A Comparative Analysis of the Perceived Quality of VoIP under Various Wireless Network Conditions

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Abstract. This paper performs a comparative analysis of the perceived quality of (unidirectional, non-interactive) VoIP calls under various wireless network conditions (e.g., handover, high traffic demand). It employs the PESQ tool, E-model and auditory tests to evaluate the impact of these network conditions on the perceived quality of VoIP calls. It also reveals the inability of PESQ and E-model to capture the quality of user experience. Furthermore, it shows that the network condition and the evaluation method exhibit statistically significant differences in terms of their reported opinion score values. Finally, the paper highlights the benefits of the packet loss concealment of the AMR 12.2kb/s and the QoS mechanisms under these network conditions.

1 Introduction

Wireless networks often experience "periods of severe impairment" (PSIs), characterised by significant packet losses in either or both directions between the wireless Access Points (APs) and wireless hosts, increased TCP-level retransmissions, rate reduction, throughput reduction, increased jitter, and roaming effects. A PSI can last for several seconds to the point that it can be viewed as an outage. The frequency and intensity of PSI events in modern home and enterprize wireless networks is not well understood. Very few studies analyze the impact of PSI events on the quality of user experience. The throughput, jitter, latency, and packet loss, have been used to quantify network performance and various studies have shown their performance under different network conditions (e.g., handoff, contention, and congestion). Some important observations have been made in the context of wireless networks: (a) handovers result to packet losses (e.g., [1,2]), (b) queue overflows at APs lead to poor VoIP quality (e.g., [3]), and (c) average delay does not capture well the VoIP quality because of the burstiness of packet losses (e.g., [4]). For various applications, a maximum tolerable end-to-end network delay has been estimated (e.g., about 150ms for

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VoIP applications [5,6]). Could such crude statistics accurately denote the quality of experience? There is evidence that depending on the temporal statistical characteristics of the packet losses and delays during a call, the impact on the user experience varies. However, there are a few comparative analysis studies of the impact of various network conditions on the perceived quality of experience.

Our attention has shifted from MAC- and network-based metrics to *application-based, objective and subjective user-perception metrics.* Specifically, our recent work [7] employed the E-model [8] and PESQ tool [9], aiming to quantify the VoIP quality under various wireless network conditions, namely, during a handover and under different background traffic conditions (normal and heavy traffic load/saturation conditions) at an AP. For each scenario/network condition, empirical-based measurements were collected from a real-life testbed. The analysis showed that both the network condition and codec type (G.711 vs. AMR 6.7kb/s vs. AMR 12.2kb/s), as well as their interaction, have a significant impact on the quality of user experience values. A comparative analysis of the E-model and PESQ with the Student's T-test reported significant differences between the estimations of these two models, which further motivated the need for more accurate user perception metrics. This paper builds on that work, extending it with *auditory (subjective listening) tests* and an analysis of the impact of QoS mechanisms on the perceived quality of VoIP. The main contribution of this paper is a methodology for evaluating the impact of different network conditions on the perceived quality of VoIP, which can be further extended to other applications and network environments. Specifically, it analyzes and discusses the following issues:

- the impact of network condition (handover, heavy TCP traffic, heavy UDP traffic), codecs, and QoS mechanisms on quality of user experience,
- the use of the E-model, PESQ tool, and auditory tests to estimate the quality of user experience.

For each scenario/network condition, empirical-based measurements were collected from a real-life testbed and listening tests were also conducted using VoIP recordings that correspond to these network conditions. The *impact* and *significance* of network conditions and evaluation metric (subjective tests, E-model, PESQ) on the estimated quality of user experience is identified using ANOVA. The rest of the paper is organized as follows: Section 2 outlines the related work. Section 3 describes our testbed, the different network conditions and discusses the analysis results. Finally, Section 4 presents our main conclusions and future work plans.

2 Related Work

While there have been several studies discussing the network statistics under different conditions, most of them focus on the impact of these conditions on the aggregate throughput and capacity. The IEEE802.11 handover has been analyzed and various improvements have been proposed. For example, Forte *et al.* [10] analyzed the various delays involved in the handoff/reassociation process in an experimental testbed and the impact of the handoff on a SIP call.

They reduced this overhead by enabling the wireless device to acquire a temporal address. SyncScan [11] reduces the network unavailability during an AP handoff by enabling the client to synchronize the scanning phase with the APs' beacons. Pentikousis *et al.* [12] measured the capacity of a WiMAX testbed in terms of VoIP calls. Ganguly *et al.* [13] evaluated various packet aggregation, header compression, adaptive routing, and fast handoff techniques. Anjum *et al.* [14] performed an experimental study of the VoIP in WLAN, quantifying the VoIP capacity under light and heavy traffic load, and the practical benefits of implementing backoff control and priority queuing at the AP. Finally, Shin *et al.* [6] performed empirical-based measurements and simulations to estimate the capacity of an 802.11 network in terms of number of VoIP calls and analyzed the impact of the preamble size, ARF algorithm, RSSI, packet loss, and scanning. They used as criterion for the quality of calls that the end-to-end delay should not exceed 150 ms and the packet loss probability should be 3% or less. In the context of Mobisense project, Deutsche Telecom Lab has developed a Next Generation Network (NGN) testbed and implemented a system that enables seamless codec changes to improve the quality during handovers [15]. They performed subjective tests to quantify the degradation in user perceived quality for various types of network changes, namely handovers between various types of networks and changeovers between various codecs [16]. An analysis of the E-model and PESQ quality estimation tools was also performed in the context of the NGN testbed [17] and an enhancement of the E-Model by adding a bandwidth switching impairment factor was proposed [18]. Chen *et al.* [19] analyzed the user satisfaction in Skype, employing the call duration as the quality benchmark. Hoene *et al.* [20] evaluated the call quality in adaptive VoIP applications and codecs and showed that high-compression codecs (with relatively low voice quality) may behave better than top-quality codecs under packet losses and limited available bandwidth. Markopoulou *et al.* [21] focused on ISP network problems and showed that ISP networks suffer from PSIs affecting the real-time applications.

3 Performance Analysis

3.1 Network Conditions, Scenarios, and Testbeds

We distinguish several network conditions that result in PSIs and form the following scenarios:

- **handover:** no background traffic, user mobility and client handover between wireless APs
- **heavy UDP traffic:** no user mobility, UDP flows saturating the wireless LAN
- **heavy TCP traffic:** no user mobility, TCP flows, generated by a BitTorrent client, saturating the wireless LAN

We setup two control testbeds, namely the *handover testbed* (in which a user, performing a VoIP call, roams in the premises of FORTH) and the *background traffic*

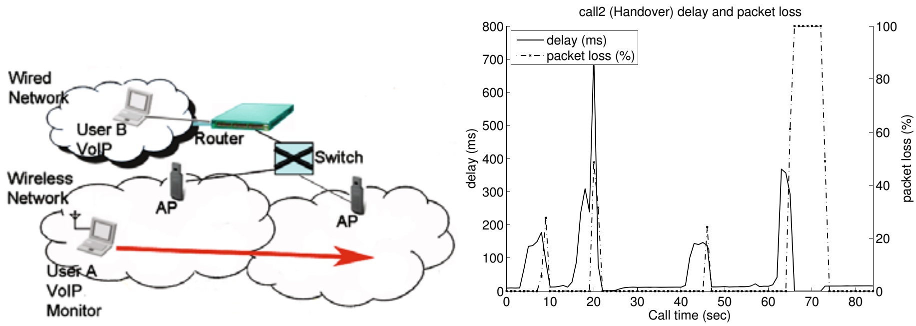

Fig. 1. Handover scenario: User A moves to the coverage area of a different AP while (s)he participates in a VoIP call with user B

testbed (in which background traffic that corresponds to the last two scenarios is generated). A recording of a female voice around 1:30 minutes long (source file) was "replayed" under the aforementioned network conditions. In each testbed, we emulated the corresponding conditions (background traffic/user mobility) of each scenario, "replayed" the source file, and collected the traces at the wireless VoIP client for analysis. Specifically, we analyzed the impact of each condition on the perceived user experience of the VoIP call. Note that these VoIP calls are essentially unidirectional (streaming-like and non-interactive). The network adapter of the wireless VoIP client captures packets in promiscuous mode with IEEE802.11+Radiotap pseudo-header provided by libpcap, using tcpdump with the appropriate settings. This header contains the RSSI value for each packet, the data rate, and the operating channel. The VoIP clients used H323 software with an G.711 codec (64kb/s).

The handover testbed includes one VoIP client connected via FastEthernet and one VoIP client connected via IEEE802.11 to the ICS-FORTH infrastructure network. A user holding a wireless laptop (User A) roams in the premises of ICS-FORTH. While moving, the wireless client slowly walks out of range of the AP and a handover is performed. As empirical studies have shown, handoff between APs in wireless LANs can consume from one to multiple seconds, as associations and bindings at various layers need to be re-established. Such delays include the acquisition of a new IP address, duplicate address detection, the reestablishment of secure association, discovery of available APs. The overhead of scanning for nearby APs can be of 250ms (e.g., [22,11]), far longer than what can be tolerated by VoIP applications. The active scanning in the handoff process of the IEEE802.11 is the primary contributor to the overall handoff latency and can affect the quality of service for many applications. The background traffic testbed includes a VoIP client connected via IEEE802.11, a VoIP client connected via FastEthernet, four wireless nodes connected via IEEE802.11 and one node connected via FastEthernet. The four wireless nodes produce the background traffic according to the predefined scenarios. All wireless nodes are connected to a single AP.

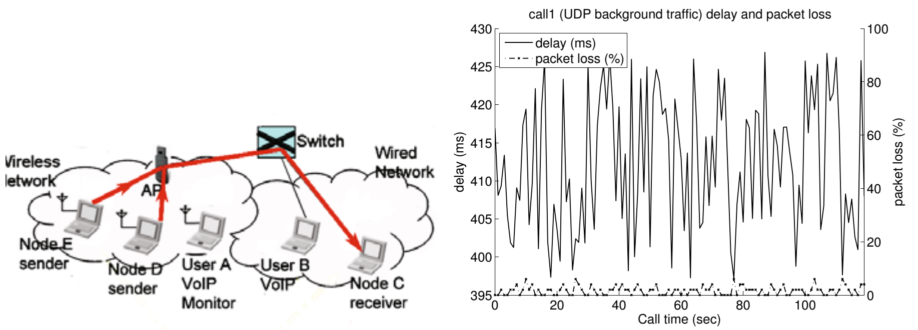

Fig. 2. Heavy UDP traffic scenario: each of the nodes D, E, F and G transmit 2Mb/s UDP traffic towards node C (nodes F and G are not shown)

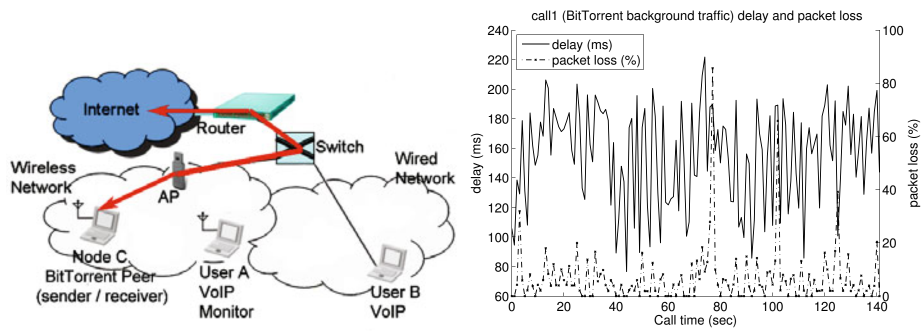

Fig. 3. Heavy TCP traffic scenario: Node C exchanges BitTorrent traffic with Internet peers (both uplink and downlink traffic)

The *heavy UDP traffic* scenario focuses on the quality of VoIP under congestion caused by a large amount of traffic load generated by a small number of flows, overloading the AP. Each of the four wireless nodes sends packets of 1500 bytes of UDP traffic to a wired node at a 2Mb/s data rate (a total of 8Mb/s). The AP operates in IEEE802.11b and the aggregate traffic exceeds the theoretical maximum throughput of an IEEE802.11 network (approximately 6Mb/s [23]). The two VoIP clients initiate a call under these conditions. These scenarios exhibit phenomena of congestion of the wireless channel and continuous contention of the wireless nodes.

In the *heavy TCP traffic* scenario, the background traffic is generated by one wireless node running a BitTorrent client, downloading three highly seeded files (while the VoIP call takes place). The BitTorrent protocol splits the files into small chunks and simultaneously downloads and uploads the shared chunks. In general, the number of generated flows in BitTorrent is high, often causing low-end routers to run out of memory and CPU. As in the previous scenarios, the AP operates in IEEE802.11b mode. The BitTorrent protocol introduces a high number of small TCP flows in both uplink and downlink directions, contending for the medium. This behavior puts stress on the queue, CPU and memory of APs.

3.2 Measurements and Evaluation

We performed a number of VoIP calls for each of the aforementioned scenarios (as shown in Figures 1, 2, and 3, user A initiates VoIP calls with user B), and collected the VoIP traces for analysis. Specifically, we measured the end-to-end delay and packet loss of the VoIP flow under the different network conditions, namely, handover, heavy UDP traffic, and heavy TCP traffic at the application layer. To measure the performance of a VoIP call, we used subjective and objective tests. The objective tests include the E-model and PESQ tool that report a Mean Opinion Score (MOS) value. In both studies, the same VoIP calls were used. For the auditory tests, a recording of a female voice of around 1:30 minutes long (source file) was "replayed" under the aforementioned network conditions. The received files (recorded at the wireless VoIP receiver of the testbed), each corresponding to a network condition, were used in the subjective study. The corresponding opinion scores reported by ten subjects that listened to these files were analyzed.

The E-model depends on various factors, such as voice loudness, background noise, equipment impairment, packetization distortion, codec robustness under various packet loss and end-to-end delays and impairments introduced by the packet loss and end-to-end delays and produces an *R-factor*, a rating that estimates the voice quality [8]: $R = R_o - I_s - I_d - I_{e-eff} + A$. The term R_o accounts for the basic signal-to-noise ratio the user receives and takes into consideration the loudness of the voice, the noise introduced by the circuit and by background sources. The term I_s represents voice specific impairments, such as too loud speech level, non-optimum sidetone and quantization noise, while the term I_d represents the impairments introduced by delay and echo effects. The term I_{e-eff} is the equipment impairment factor, which corresponds to impairments due to low bit-rate codecs and packet losses (i.e., percentage of packet losses and their burstiness index $BurstR$). Finally, the term A is an "advantage factor" that takes into consideration the user's expectation of potential glitches. All these factors have been extensively analyzed in ITU-T's G.107 recommendation (E-model). All E-model parameters are set to their default values, except for B_{pl} which is set to 25.1 (as G.113 recommends for G.711). The ITU-T provides an R-to-MOS conversion formula.

To extend our assessment with additional quality metrics, we also employ the PESQ test. As mentioned earlier, the E-model takes into account both delay and packet losses, while the PESQ focuses on packet loss effects, which were more significant than delays. PESQ gives the estimated perceptual difference between two audio signals, with the limitations that the samples must be temporarily synchronized and of 6 s to 20 s duration. The former requirement proved to be difficult when comparing recordings, so we opted to employ only the effects of packet loss and disregard any delay. The packet loss data from different scenarios was used with each of the three codecs (G.711 64kb/s, AMR 6.7kb/s, and AMR

12.2kb/s). Specifically, we first employed the collected packet traces (of the VoIP calls) with the packet loss information and encoded an audio signal based on each codec. We repeated the encoding using the same packet trace but without considering any packet loss to construct a *baseline audio signal*. Then, the PESQ tool estimated the MOS by comparing these two audio signals for each codec. Note that in this analysis, PESQ does *not consider any delay* information. In the case of G.711 codec, the packet loss locations were simply removed from the pulse-code modulation (PCM) audio, whereas in the case of AMR codecs, the lost packets were indicated by manipulating the bad frame bit of the packet headers in the encoded bitstream. In all cases, the PESQ test was performed between the coded audio *without* and *with* simulated packet losses in 10 s frames with 1 s "step size" (sliding window) for the entire call duration. The metric for a call was the average of all MOS values, each corresponding to a 10 s frame of that call.

We observed two types of handover, namely the *fast handover* and the *handover with deauthentication* (that lasts longer). Calls with *fast handovers* are characterized by minor packet losses and delays, resulting in close to excellent quality. Within a handover, the client initiates an active AP discovery during which packets are queued up. On the other hand, if a handover with deauthentication occurs, the inter-AP protocol will not handle the pending packets (at the old AP). In this case, the error rate and unacknowledged retransmissions will increase, and as a result, the degradation of MOS will become more prominent.

In the VoIP under heavy UDP traffic scenario (e.g., Figure 2), the MOS deteriorates due to the high packet delays. In this scenario, the very large delays are due to the presence of heavy background traffic resulting in an arrival rate higher than the 'service' rate at the AP (also observed in other studies, e.g., [24]). Indeed, a saturated network with full buffers will increase the mean delay values, trying to deliver all packets and occasionally dropping packets from the queue when a timeout occurs. The E-model reports a mediocre quality for these VoIP calls while PESQ results in a relatively good performance. The subjects in the auditory tests also report a reasonably high opinion score value. This is due to the "unidirectional" (and non-interactive) nature of these VoIP tests. In general, this scenario highlights the need for a prioritization scheme for different traffic classes, such as IEEE802.11e (also indicated in other studies, e.g., [3]).

In the VoIP under heavy TCP traffic scenario, the calls suffer from relatively high packet losses and delays. Although packet delays exceed the 150ms threshold, the overall voice quality is acceptable, consistently across E-model, PESQ, and subjective tests (in disagreement with the "rule-of-thumb") [7]. The nature of the BitTorrent protocol can explain this behavior: a BitTorrent client initiates many flows, with small payload sizes. Each flow tries to expand its TCP window, up to the point that packet losses occur, triggering the TCP congestion control which will drop the throughput of that flow. Other flows active at that time will also manifest this behavior. Since the number of flows at any given time is large, this phenomenon is repeated frequently, causing severe performance degradation (e.g., packet drops at the AP). In some calls, the large number of flows initiated by the BitTorrent client saturates the wireless LAN.

A preliminary analysis of VoIP calls shows a prominent discrepancy between the E-model MOS and PESQ MOS. In addition, it illustrates that not all the network conditions impact the MOS in the same manner [7]. We statistically analyzed the impact of the different codecs and scenarios on the user perception metrics. To investigate which parameters have a dominant impact on MOS, a two-way ANOVA was performed. The PESQ MOS was used as the user perception metric. Dependent variable is the average PESQ MOS value of each call, and the independent variables are the scenario and codec type. ANOVA indicates that *scenario* and *codec* type, as well as their interaction, have a *significant effect* on the PESQ MOS values. Furthermore, a multiple comparison test with Tukey's HSD criterion reveals the following: The handover exhibits higher MOS values than all other scenarios. The heavy TCP traffic performs similarly (in terms of MOS) as the heavy UDP traffic. The performance of AMR 6.7kb/s is similar to the performance of G.711 64kb/s (lower data rate vs. concealment tradeoff). The AMR 12.2kb/s performs *significantly better* (higher MOS) than G.711 64kb/s and AMR 6.7kb/s. The level of significance in all tests was set to 0.05. The AMR 12.2kb/s more sophisticated packet loss concealment justifies its better performance. The similar performance of AMR 6.7 kb/s and G.711—significantly lower than the performance of AMR 12.2kb/s—highlights the benefits of the packet loss concealment of the AMR 12.2kb/s under these network conditions. Note that the PESQ MOS of each call is the *average* of the values that correspond to all 10 s frame of the call.

To investigate if there are significant differences between the measurements of E-model and PESQ, we employed the Student's T-test. Specifically, we compared the average call MOS values of the G.711 codec across all scenarios. The test indicates *statistically highly significant* ($p < 0.01$) differences between the estimations of the two models. Especially, under heavy packet loss, E-model reports lower MOS values than PESQ. Finally, as the AMR codec tests show, in the context of heavy losses, it is beneficial to increase the codec bit-rate. A detailed discussion can be found in [7]. The statistically significant differences between the estimations of the PESQ tool and E-model motivated the need for auditory tests. Ten members of the FORTH-ICS, of age between 22-35 years old, without any hearing impairments, participated in an auditory test study. Specifically, for this study, we selected three calls, each corresponding to a network condition. The subjects listened to these three calls and reported an opinion score for each of them.

To investigate if there are significant differences between the measurements of E-model, PESQ and subjective tests, we again employed the ANOVA and Tukey's HSD test. The G.711 codec was used in all the calls. The test indicates statistically significant differences based on the evaluation method (criterion), scenario and their interplay (scenario and criterion). From the ANOVA and Tukey's HSD test results (Figure 4), we conclude that the heavy UDP is significantly different from the heavy TCP for each criterion. The heavy TCP is

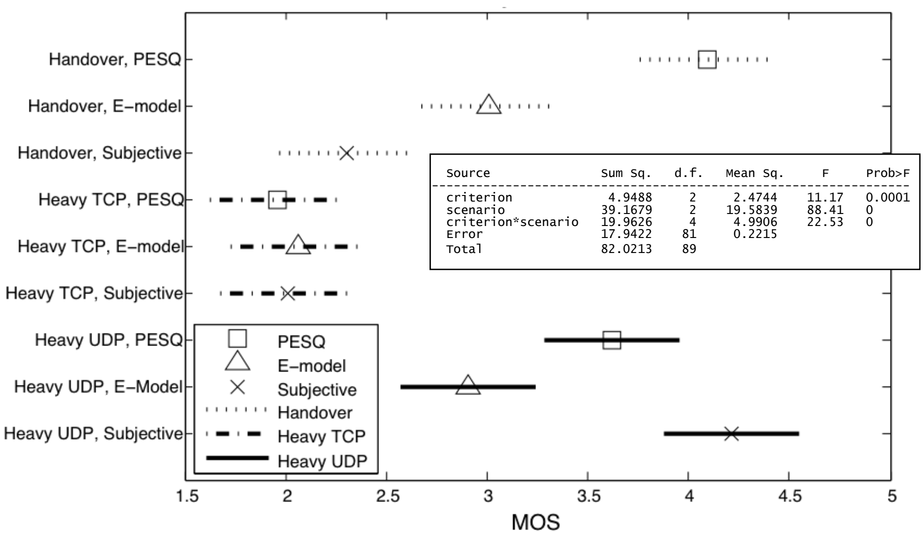

Fig. 4. Comparative statistical analysis of the impact of criterion (PESQ, E-model, and subjective tests) and scenario on MOS using the Tukey's HSD test. The corresponding ANOVA report is in the inset figure.

significantly different from the handover for the E-model and PESQ, respectively. Interestingly, in the case of handover, the three evaluation methods have significant differences from each other, while in heavy TCP, there are no significant differences. The comparison of the E-model and PESQ with the subjective tests reveals several weaknesses and distinct characteristics of these two metrics. For example, after the subjective tests, some users commented that the long pauses of the handover scenario had a strong negative impact on their experience. Due to its averaging, PESQ "masks" the negative impact of the intervals that correspond to the long pauses of the handover. On the other hand, although the E-model (using the packet loss burst ratio) deviates less from the subjective tests than PESQ, it still cannot capture accurately the effect of these long pauses. Moreover, the E-model considering the large delay may underestimate the performance (e.g., in the heavy UDP traffic scenario), while the lack of interactivity may mask its impact in the subjective tests. We plan to investigate the impact of the relative position and duration of long pauses on the perceived quality.

In the above measurements, there was no QoS mechanism enabled. To understand the impact of QoS, we enabled IEEE802.11e and WiFi Multimedia (WMM) on the Cisco APs and Class-based Weighted Fair Queuing on the Cisco router and repeated the empirical study. In this QoS-enabled empirical study, we only consider the G.711 codec. Note that the two VoIP experiments used in the subjective study in the context of handover are different. However the mobile user participating in the experiment followed the same path in the premise of the ICS-FORTH. In the handover scenario without QoS, we had observed

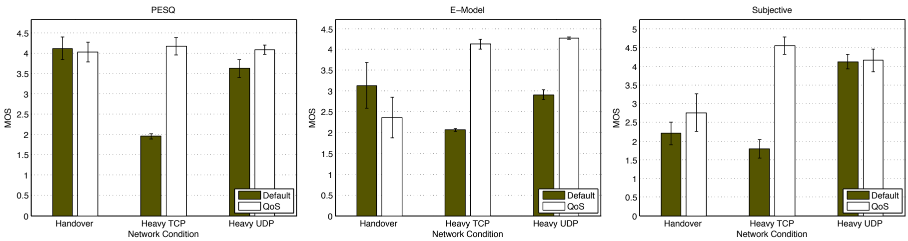

Fig. 5. Impact of QoS on MOS values of the VoIP calls, for different network conditions (all with G.711 codec). The default corresponds to the testbed without QoS. (95% confidence interval).

the presence of *handovers with deauthentication* that cause long pauses (6s or more) during a call. In these cases, the AP deauthenticates the client by sending *Previous authentication no longer valid* messages. According to the AP manufacturer [25], such deauthentication occurs when the error rates and the number of unacknowledged retransmissions reach an AP-specific threshold. When we repeated the experiment with QoS enabled, the presence of such handovers was even more prominent. Interestingly, the majority of the handovers lasted for 6s or more, because the AP was deauthenticating the client during the scanning phase. We speculate that a QoS-enabled AP attempts to transmit a larger number of high-priority frames during the client's scanning phase than a QoS-agnostic AP, reaching the aforementioned deauthentication threshold faster. We plan to investigate further this behavior. However, regardless of the causes for deauthentication, such long pauses severely impact the user perceived quality.

As expected, the QoS mechanisms improve the user experience of VoIP calls under all network conditions. Specifically, the QoS mechanisms improve the performance of VoIP calls under heavy UDP traffic. Especially, in the case of the E-model, their benefits are noticeable (as shown in Figure 5). In the case of heavy TCP traffic, the improvement is even more prominent, exceeding 100%. In the above QoS-enabled performance analysis, the G.711 64kb/s codec was used. In the context of a QoS-enabled emulation testbed, we also analyzed the performance of AMR 12.2kb/s and AMR 6.7kb/s. In the case of heavy UDP traffic and heavy TCP traffic, the user perceived quality using AMR 12.2kb/s is excellent, while the AMR 6.7kb/s performs close to G.711. In handover, the PESQ performs consistently higher than the subjective tests and E-model. However, this is an "overestimation" of the MOS due to the averaging that we perform by taking into consideration all the MOS values that correspond to individual 10s frames. A different approach in estimating the MOS value of a VoIP call using PESQ could potentially improve its estimation.

4 Conclusion and Future Work

The paper discusses situations in which common "rule-of-thumb" metrics cannot reflect the user-perceived quality. A comparative evaluation of the quality of

VoIP calls using PESQ, E-model, and subjective tests demonstrates the need of more accurate metrics, tailored to the specific requirements of the application at hand. In the context of VoIP calls, our analysis reveals the inability of PESQ and E-model to capture the user experience under specific network conditions. It also shows that the impact of network conditions, codecs, and their interplay on the perceived quality of experience varies. The analysis highlights the benefits of the packet loss concealment of the AMR 12.2kb/s and the QoS mechanisms under these network conditions. However, our experiments also indicate instances of deauthentication during handovers, resulting in severe performance degradation. As shown in this paper, not all network conditions impact the quality of VoIP applications in the same manner. Understanding which conditions cause severe impairment in the VoIP application, and which cross-layer measurements can be used to predict such impairment, is important in the design of adaptation mechanisms.

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