The Combined Effect of Signal Strength and Background Traffic Load on Speech Quality in IEEE 802.11 WLAN

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Abstract. *This paper deals with measurements of the combined effect of signal strength and background traffic load on speech quality in IEEE 802.11 WLAN. The ITU-T G.729AB encoding scheme is deployed in this study and the Distributed Internet Traffic Generator (D-ITG) is used for the purpose of background traffic generation. The speech quality and background traffic load are assessed by means of the accomplished PESQ algorithm and Wireshark network analyzer, respectively. The results show that background traffic load has a bit higher impact on speech quality than signal strength when both effects are available together. Moreover, background traffic load also partially masks the impact of signal strength. The reasons for those findings are particularly discussed. The results also suggest some implications for designers of wireless networks providing VoIP service.*

Keywords

IEEE 802.11, signal strength, background traffic load, speech quality, Perceptual Evaluation of Speech Quality (PESQ).

1. Introduction

Wireless multimedia transmission across Wireless Local Area Networks (WLANs) has been gaining attention in the recent years because of the proliferation of technologies like Bluetooth, IEEE 802.11, 3G, and WiMAX. In particular, IEEE 802.11 WLAN [1] has emerged as a prevailing technology for (indoor) broadband wireless access because it supports real-time conversational multimedia applications like Voice over Internet Protocol (VoIP) and video conferencing.

Quality of Service (QoS) support for Voice over WLAN (VoWLAN) is an important issue for technical and commercial reasons. However, speech quality for VoWLAN suffers from high packet loss rates and other impairments in the wireless link. Hence, it is essential to enhance the QoS support capability of current WLAN standards, such as the most popular IEEE 802.11 standard. Before doing it, we should be aware of all potential im pacts of the combination of some impairments on speech quality in IEEE 802.11. The combined impact of signal strength and background traffic load seems to be one of the most frequently occurring effects in current networks.

Speech quality is judged by human listeners and hence it is inherently subjective. The Mean Opinion Score (MOS) test, defined by ITU-T P.800 [2], is widely accepted as a norm for speech quality assessment. Subjective testing is expensive and time-consuming. That is the reason that subjective testing is impractical for frequent testing such as routine network monitoring. Objective test methods have been developed in recent years. They can be classified into two categories: signal-based methods and parameter-based methods. Intrusive signal-based methods use two signals as the input to the measurement, namely, a reference signal and the degraded signal, which is the output of the system under test. They identify the audible distortions based on the perceptual domain representation of two signals incorporating human auditory models. These methods include Perceptual Speech Quality Measure (PSQM) [3], Measuring Normalizing System (MNB) [4], [5], Perceptual Analysis Measurement System (PAMS) [6], and Perceptual Evaluation of Speech Quality (PESQ) [7], [8]. Among them, PSQM and PESQ were standardized by ITU-T as P.861 [9] and P.862 [10] respectively. In contrast to intrusive methods, the idea of the single-ended (non-intrusive) signal-based methods is to generate an artificial reference (i.e., an "ideal" undistorted signal) from degraded speech signal and to use this reference in a signal-comparison approach. Once a reference is available, a signal comparison similar to the one of PESQ can be performed. The result of this comparison can further be modified by a parametric degradation analysis and integrated into an assessment of overall quality. The most widely used algorithms include Auditory Non-Intrusive Quality Estimation (ANIQUE) [11] and standardized P.563 [12], [13]. Parameter-based methods predict the speech quality through a computation model instead of using real measurement. E-model is the typical model, defined by ITU-T Recommendation G.107 [14]. The E-model includes a set of parameters characterizing end-to-end voice transmission as its input and the output (R-value) can be transformed into the MOS-Listening Quality Estimated narrow-band (MOS-LQEn) values.

Some works have studied the impact of node location and the combined effect of signal strength and traffic type on the performance of IEEE 802.11 WLANs. In [15], authors examined the quality of links in home wireless networks and effect of transmission rate, transmission power, node location, type of the house and 802.11 technology. Their results clearly confirm that distance has no impact on the quality of wireless links in the home, while small changes in antenna orientation and node location can dramatically change the performance of an individual link. They also found little difference between network performances at a moderate transmit power and the maximum transmit power as well as that topology (direct communication, Access Point (AP) topology and mesh topology) has largest impact on overall network performance in the home. In [16], the combined effect of received signal strength and traffic type (video and audio) on the performance of WLAN was investigated. The results show that signal strength has a significant impact on the playback delays of audio and video traffic, especially under higher bit rates. In addition, they studied the relationship between wireless channel state and the user-perceived quality of audio and video streaming applications. Unfortunately, it was just done in non-standard way. It looks like that the authors only assessed the impaired samples without following the important recommendations and approaches (informal evaluation). This fact makes their quality results not reliable.

As can be seen above, any work available did not investigate the impact of either signal strength or the combined effect of signal strength and background traffic load on speech quality in IEEE 802.11 WLAN. Finally, I decided to investigate the combined effect because of more complex scenario and impacts. Such scenario is much closer to reality than the first one therefore it can provide us more realistic data. In particular, this work investigates how the different values of signal strength and background traffic load can affect speech quality in IEEE 802.11 WLAN. The ITU-T G.729AB [17] encoding scheme is deployed in this experiment and the Distributed Internet Traffic Generator (D-ITG) [18] is used for the purpose of background traffic generation. The speech quality and background traffic load are assessed by means of the accomplished PESQ algorithm and Wireshark network analyzer [19], respectively. One can argue why this analysis is going to be done by PESQ, instead of using listening test results. There are several reasons to deploy PESQ as a useful tool for this analysis. Firstly, PESQ is recommended by ITU-T to assess the speech quality in wide range of network conditions. Moreover, currently this algorithm is mainly used for such measurements around the globe and widely accepted by community for this purpose because of its very good accuracy in recommended conditions. Secondly, several studies have investigated the performance of PESQ in wireless environment as well as with regard to coding impairments, for instance [20], [21] and [22]. All studies have reported very good performance of PESQ under both conditions, namely wireless environment and coding impairments. Thirdly, PESQ enables us to do very comprehensive study of the combined impact of those parameters (signal strength, background traffic load) on MOS-scores, which probably would not be feasible by doing only listening tests, because of known limitations of such tests (e.g. duration of test, number of samples presented without subjects fatigue, etc.). Such comprehensive study is really required in such conditions as wireless environment produces. Finally, we believe that all aforementioned reasons (high acceptability and accuracy, comprehensive study, etc.) allow us to deploy this algorithm in this case.

The rest of the paper is organized as follows. Section 2 describes the experimental scenario. Section 3 presents the experimental results. Section 4 concludes the paper and suggests some future studies.

2. Experimental Scenario

2.1 Experimental Setup

One-way VoIP session was established between two hosts (wireless VoIP Sender and VoIP Receiver), via the AP, in IEEE 802.11b WLAN (see Fig. 1). Two stations (ITG Sender and wireless ITG Receiver) equipped with the accomplished D-ITG traffic generator were used to generate and receive background traffic. ITG Sender generated the User Datagram Protocol (UDP) packets of length of 1024 bytes. For this experiment, the 0, 3 and 5 Mbps background traffic loads with Poisson distribution packet rate were deployed. The background traffic load was measured by means of Wireshark network analyzer [19]. The reasoning behind choosing UDP and not Transmission Control Protocol (TCP) as a transport protocol for carrying background traffic is threefold: 1) UDP background traffic gives more accurate estimate of the actual load in the network (no retransmissions at transport layer); 2) results obtained with UDP constitute an upper bound for the throughput possible with TCP; 3) retransmissions of lost or corrupted packets is done by the 802.11 MAC-layer so TCP do not get affected by the packet loss [23], [24].

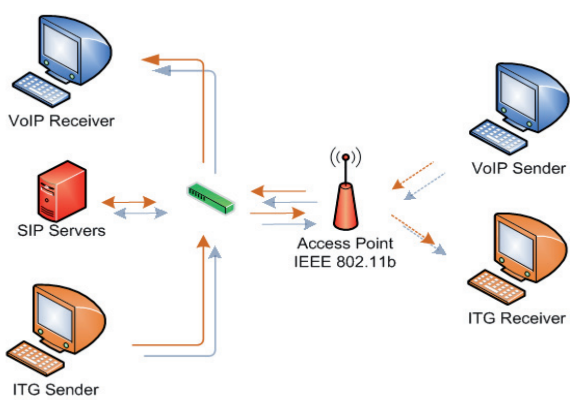

Fig. 1. Experimental setup.

Voice traffic was generated using VoIP clients. Both types of traffic (voice and background) were mapped into same priority class (no prioritization used). Session Initiation Protocol was used for established VoIP connection. For the measurement the ITU-T G.729AB [17] encoding scheme was chosen. In the measurement, two frames were encapsulated into a single packet; thus corresponding to a packet size of 20 ms. Adaptive jitter buffer, G.729AB's native Packet Loss Concealment and Voice Activity Detection/Discontinuous Transmission were implemented in the VoIP clients used.

The measurements were performed for four signal strength values (50%, 70%, 80% and 100%) and three background traffic loads (0, 3 and 5 Mbps) for each signal strength value. The investigated values reflect the values usually obtained in real-life. Moreover, I decided to avoid much extreme values because such values can cause a very unreliable transmission or a disruption of communication. The signal strength was only changed for VoIP Sender; this wireless terminal changed position to obtain the signal strength values mentioned above. When the desired value was obtained, this value was kept fixed during all measurements realized at that position (for all reference signals, all repetitions as well as for all investigated background traffic loads). The second wireless terminal had a fixed position (no mobility) during all measurements and the signal strength was also kept fixed (100%) for all performed measurements. The AP parameters were not changed during measurements and default values were applied in this case. The reference signals described in Section 2.2 were utilized for transmission through the given VoIP connection. Finally, speech quality was assessed by currently accomplished PESQ algorithm [7], [8] and [10], the most recent ITU-T standard for objective speech quality assessment and then converted to MOS-Listening Quality Objective narrow-band (MOS-LQOn) values by this equation:

$$y = 0.999 + \frac{4.999 - 0.999}{1 + e^{-1.4945 * x + 4.6607}}, \qquad (1)$$

where x and y represent the raw PESQ score and the mapped MOS-LQOn, respectively. The equation mentioned above is defined by ITU-T Recommendation P.862.1 [25].

Moreover, the accuracy of measurement was improved by addressing the following issues:

- People movement: The measurements were conducted during the weekends and night hours to avoid the impact of the movement of people on system performance.
- Co-channel interference: During measurements, no co-interference APs was available in measurement area (area covered by AP).
- Validation: The measurements were repeated 40 times (4 reference signals, each 30 seconds long, 10 repetitions per each signal) under same signal strength and background traffic load value to obtain repeatability of results, ensuring the correctness of measured data collection.

2.2 Reference Signals

The reference signals selection should follow the criteria given by ITU-T Recommendations P.830 [26] and P.800 [2]. The reference signals should include bursts separated by silence periods. They are normally of 1-3 seconds long, although this does vary considerably between languages. They should be active for 40-80% of their duration. The reference signals are composed of speech records. In this experiment, these speech records were taken from a Slovak speech database. In each set, two female and two male speech utterances were used. The reference signals, which are of 30 seconds long with 57 % average value of active speech interval, were stored in 16-bit, 8000 Hz linear PCM. Background noise was not present.

3. Experimental Results

The measurements were independently performed 40 times under same signal strength value and background traffic load, 10 times for each type of used reference signals in order to have sufficient amount of data from statistical perspective. The obtained MOS-LQOn values were averaged.

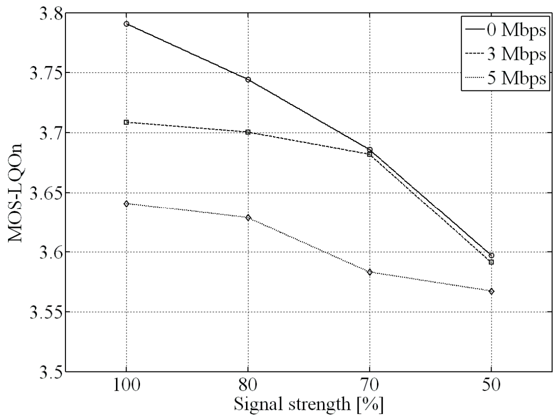

Fig. 2. MOS-LQOn as a function of signal strength and background traffic load.

In Fig. 2, it can be seen that the signal strength changing (smaller values of signal strength) has a negative impact on overall speech quality, expressed by MOS-LQOn (see solid line). The mentioned negative impact is caused by packet loss generated by unsuccessful transmissions. Because of worse transmission conditions (smaller values of signal strength), the terminal tries to resend unsuccessfully transmitted packets. Naturally, there are some limitations with respect to the number of retransmissions. When the terminal reaches the maximum number of retransmissions (normally around 7) and the packet is still not successfully delivered, such packet is dropped. On the other hand, jitter can rise by retransmission. It is widely known that higher values of jitter can have a negative impact on speech quality. In current VoIP clients, jitter can be compensated by adaptive jitter buffer. The length of buffer is commonly updated by actual delay and jitter values, from time to time. If the packet arrives after playout time (length of buffer), this packet is dropped. More details about jitter buffer schemes can be found in [27]. The dropped packet can be concealed by packet loss concealment algorithm [28]. The concealment can partly alleviate the impact of this kind of loss on speech quality. Moreover, the discussed effect (decrease of quality due to signal strength) is not so evident when the background traffic load was added as additional impact; see dash and dotted lines in Fig. 2. It appears that background traffic load became more influential factor. The background traffic generates collision situations (as mentioned above, both traffics were mapped into same priority class) because of more packets in wireless network and thus more transmission attempts. After the collision, the station has to wait random time (defined by counter on the basis of access procedure used) for contending for the medium. This reduces the probability of next collisions but also negatively influences the jitter (a rise of jitter achieved). As also mentioned above, higher values of jitter can have a negative impact on speech quality but this impact can be partly compensated by adaptive jitter buffer and packet loss concealment algorithm. Finally, it looks like that the background traffic load partially masks the impact of signal strength on speech quality.

Effect	SS	df	MS	*F*	*P*
Signal strength (1)	1.1403	3	0.38011	10.87	0.0000
Background traffic load (2)	0.8196	2	0.40982	11.72	0.0000
(1)*(2)	0.1955	6	0.03259	0.93	0.4717
Error	16.3661	468	0.03497		
Total	18.5216	479			

Tab. 1. Summary of ANOVA conducted on MOS-LQOn values.

Two-way analysis of variance (ANOVA) was conducted on MOS-LQOn values using signal strength and background traffic load as fixed factors (see Tab. 1). The highest F-ratio was obtained for the background traffic load (F = 11.72, p < 0.01). Moreover, the signal strength showed a little bit weaker effect on quality than firstly mentioned factor, with F = 10.87, p < 0.01. The realized ANOVA test revealed that background traffic load has a bit higher impact on speech quality than signal strength. On the other hand, there is no evidence of synergistic effect (interaction) of both impairments (F = 0.93, p < 0.01), as clearly expected, because they cannot happen simultaneously. In other words, if the packet is dropped due to the signal strength, the same packet can not also be dropped due to collision generated by background traffic load. Contrariwise, if the packet is successfully transmitted, this packet can still be dropped due to collision. Finally, ANOVA results support the assumptions presented in previous clause, namely that the background traffic load is more dominant factor (higher F-ratio) as well as partially masks the impact of signal strength (dominance of background traffic load and no interaction effect of the two).

On the basis of the results, I can pronounce that speech quality in WLAN environment is a bit more impaired by background traffic load than by signal strength when they are available together. Moreover, the higher-impact parameter (background traffic load) partly masks the impact of the second parameter (signal strength).

4. Conclusion and Future Work

This paper investigated the combined impact of signal strength and background traffic load on speech quality in IEEE 802.11 network. The speech quality was assessed by means of the accomplished PESQ algorithm. Experiment shows that background traffic load has a bit higher impact on speech quality in WLAN environment than signal strength when both effects are presented jointly. In addition, background traffic load also partially masks the impact of signal strength.

The results have some implications for the designers of wireless networks providing VoIP service. As mentioned before, background traffic load partly masks the impact of signal strength. Probably, there are also other combined effects which have a similar behavior. This allows designers to mainly focus on higher-impact parameters (e.g. background traffic load in this case) while doing the cross-layer optimization.

A future work will focus towards the following issues. Firstly, I would like to investigate other combined effects in IEEE 802.11 WLAN. Secondly, I plan to design VoIP link adaptation algorithms based on the cross-layer optimization approach and combined effects.

Acknowledgement

The work has been supported by the VEGA grant agency, project No. 1/0313/08, *The investigation of the methods of detection of the critical conditions in telecommunication networks from the speech quality point of view.*

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