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VOICE QUALITY ESTIMATION IN WIRELESS NETWORKS

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Abstract

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This article deals with the impact of Wireless (Wi-Fi) networks on the perceived quality of voice services. The Quality of Service (QoS) metrics must be monitored in the computer network during the voice data transmission to ensure proper voice service quality the end-user has paid for, especially in the wireless networks. In addition to the QoS, research area called Quality of Experience (QoE) provides metrics and methods for quality evaluation from the end-user's perspective. This article focuses on a QoE estimation of Voice over IP (VoIP) calls in the wireless networks using network simulator. Results contribute to voice quality estimation based on characteristics of the wireless network and location of a wireless client.

Keywords: QoE, voice, VoIP, wireless, estimation

INTRODUCTION

The demand for the VoIP services and Wi-Fi networks has been increased significantly in recent years. The wireless Internet connection becomes a standard in companies, public spaces and households as well. Besides the traditional data, more voice and video services traverse converged networks. Traffic of these multimedia services is more sensitive to network distortions in comparison with traditional data traffic. These network traffic impairments occur especially in wireless networks. A high quality design of the wireless network should be performed before the Wi-Fi deployment in the case of VoIP providing. The appropriate signal coverage should be ensured to prevent the VoIP quality degradation. On the other hand, the implementation cost on wireless infrastructure influenced mainly by number of access points (AP) may not be too high. The necessary number of APs can be easily obtained when the VoIP call quality behaviour in the wireless network is known. The VoIP call quality will be evaluated using the QoE methods that apply end-user perspective. The elementary QoE metric is MOS (Mean Opinion Score), a qualitative value, where the value of 5 means excellent and 1 means bad.

The aim of this paper is to propose an approach how to estimate VoIP call quality estimation in Wi-Fi networks. Results can contribute to VoIP quality assurance and prevent companies from wasting unnecessary cost on wireless network implementation.

MATERIALS AND METHODS

Related Work

The area of QoE and factors influencing the service quality provided in the converged networks are thoroughly described by Stankiewicz and Jajszczyk (2011). A detailed introduction to the QoE measurement techniques is discussed by Schatz *et al.* (2013) including the subjective (human participation) and objective (no human participation) QoE assessment methods, e.g. the objective method PESQ (ITU-T, 2001) or other approaches based on neural networks (Lýsek *et al.*, 2013).

The impact of the wireless physical layer on the VoIP QoE is examined by Sanchez-Iborra *et al.* (2013). The authors performed several simulations to examine the impact of distance between two wireless nodes on the delay, packet loss and MOS of VoIP calls. They focused mainly on the capacity performance of the IEEE 802.11g network, while the necessary quality of VoIP calls (3.1 MOS) had to be achieved. The VoIP codecs G.711 and G.726 were tested. The E-model was chosen as the objective QoE evaluation tool. Mullins *et al.* (2009) deal with voice and video capacity in the IEEE 802.11g network. Real measurements are performed and subjective quality assessment is used. De Moore *et al.* (2010) related the subjective assigned MOS values to the Signal-to-Noise ratio.

Signal attenuation can be simulated by several propagation models published e.g. by Burbank *et al.* (2013). The propagation model can be customized using the α value that can be seen in Free Space Loss formula (FSPL) below where *F* represents a used frequency and *D* represents a distance between two nodes. Recommended α value of propagation models are discussed by Nwizege *et al.* (2013) and Faria (2005). One can say that α value less than 2 represents line-of-sight and α greater than 2 imitates no-line-of-sight situation. Still, there will be some disproportion between the real and simulated environment. Signal distortion caused by obstacles and human bodies is a nontrivial problem as discussed by Fet *et al.* (2013).

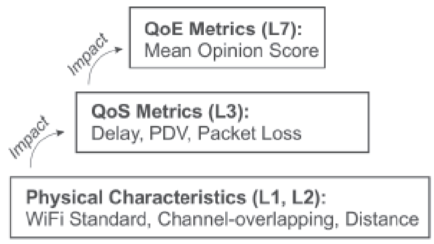
$$FSPL = 32.44 + 20 \cdot \log_{10} F + \alpha \cdot 10 \cdot \log_{10} \mathrm{D}.$$

As evident from the previous paragraphs, the QoE is an interesting and still developing area of varied approaches. There is plenty of research dealing with QoE of VoIP in wireless networks. However, there was no publication found discussing the estimation approach described below. Moreover, a non-standardized tool AQuA was employed for the objective audio quality estimation in this article. Sevana Oy (2015) provides comparison of AQuA and PESQ accuracy.

Estimation Approach

When examining the cause of the QoE decrease, researches widely relate the QoE indicators to the QoS metrics. On the other hand, there are several publications focusing on the impact of the wireless network configuration on the QoS metrics. In this article, we try to merge these approaches and deliver the solution for QoE estimation of VoIP calls directly from the physical layer (L1) and the datalink layer (L2) attributes.

There are plenty of voice codecs differing in e.g. packetization interval[1], sample size, required bandwidth and the resulting MOS value estimated by PESQ. The attributes of the examined codecs are summarized in Tab. I. In a similar way, there are several wireless standards and an unlimited number of aspects influencing the resulting wireless signal in indoor environment (data rate, modulation, signal reflection, absorption, dispersion, etc). Therefore, the QoE estimation of the selected voice codec based on the particular wireless network conditions is very inflexible because each codec would have to be related to the specific settings of the wireless network and physical environment. These facts directed us to use a quality estimation approach depicted in the Fig. 1. According to this model, we have divided the estimation into two parts. Firstly, the MOS score is estimated from the QoS metrics (QoE-QoS part). Secondly, the relationship between selected QoS metrics (delay variation, loss) and specific wireless network settings (standard, data rate, etc.) can be examined (QoS-WiFi part). In other words, the direct relation between QoE and specific L1 and L2 parameters was divided into two separate parts. Using this approach, the QoE of different voice codecs can be examined based on particular QoS-WiFi measurements. On the other hand, the effect of several measurements in QoS-WiFi part can be easily applied on one selected voice codec obtained in QoE-QoS part.



1: *Estimation model hierarchy*

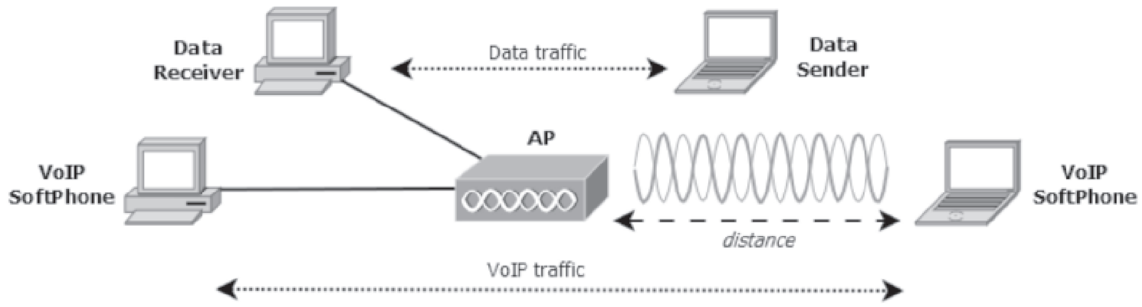
Simulation and Topology Configuration

The simulation process was performed using OMNeT++ simulator, version 4.6, and INET framework, version 2.99 (OMNeT++, 2015). The simulated topology contains a wireless IEEE 802.11g network, while different transmission rates are used – 54 Mbps with ERP-OFDM modulation and 11 Mbps with DSSS modulation. The default Free Space propagation model could be generally employed to simulate an outdoor free-space environment. The indoor conditions can be simulated e.g. by Rice model, that pretend the situation with a line-of-sight between transmitter

I: *Voice codecs characteristics*

Codec	Sample size	Packet size	Packets per sec	Required BW	MOS
G.711	20 ms	160 B	50	64,0 kbps	4.1
G.729	20 ms	20 B	50	8,0 kbps	3.92

1 The packetization interval determines the size of samples contained within a single packet.



2: *Simulation topology*

and receiver. On the other hand, Rayleigh propagation model suits an indoor environment without the line-of-sight between the wireless sender and receiver (Sanchez-Iborra, 2013). Following work considers the Rayleigh propagation model. The impact of the channel fading can be customized using α parameter described in section Materials and Methods, which is configured to the default value of 2 in this simulation. A noise level was set to −95 dBm as recommended by Ciampa (2013).

The simulated network (Fig. 2) consists of one access point (AP), two PCs connected to the AP by an Ethernet cable (100 Mbps) and two laptops (stations – STA) connected to the AP wirelessly. The position of the VoIP SoftPhone station and output signal power of the AP were changed before the particular measurement to explore their impact on QoS metrics – PDV (Packet Delay Variation) and packet loss. The distance ranged between 1 and 500 meters. The transmission power of station's wireless network card was set to 32 mW (cca 15 dBm), the transmission power of the AP was changing before the measurements (1 mW, 3 mW, 5 mW, 10 mW, 50 mW, 100 mW). All the wireless interfaces had the hardware packet queue configured to store 50 frames, retry limit was established to the default value 7, signal sensitivity is set to −85 dBm and the SNIR (Signal to Interference plus Noise Ratio) threshold to 4 dB. The entire topology is a single switched network with 192.168.0.0/24 prefix.

Two different communications were present in the simulations. First pair (VoIP SoftPhones) initiates the VoIP traffic, second pair (Data Sender and Data Receiver) establishes the data traffic. The VoIP flow was simulated by the SimpleVoIPSender and SimpleVoIPReceiver applications available in OMNeT++. To reflect real conditions, VoIP call was established in both directions. Each VoIP flow was emulated by a stream of UDP packets with appropriate size and count per second according to the voice codecs parameters (see Tab. I). Real VoIP data was not used to ensure the modularity (QoE-QoS and QoS-WiFi parts) of the estimation approach proposed in previous section. In other words, parameters of VoIP flow (packet size, packets per second, etc.) can be applied on multiple codecs in the future. The data traffic is represented by ICMP packets generating a bandwidth up to 6 Mbps.

RESULTS

QoE Estimation from QoS Metrics

The recommendation P.800.1 (ITU-T, 2006) defines several basic types of MOS. In this article, we evaluate the QoE level using the MOS-LQO (Listening Quality Objective). The MOS value is calculated by means of an objective model, which aims to predicting the quality for a listening-only test situation. Although the MOS-LQO is usually measured using the PESQ algorithm according to the Rec. P.862 (ITU-T, 2001), we have used a non-standardized method called AQuA provided by Sevana Oy. The entire QoE-QoS measurement process was published by Zach, Pokorny, Balej (2014). Equations below follow this article and describe improved regression models that provide a better accuracy (see Tab. II). The Pearson coefficient (PC) indicates a high correlation between the measured (MOS_M) and estimated (MOS_E) quality. The coefficient of determination (R^2) proofs good model quality. Graphical representation of the QoE quality of voice calls using G.711 and G.729 voice codecs is depicted in Fig. 3.

$$MOS_{G.711} = \frac{1}{0.186908 + 0.0137091 \cdot Loss + 0.00360137 \cdot PDV}.$$

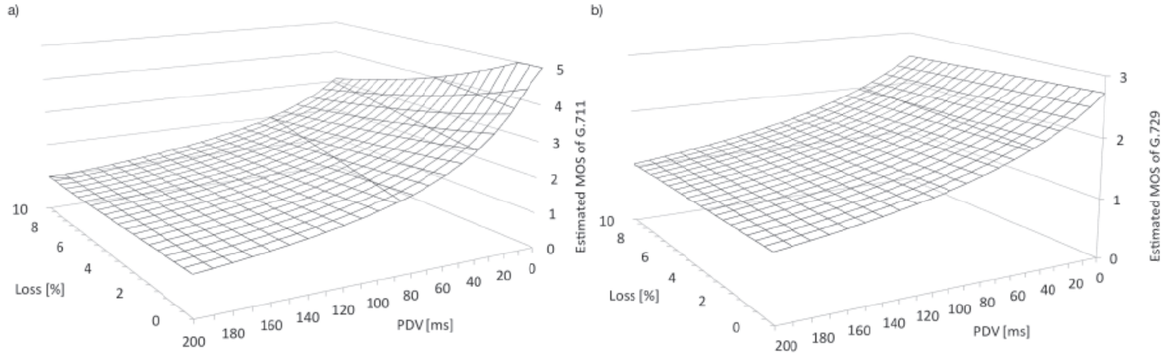
$$MOS_{G.729} = \frac{1}{0.36638 + 0.00278493 \cdot Loss + 0.00282423 \cdot PDV}.$$

Although delay, jitter and packet loss are widely considered as the basic QoS metrics, the measurements proved that the delay is irrelevant in the case of the MOS-LQO estimation. Moreover, jitter is replaced by PDV metric in this article. PDV represents the range between the minimal and maximal detected delay. The transmitted voice was represented with an instrumental music record stored into WAV (16 bit PCM) format. Compared with the speech record, this one has no blank spaces in its waveform to better recognize each quality decrease over the whole record during the measurement process.

As obvious, the resulting regression model for voice codec G.711 reaches higher maximal MOS value than the G.729 voice codec. On the other hand, this model (see Fig. 3a) is more sensitive to

II: *Regression model statistics*

Description	MOS_M/MOS_E PC	R^2	σ of regression	σ of MOS_M
G.711	0.8225	0.7457	0.1296	0.2571
G.729	0.7142	0.7125	0.1088	0.6438



3: *Estimated values of G.711 (a) and G.729 (b) voice calls*

the QoE metrics change, especially the PDV, than the G.729 voice codec (Fig. 3b). Because of the lossy compression of the G.729 codec, the maximal MOS value is noticeably lower in comparison with the G.711 voice codec. Despite this, the QoE score of the G.729 codec reaches lower maximal values than expected. Based on PESQ method, a voice call with G.729 should reach the MOS value of 3.92. The difference in the absolute MOS value of the G.729 voice codec can be caused by different evaluation process used by the AQuA tool or by an unknown inaccuracy in our measurement.

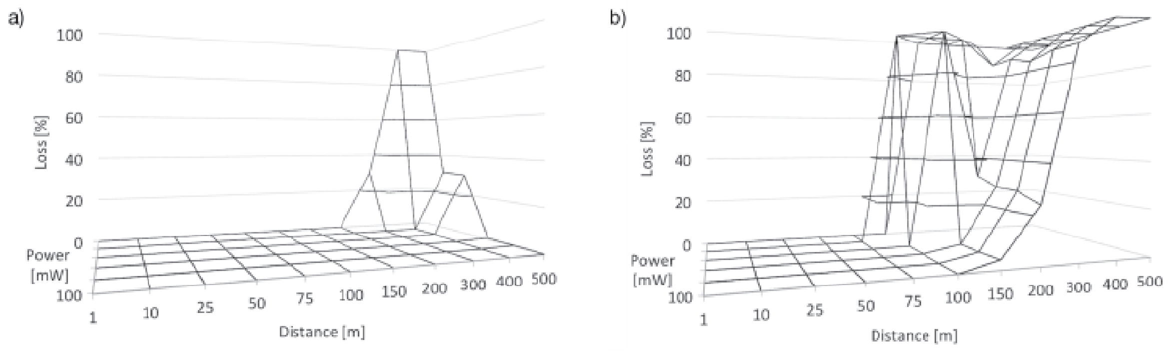
Nevertheless, these models offer a valid non-intrusive approach how to estimate the resulting MOS value based on selected QoS metrics (PDV and packet loss) in real-time. These results are further used in section Resulting models.

QoS Estimation based on Physical and Datalink Layer

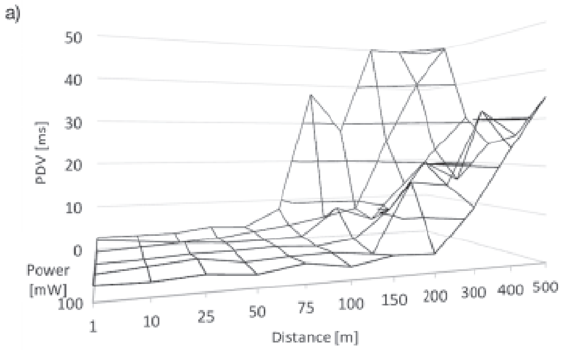
The results of the simulation are depicted in Fig. 4 and Fig. 5. As mentioned earlier, packet loss and PDV were identified as the key QoS metrics used in this work. The aim of simulation process described above is to recognize the impact of selected wireless network characteristics on the packet loss and PDV (the QoS metrics).

Among others, the influence of the data (background) traffic on the VoIP transmission was tested. The simulation proved that the background traffic up to 6 Mbps has no significant impact on the transmitted VoIP data. Therefore, following results consider only the highest simulated data traffic (6 Mbps). Moreover, the detected differences between VoIP flows configured according to G.711 and G.729 voice codec requirements were negligible in the case of their graphical representation, which means that Fig. 4 and Fig. 5 reflect the simulation results of VoIP flow configured based on the G.711 codec parameters only. Nevertheless, the final models in section below take the results of VoIP flow based on G.729 characteristics into account.

Two different data rates of the IEEE 802.11g wireless network were simulated. Diagrams in Fig. 4 indicate the packet loss. In the case of IEEE 802.11g with 11 Mbps data rate, the less complex DSSS modulation is employed that enables the wireless signal to reach longer distances. That corresponds to the results in Fig. 4a. Almost no packet loss was detected among the tested distance



4: *Packet loss of VoIP traffic in wireless network with data rate 11 Mbps (a), resp. 54 Mbps (b)*



5: *PDV of VoIP traffic in wireless network with datarate 11 Mbps (a), resp. 54 Mbps (b)*

and signal output power of the AP. On the other hand, the IEEE 802.11g network configured with the 54 Mbps data rate using OFDM modulation suffers from higher signal attenuation and hence the packet loss. One can say that about one hundred meters distance between the AP and STA VoIP SoftPhone is the maximal distance for successful communication under ideal conditions.

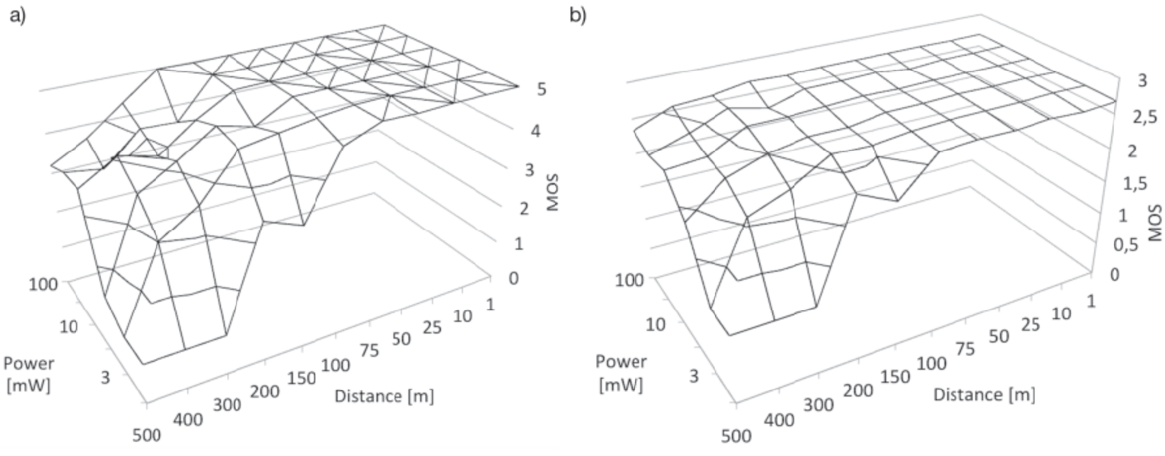
The PDV results are depicted in Fig. 5. As obvious from these diagrams, a non-zero PDV was reached during almost all measured configurations, but the higher PDV values were reached in the cases where the packet loss was increasing (compare with Fig. 4). Moreover, it must be said that the PDV values cannot be quantified because simply no VoIP packet reached its destination in the case of 100% packet loss. Therefore, these PDV values are substituted with very high value, e.g. 1000 ms. When 11 Mbps data rate was employed, the PDV began to grow around 100 m distance between STA VoIP SoftPhone and AP. On the other hand, the IEEE 802.11g network with 54 Mbps data rate (Fig. 5b) reached significantly higher PDV in general, while the significant PDV grow started around 50 m distance.

Resulting Models

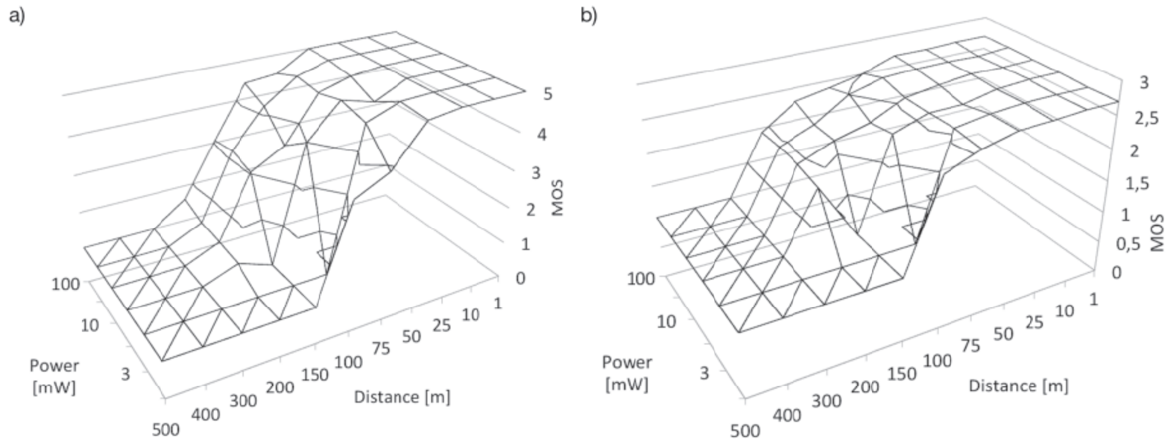
The proposed QoE-QoS models and QoS-WiFi simulation results are combined in following paragraphs to evaluate the resulting estimated MOS value in the wireless network based on estimation approach described above. Simulation output (packet loss and PDV) were filled in the appropriate QoE-QoS model. Overall, four model have been created because two VoIP codecs (G.711 a G.729) and two Wi-Fi data rates (11 Mbps and 54 Mbps) were tested.

The models in Fig. 6 depict the behaviour of G.711 (Fig. 6a) and G.729 (Fig. 6b) voice codecs in the IEEE 802.11g network configured with 11 Mbps data rate. Compared with the MOS value of the G.729 voice codec, G.711 reaches higher MOS under ideal wireless conditions, but the perceived quality decreases with the larger distance and lower output signal power.

The dependence of VoIP call quality in wireless network with 54 Mbps data rate is described in Fig. 7. According to higher values of the QoS metrics detected in this network configuration, the overall perceived quality decreases earlier. In the case of the G.711 voice codec, the maximal MOS value (5) can be achieved only up to circa 25 meters from the AP. Once the STA moves around 100 meters



6: *The impact of wireless network characteristics with 11 Mbps bitrate on the estimated MOS value of voice codec G.711 (a), resp. voice codec G.729 (b)*



7: *The impact of wireless network characteristics with 54 Mbps bitrate on the estimated MOS value of voice codec G.711 (a), resp. voice codec G.729 (b)*

away from the AP, the MOS value reaches the worst available value (1). Assuming that the minimal acceptable MOS value is 3.1 for the end user, the maximal distance is located about 75 meters from the AP. On the other hand, the G.729 voice codec is more resistant to the QoS impairment than G.711. The optimal MOS value begins to decrease about 50 meters from the AP. At the same time it must be said that the maximal possible MOS value of the G.729 VoIP codec was relatively small, thereby the G.729 delivers lower overall quality experience to the end user.

DISCUSSION AND CONCLUSION

The results presented in this paper are based on deterministic network simulator. Each configuration is based on limited range of variables. However, the real physical environment can be influenced by significantly more aspects, which couldn't be covered by the network simulator entirely. Therefore, one has to take into account that the actual conditions can differ from the results presented above in the real environment. Moreover, the wireless network conditions may slightly vary over the time.

The wireless network simulation considered the IEEE 802.11g standard. Nowadays, the g standard is mostly replaced with the IEEE 802.11n standard providing significantly higher throughput. Unfortunately, the MIMO (Multiple-input multiple-output) method used by the IEEE 802.11n standard has not been entirely covered by available network simulators yet.

During the QoE-QoE phase, the QoE estimation models were proposed based on a set of measurements where instrumental music was streamed over the experimental network. The future work will focus on QoE estimation models based on human voices. Moreover another VoIP codecs can be examined. Fixed networks contain a complex set of QoS mechanisms. On the other hand, IEEE 802.11e (called WME, Wireless Multimedia Extensions) is the only one widely used QoS mechanism in Wi-Fi networks. Also the effect of WME on the perceived service quality will be the part of future work.

This article focuses on the impact of the wireless network characteristics on the VoIP call quality perceived by the service consumer. Presented results can contribute to the wireless networks design. Benefit of the proposed models is twofold. Firstly, better QoE of VoIP service can be delivered to end users. Secondly, economical aspects can be targeted during the wireless network design considering the proposed models.

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