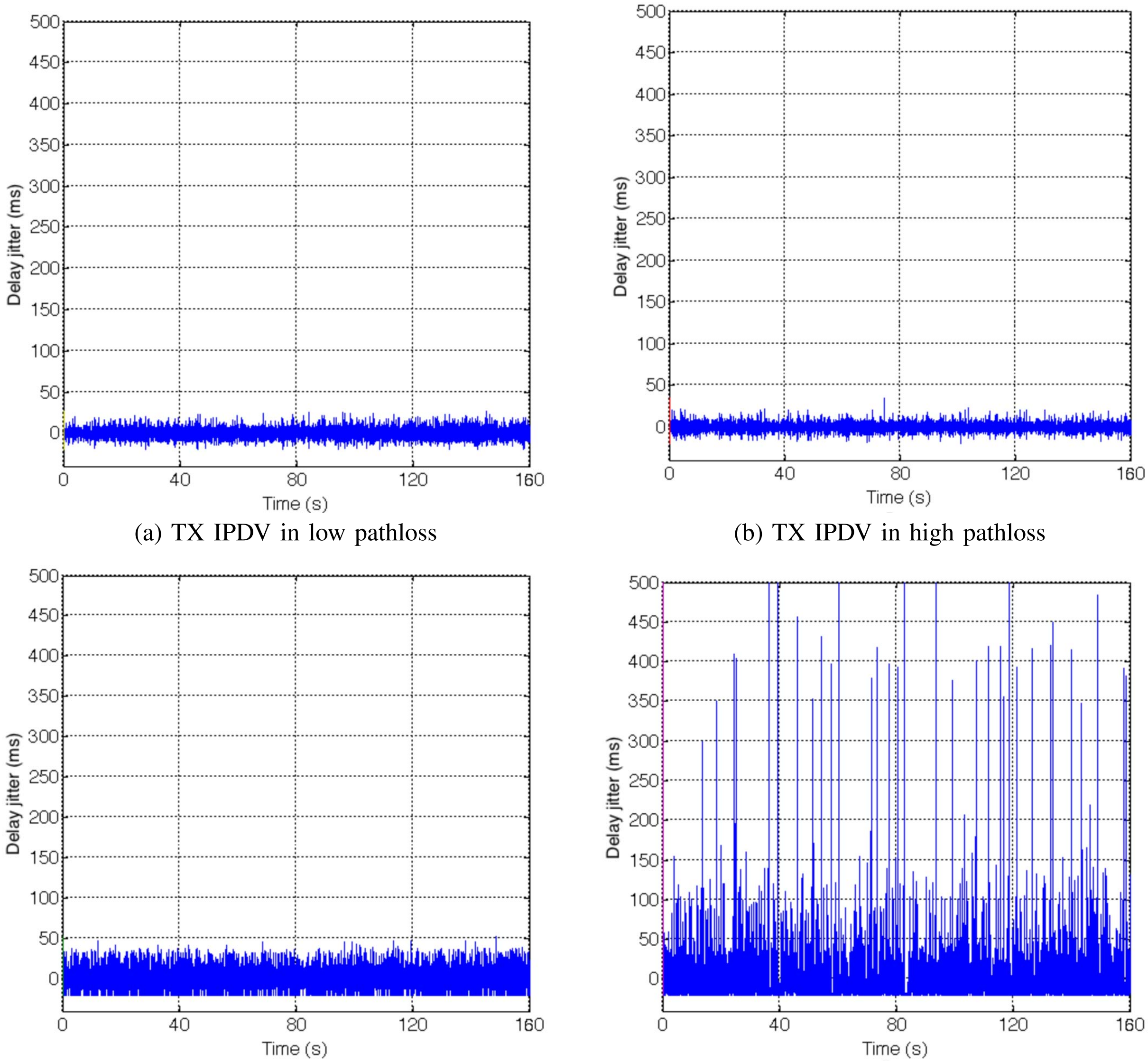
In this section, we report the analysis conducted on the incoming and outgoing RTP packet traces captured by the network sniffer at the two mobile phones. This analysis showed that the packet loss rate was $< 0.1\%$ in the worst case (near the coverage limit). This low packet loss rate can be explained by the acknowledged mode (AM) data transfer in the LTE Radio Link Control (RLC) protocol layer where unacknowledged packets are retransmitted – noting that hybrid automatic request (HARQ) is also used at the physical layer. The negligible packet loss rate during transmission cannot fully explain the slight MOS degradation when increasing radio pathloss. We attribute this degradation to jitter buffer induced packet losses and delay adjustments, when the WebRTC jitter buffer adapts to degrading network conditions. The jitter buffer in Chromium (called NetEQ) is designed for a specific compromise between quality and delay [26], with a target delay derived from an interarrival time (IAT) histogram and a specific delay peak detector, and an adaptative decision on receiver audio processing (normal decoding, expand, merge, or accelerate). Based on results in Fig. 2 (b), one can verify that the jitter buffer can adapt to wait for late packets or to skip some packets causing jitter-buffer induced losses. See also section VI-B.

A jitter buffer compensates for packet delay variations (jitter). Therefore, we calculated the instantaneous inter-packet delay variation (IPDV) defined by $IPDV(i) = D(i) - D(i-1)$, where $D(i)$ denotes the one-way delay of the ith packet. In the absence of jitter (i.e. for a perfectly synchronized transmission), the IPDV would always be 0. We present here the IPDV for one EVS bit rate coding (24.4 kbit/s) as an example. The other bit rates show similar behaviors. We present the instantaneous jitter in seconds for low and high pathloss level. Fig. 3 (a) and Fig. 3 (b) show the IPDV in sending direction in good and bad radio coverage, respectively. The IPDV presents here the delay variation observed at the mobile antenna of the sending phone (in the uplink of Mobile 1). As can be expected, the IPDV in sending (at the antenna point of Mobile1) does not depend on radio conditions. The IPDV in sending is centered and bounded by ± 20 ms. This means that speech frame are transmitted every 20ms as required by real-time operation, but there are internal timing variations (processing, buffering, operating system scheduling, etc.). Fig. 3 (c) and Fig. 3 (d) show the transmission delay variation observed at the receiving phone antenna (in the downlink of Mobile2) in good and bad radio coverage, respectively. For low radio pathloss (114 dB), few packets have an IPDV higher than 20 ms. Only 0.2% of packet had an IPDV higher than 40 ms. However, in high radio pathloss (142 dB), jitter is significantly higher. More than 7% of packets had an IPDV higher than 40 ms and IPDV was higher than 500 ms for some packets.

Note that the delay increase in degraded coverage condition cannot be explained only from jitter buffer adaptation; retransmissions (in particular in the LTE RLC layer) can also cause increased one-way transmission delay which impacts mouth-to-ear delay.

B. Analysis of recorded audio signals

Informal subjective listening was conducted on the speech samples recorded for different pathloss conditions. We noticed that in higher pathloss cases there can be significant artifacts due to jitter buffer time scaling, where speech segments were either shortened or expanded (stretched in time). In worst cases there were missing audio segments or talkspurts. Fig. 4 shows two waveforms corresponding to the reference signal $s_{in}(t)$ and an example of degraded signal $s_{out}(t)$ in high pathloss (140 dB) for EVS at 24.4 kbit/s. To better illustrate jitter buffer impacts, each 4s sentence is divided into 3 parts of different lengths A_i, B_i, C_i, where i =1 or 2 is the sentence index. In the degraded signal shown in Fig. 4, the first part A_1 was expanded, the middle part B_1 was significantly shortened, possibly to compensate for the previous expansion, and the



(a) TX IPDV in low pathloss

(b) TX IPDV in high pathloss

(c) RX IPDV in low pathloss

(d) RX IPDV in high pathloss

Fig. 3. Inter-Packet Delay Variation (IPDV) in low pathloss (114 dB) and high pathloss (142 dB) conditions, in sending (TX) and receiving (RX).

last part C_1 had a time scale close to the reference signal; in the second sentence, the middle part B_2 was expanded. An informal subjective test verified that the overall quality for the specific degraded signal shown in Fig. 4 was quite bad, because some words were lost in the first sentence and the rythm in the second sentence sounded distorted due to the expanded middle part. Still, the corresponding POLQA score was MOS-LQO$_s$ = 3.1. POLQA may not have been trained to predict such time scaling behaviours. A more extensive analysis including formal subjective tests would be required to better assess QoE; in this case, it would be more appropriate to replace the 8s speech test sequence used in this work by a larger speech corpus of male/female speech.

C. Comparison with VoLTE

Coverage tests were conducted with the setup described in Section III for Voice over LTE (VoLTE) calls and the AMR-WB codec at 23.85 kbit/s. Note that VoLTE with EVS will be tested in near future, hence the comparison to WebRTC with EVS was not yet possible at the time of writing. Results showed that coverage is better for VoLTE (with AMR-WB) than for WebRTC (with EVS) with a pathloss gain up around 3–5 dB, thanks to LTE radio optimizations (in particular TTI bundling and robust header compression). Speech quality (MOS) was found to be more stable for VoLTE calls. Moreover, mouth-to-ear delay increase with degraded conditions was smaller for VoLTE. Note that VoLTE is based on Unacknowledged Mode (UM) transmission over LTE, hence quality degradations at cell edge are typically more dominated by packet loss than by jitter effects.

VII. Conclusion

In this paper, we reported experimental results on WebRTC voice call quality in different LTE radio conditions. We discussed results by analyzing the overall jitter buffer impact on the measured MOS and delay, based on captured

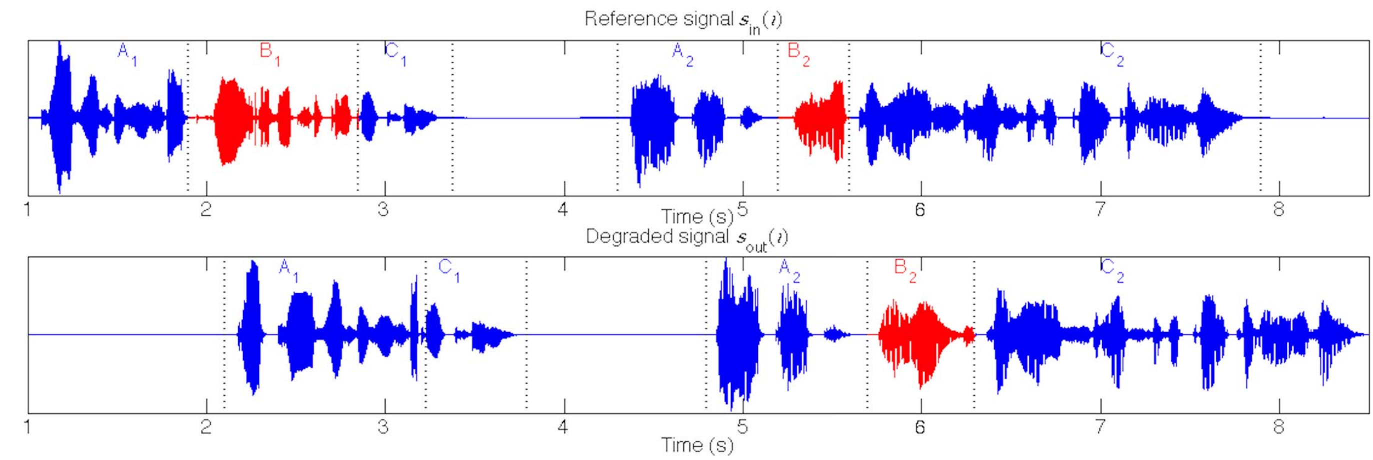


Fig. 4. Illustration of jitter buffer adjustments in high pathloss (140 dB) for EVS at 24.4 kbit/s.

packet streams and recorded audio signals. Future work will consider using WebRTC statistics (from the getstats API) and extending this study to evaluate WebRTC voice quality in other access networks (e.g. Wifi). A comparison of subjective quality and POLQA predictions also will be of interest to assess accurately the impact of WebRTC's jitter buffer (NetEQ) on QoE. Adaptation algorithms (e.g. bit rate adaptation, use of inband FEC) implemented for OPUS in Chromium's WebRTC library will also be studied.

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