Analyzing the impact of various wireless network conditions on the perceived quality of VoIP

Ilias Tsompanidis Georgios Fortetsanakis Toni Hirvonen Maria Papadopouli*
Department of Computer Science, University of Crete &
Institute of Computer Science, Foundation for Research and Technology - Hellas

***Abstract*—This paper focuses on a comparative statistical analysis of the performance of VoIP calls under various situations, namely, during a handover and under different background traffic conditions at a wireless access point (AP). Using empirical-based measurements, it demonstrates that these network conditions exhibit distinct statistical behaviour, in terms of SNR, packet losses and end-to-end delays, and thus, impact the VoIP user quality in a different manner. The analysis shows that both network conditions and codec type, as well as their interaction, have a significant effect on the PESQ MOS values. Moreover, it indicates statistically highly significant differences between the estimations of the PESQ and E-model. Finally, it highlights the benefits of the packet loss concealment of the AMR 12.2kb/s under these network conditions.**

I. 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/hand-off 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 *user experience*. For various applications, a maximum tolerable end-to-end network delay has been estimated (e.g., about 150ms for VoIP applications [1], [2]). However, it is unclear whether the use of this fixed threshold as a "rule-of-thumb" can result in an efficient and effective adaptation for a wireless device. Depending on the temporal statistical characteristics of the packet losses and delays during a time period, the impact on the user experience may vary.

Network benchmarks, such as 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: (a) handoffs result to packet losses (e.g., [3], [4]), (b) queue overflows at APs lead to poor VoIP quality (e.g., [5]), and (c) average delay does not capture well the VoIP quality because of the burstiness of packet losses (e.g., [6]). However, there is no comparative statistical analysis of the impact of various network conditions on the perceived quality of experience. This paper shifts the attention from MAC- and network-based metrics to *application-based/user-perception metrics* and aims 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. A novel part of this work is the comparative analysis of these conditions and their impact on the quality of VoIP calls. One 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. The paper analyzes and discusses the following issues: (a) impact of network condition on user perception, (b) impact of codecs on user perception and their interplay with network conditions, and (c) comparative analysis of the various user-perception metrics. Specifically, for each scenario/network condition, we perform empirical-based measurements, using traces collected from a real-life testbed. The measurements show that various network conditions exhibit *distinct statistical behaviour* in terms of SNR, packet losses and end-to-end delays. We then assess the user perception using the PESQ tool [7] and E-model [8]. Furthermore, we experiment with various codecs, such as G.711 64kb/s, AMR 6.7kb/s, and AMR 12.2kb/s and show their impact on user perception in the context of these network conditions.

Using ANOVA, we identified the *impact* and *significance* of these network conditions and codecs on the user experience. The analysis shows that both the network condition and codec type, as well as their interaction, have a significant impact on the MOS values. Finally, a comparative analysis of the E-model and PESQ with the Student's T-test shows that there are significant differences between the estimations of these two models, which further motivates the need for more accurate user-perception metrics. The rest of the paper is organised as follows: Section II outlines the related work. Section III describes our testbed, the different network conditions and discusses the statistical analysis results. Finally, Section IV presents our main conclusions and future work plans.

II. 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.* [9] analyzed the various delays involved in the handoff/reassociation process, namely the DHCP request, ARP

* Contact author: Maria Papadopouli (email: mgp@ics.forth.gr)

query, and SIP invite, 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 [10] aims to reduce the network unavailability during an AP handoff that causes a decrease in throughput due to the background scanning. It minimises the scanning by enabling the client to synchronize the scanning phase with the APs' beacons. Ganguly *et al.* [11] evaluated various packet aggregation, header compression, adaptive routing, and fast handoff techniques. Anjum *et al.* [12] 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.* [2] perform empirical-based measurements and simulations to estimate the capacity of an IEEE802.11 network in terms of number of VoIP calls and analyse the impact of the preamble size, ARF algorithm, RSSI, packet loss, and scanning. Their criterion for the quality of calls is that the end-to-end delay should not exceed 150ms and the packet loss probability should be 3% or less.

The following papers are closer to this work: Chen *et al.* [13] analyzed the user satisfaction in Skype, employing the call duration as the quality benchmark. Specifically, they assumed that the more a user keeps a VoIP call active, the better the quality of the call. Hoene *et al.* [14] evaluated the call quality in adaptive VoIP applications and codecs. They showed that low-bitrate codecs (with relatively low voice quality) may behave better than higher-bitrate codecs under packet losses and limited available bandwidth. Markopoulou *et al.* [15] focused on ISP network problems and showed that ISP networks suffer from PSIs (e.g., outages due to routing table changes, relatively frequent spikes in packet losses and delays) affecting the real-time applications.

III. Performance analysis

A. 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
- **normal AP hotspot traffic**: no user mobility, TCP flows corresponding to background traffic at a campus-wide wireless hotspot AP
- **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 testbed* (in which background traffic that corresponds to the last three scenarios is produced). Under each scenario, we performed a number of VoIP calls and collected the traces that correspond to the VoIP traffic (as received at the VoIP client). Using the collected traces, we measured the network performance and analyzed

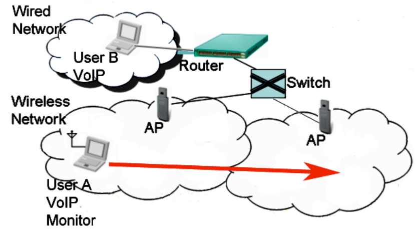

Fig. 1. Handover scenario. User A moves to the coverage area of a different AP while in a VoIP call with user B. (Set 1: 2 calls, Set 2: 12 calls)

the impact of each condition on the perceived user experience of the VoIP call.

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 fades 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., [16], [10]), far longer than what can be tolerated by VoIP applications. The active probing (*i.e.*, sending probe requests for discovering the available APs in different channels) in the handoff process of the IEEE802.11 MAC 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 predefined scenarios. All wireless nodes are connected to a single AP.

1) Normal hotspot AP traffic: We employed traces from the UNC/FORTH data repository [17], [18], [19], [20] that correspond to a real-life hotspot AP under normal traffic conditions in a large campus-wide wireless infrastructure. Each flow of the trace is "replayed" as a TCP flow with its sender and receiver corresponding to a wired and a wireless node, respectively. 5-minute traces are replayed as background traffic, while a VoIP call of the same duration is established between the two VoIP clients. The AP operates in IEEE802.11b, in accordance with the APs of the UNC/FORTH traces. This type of traffic cannot overload the AP or cause any bottleneck at the AP, and thus, the network provides a near-excellent VoIP

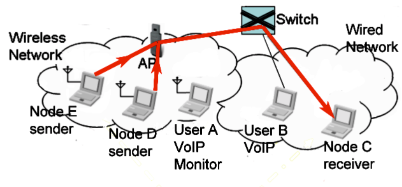

Fig. 2. Heavy UDP traffic scenario. Nodes D, E, F and G transmit 2Mb/s UDP traffic each towards node C (nodes F and G are not shown). (Set 1: 2 calls, Set 2:21 calls)

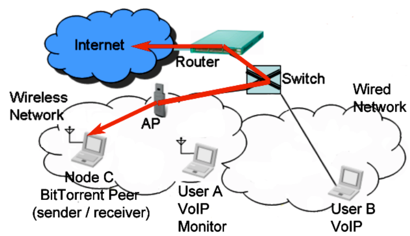

Fig. 3. Heavy TCP traffic scenario. Node C exchanges BitTorrent traffic with Internet peers (both uplink and downlink traffic). (Set 1: 2 calls, Set 2:10 calls)

quality.

2) Heavy UDP traffic: This 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. We have performed 3 different sets of calls, namely the *UDP-8, UDP-16* (extremely heavy UDP traffic), and *UDP-7*, varying the number of wireless nodes producing the heavy background UDP traffic and their bitrate. Specifically, in *UDP-8*, 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). In the *UDP-16*, there are also four wireless clients, each transmitting with a 4Mb/s data rate (a total of 16Mb/s), while in *UDP-7*, there are two clients, each sending with a 3.5Mb/s date rate (a total of 7Mb/s). The aggregate traffic exceeds the theoretical maximum throughput of an IEEE802.11 network (approximately 6Mb/s [21]). The two VoIP clients initiate a call under these conditions. The AP operates in IEEE802.11b mode. These scenarios exhibit phenomena of congestion and continuous contention in the wireless channel.

3) Heavy TCP traffic: In this 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 the chunks and uploads the already acquired ones to other peers. 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 behaviour puts stress on the queue, CPU, and memory of the AP.

B. Measurements and evaluation

We performed a number of VoIP calls for each of the aforementioned scenarios, in which 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, normal AP hotspot, heavy UDP traffic, heavy TCP traffic at the application layer.

To measure the performance of a VoIP call, we first employed the E-model[8], which gives an estimated Mean Opinion Score (MOS) value. We also performed an MOS evaluation using the PESQ tool. Like E-model, PESQ also utilizes the packet loss information of the VoIP calls. In both evaluation studies, the same VoIP calls were used. In the next paragraphs, we first describe the E-model and the E-model MOS performance results, and then, present the PESQ, the PESQ (MOS), and a comparative analysis study using these models.

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 ITU-T provides an R-to-MOS conversion formula.

The term R_o accounts for the basic signal-to-noise ratio the user receives and takes into consideration the loudness of the voice, and the noise introduced by the circuit and background sources. The term I_s represents voice specific impairments, such as too loud speech level (non-optimum OLR), non-optimum sidetone (STMR), and quantization noise (qdu), 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 users understanding of trading voice quality over convenience. All these factors have been extensively analyzed in ITU-T's G.107 recommendation (E-model).

We performed two set of calls under each scenario: The second set of calls was used to validate the results of the first one. Also, a part of the calls of the second set was used in the statistical analysis with ANOVA and Student's T-test. The number of calls for each scenario/set is indicated in the caption of Figures 1, 2, and 3. We observed two types of handover, namely the *fast handover* and the *handover with deauthentication*. During a fast handover, the overall quality is close to excellent with a small degradation when minor packet losses or delays occur. During a handover, the client initiates an active AP discovery during which packets are queued up. If a handover with deauthentication occurs, the inter-AP protocol will not handle the pending packets at the old AP. Thus, the error rates and unacknowledged retransmissions will increase,

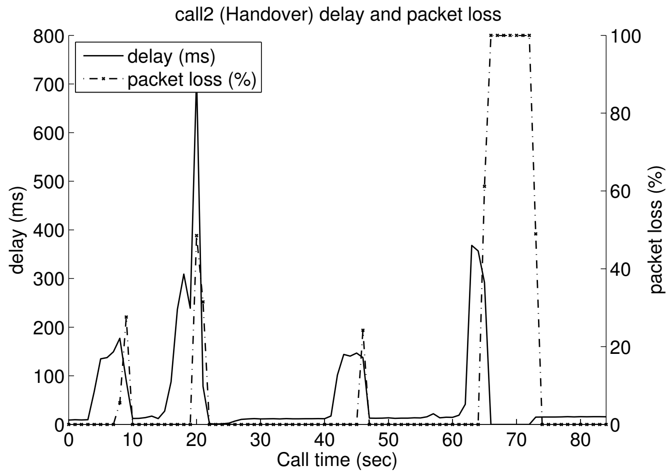

Fig. 4. A VoIP call under handover.

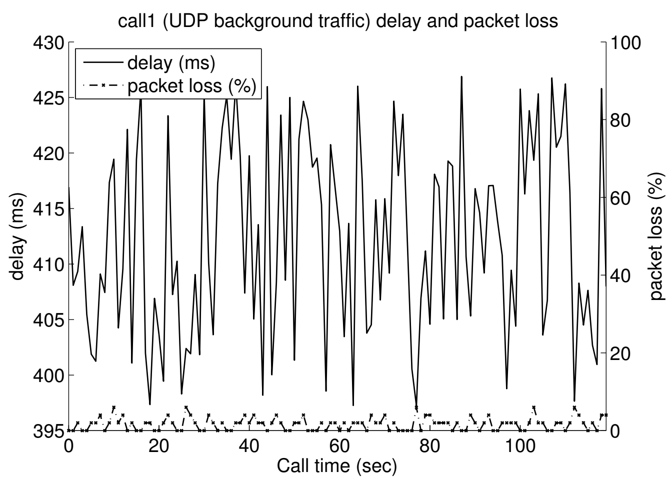

Fig. 5. A VoIP call under heavy UDP traffic.

and as a result, the degradation of MOS will become more prominent.

Both VoIP calls under the normal hotspot AP traffic enjoy near packet-loss free transmissions with low delay. (Due to lack of space, we omit information about this scenario. More information can be found in [22]). In this scenario, there are no saturation conditions: even though the background traffic corresponds to traces collected from a hotspot campus-wide AP (with the largest amount of traffic among all APs in that campus), the generated flows are of small size and are rarely simultaneous. Their small size results in low throughput, since the TCP slow start algorithm does not reach a large window size. On the other hand, unlike in the heavy TCP traffic scenario (BitTorrent case), there is a relatively small number of flows that transmit simultaneously, leaving the network underutilized. Even if a small number of flows fully utilized the AP bandwidth, the TCP congestion control mechanism would limit its throughput, and the small UDP VoIP flows would be able to "sneak through".

In the heavy UDP traffic scenario (e.g., Figure 5), the MOS

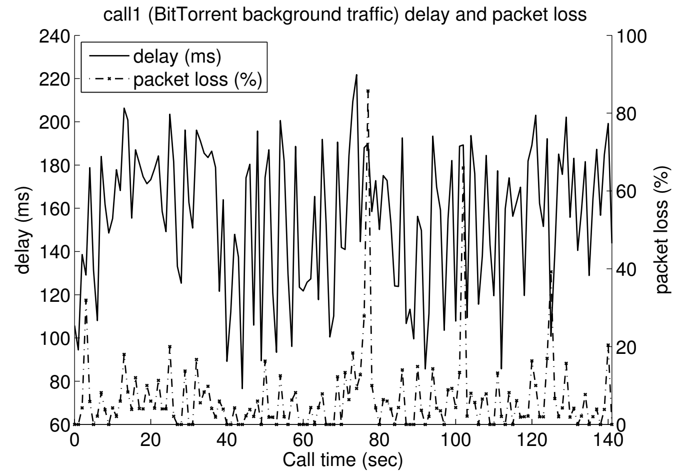

Fig. 6. A VoIP call under heavy TCP traffic.

deteriorates due to the high packet delays. In this scenario, the large delays are due to the presence of heavy background traffic resulting in a packet arrival rate higher than the effective bandwidth of the AP (also observed in other studies, e.g., [23]). 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 quality of these VoIP calls is mediocre. 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., [5]).

Under heavy TCP traffic, the calls suffer from relatively high packet losses and delays. Although packet delays exceed the 150ms threshold, the overall voice quality is acceptable (in disagreement with the "rule-of-thumb"). 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. This phenomenon will be also manifested with the other flows that are active at that time. Since the number of flows at any given time is relatively large, this behavior 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.

Figures 4, 5 and 6 show three examples of VoIP calls under the handover, heavy UDP traffic (*UDP-8*) and heavy TCP traffic, respectively. Table I summarizes statistics about the end-to-end delay, packet loss and (E-model) MOS values for different VoIP calls and scenarios. More interestingly, it illustrates several cases (e.g., calls under the handover scenario) in which the criterion of an average 150s end-to-end delay for a tolerable VoIP quality may not be an adequate metric. Note the high burstiness of packet losses in handover compared to the significantly lower burstiness of packet losses but higher mean delay and packet loss (%) in the heavy TCP

TABLE I
EXAMPLES OF VOIP CALLS UNDER VARIOUS SCENARIOS.

Scenario/Network conditions	Mean delay (ms)	Std delay (ms)	Packet loss (%)	Std packet loss (%)	BurstR	E-model MOS
Handover	24.23	40.43	8.66	26.61	201.40	3.07
Handover	55.99	72.20	8.12	25.53	131.35	3.14
Handover	32.25	81.67	8.51	26.97	198.55	3.09
Handover	58.91	71.66	8.22	26.90	394.68	3.11
Heavy UDP traffic (*UDP-8*)	499.23	11.4	2.57	2.09	1.78	2.57
Heavy TCP traffic	96.28	104.10	11.24	15.42	2.11	2.86

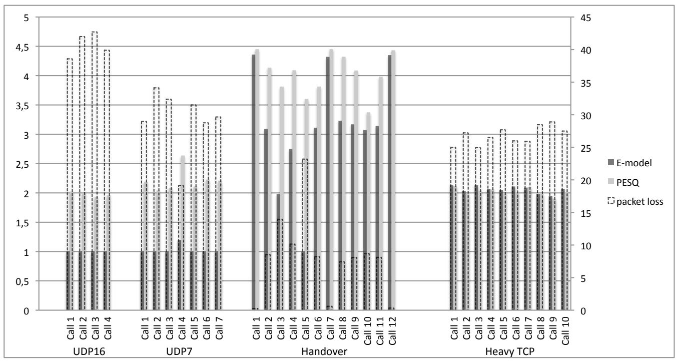

Fig. 7. Summary of the packet loss, PESQ MOS, and E-model MOS for each VoIP call (Set 2).

traffic scenario.

To extend our assessment with additional quality metrics, we also employed the PESQ test [7]. As mentioned earlier, E-model takes into account both delay and packet losses. In PESQ, we focus 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 6s to 20s duration. The former requirement proved to be difficult when comparing recordings, so we opted to employ the effects of packet loss and *disregard any delay*. The packet loss data from different scenarios was used with three different coding schemes, namely G.711 64kb/s, AMR codec 6.7kb/s, and AMR 12.2kb/s.

For the assessment with PESQ, 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 that corresponds to a transmission without any packet loss. 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, packet loss locations were simply removed from the pulse-code modulation (PCM) audio, whereas with AMR codecs, 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 10s frames with 1s “step size” (sliding window) for the entire call duration. The metric for a call was the *average of all MOS values* reported for that call.

Figure 7 provides further statistics for a larger number of VoIP calls (calls from set 2) in terms of packet losses and E-model and PESQ MOS. It shows a prominent discrepancy between the E-model MOS and PESQ MOS. In addition, it illustrates that the impact of the network conditions on MOS varies. (More detailed information can be found in [22].) To obtain more conclusive results about these phenomena, we statistically analyzed the impact of these codecs and scenarios on the user-perception metrics.

To investigate the *statistically significant factors* of the MOS values, a two-way ANOVA was performed. In this analysis, all calls of the second set that belong to the handover, heavy UDP traffic *UDP-7* and *UDP-16*, and heavy TCP traffic scenarios were used. The PESQ MOS was used as the user-perception metric. The dependent variable is the average PESQ MOS value of each call and the independent variables are the scenario and codec types (being the most significant source of variance). The analysis with ANOVA indicates that both *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 revealed the following:

1) the handover exhibits higher MOS values than all other scenarios
2) the heavy TCP traffic performs similarly as the heavy UDP traffic (in terms of MOS)
3) the *UDP-16* results in a lower MOS value than the *UDP-7* (the analysis indicates non-significant differences)
4) the performance of AMR 6.7kb/s is similar to the performance of G.711 64kb/s (lower data rate vs. concealment tradeoff)
5) 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 more sophisticated packet loss concealment of AMR 12.2kb/s justifies its better performance. The similar performance of AMR 6.7kb/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 comparative performance of the three codecs assumed the same packet losses. In a real life experiment where the codecs would actually be used (as opposed to being emulated), the performace of these codecs would be subject to different packet losses. The PESQ MOS of each call is the average MOS value that corresponds to each 10s frames of the call (using a sliding window of 1s). We plan to perform subjective tests to verify the impact of bursts of low MOS values (and their relative position during the call) on the user perception (as indicated in [6]).

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 with all scenarios. The test indicates *statistically highly significant* ($p < 0.01$) differences between the estimations of the two models.[1] Finally, in the context of heavy losses, as the AMR codec tests show, it is beneficial to increase the codec bit-rate. In addition, packet loss concealment and error correction algorithms could be further improved to provide even more adaptivity. The prediction of the delay and packet loss probability would enhance the adaptivity of the encoding.

IV. Conclusion and future work

The paper demonstrates situations in which common "rule-of-thumb" metrics (e.g., the "150ms threshold of the mean end-to-end delay") cannot reflect the perceived user experience and better benchmarks are required. Examples of such situations are handovers with long bursts of packet losses or heavy TCP background traffic. Also, network conditions that correspond to low VoIP quality exhibit different statistical characteristics in terms of packet loss and delays. The analysis with ANOVA indicates that both *scenario* and codec type, as well as their interaction, have a *significant effect* on the PESQ MOS values. A comparison of the E-model with PESQ MOS values, using the Student's T-test, showed highly significant differences in their estimations.

We plan to extend this work in the following directions: First, it would be interesting to perform a comparative study using QoS-enabled APs and SIP-enabled devices. Second, we will investigate the use of the E-model to estimate only the perceptual effect of the delay, and combine it with the PESQ to form a new user-perception model. Distinguishing these different factors (delay and packet losses) and their temporal properties may also be beneficial in the codec design/adaptation. An important next step is the performance of subjective listening tests under the aforementioned network conditions. It would be interesting to statistically analyze the ability of various cross-layer measurements, such as MAC retransmissions and delays, type of control packets, and SNR values, to predict when the MOS will reach a certain threshold and the current network condition in order to trigger an adaptation process. For example, a continuous increase of the average delay in the presence of UDP background traffic will likely result in low MOS. A smoother degradation in the MOS is more likely in the presence of a few TCP flows as background traffic. 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 and in enabling the wireless device to select the appropriate adaptation. Finally, this work sets a methodology framework for performing similar statistical comparative analysis for other applications and networks.

[1] Due to lack of space, we omit the details of the ANOVA and Student's T-test analysis. Detailed information can be found in our technical report [22].

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