Qos and MOS in the VoIP signalling

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ABSTRACT: *In the current work we have given the algorithms for VoIP signalling. While doing this task we have outlined and described the features such as jitter, packet delays, distribution of packets, signal delays and the protocol assignments. Besides we have presented the Quality of Service as well as the MOS*

Keywords: Codec, Packet, Signalization, VoIP

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1. Introduction

The main objective of this paper is about solving problems of the implementation of voice applications over wireless networks with packet switching, the ability to overcome them and to optimize this type of communication. The Extended IP structures, including wireless, provide large opportunities for the development of VoIP technologies. The paper concerns the methods to improve the quality of service, which most closely affect the end user. Concrete results are obtained proving the future application of these technologies. Studies conducted show that the problems associated with the range of wireless networks and their security cannot be a real obstacle to the implementation of Vo802.11.

The thesis which is propounded, demonstrates the broad possibilities for the implementation of VoIP over 802.11 and the provision of new services to existing ones [1].

One of the main challenges in the wireless delivery of data and Internet is the providing of a quality service. QoS is associated also with the ability of service providers to adapt their networks for transmission of voice (Figure 1).

To properly analyze the possibilities of using Vo802.11, as an alternative voice communication, it is necessary to analyzepotential weaknesses of the network to be minimized or eliminated. Largely functions and applications of network depend on signalization. With the use of modern VoIP protocols and good programming of new functions, variety of applications, satisfying the end user are achieved.

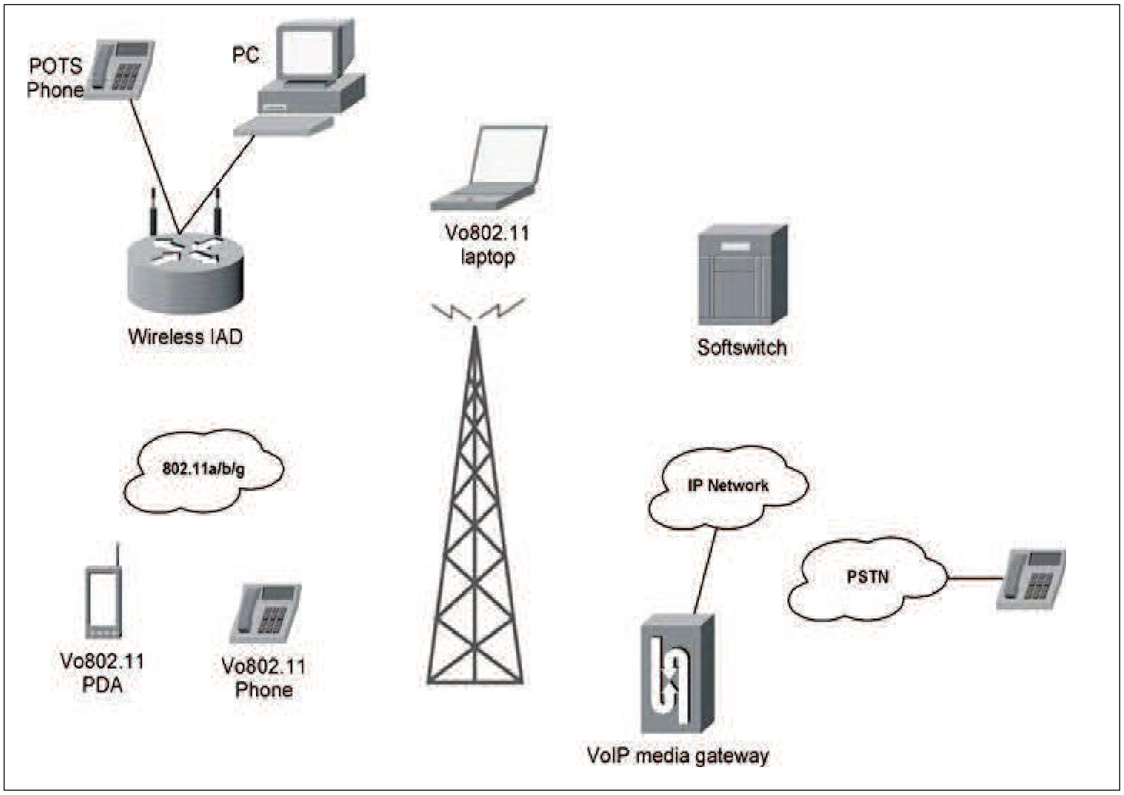

Figure 1. Broadband network for speech transmission

2. Experimental Results

To provide quality speech comparable to the PSTN, it is necessary to minimize delays, jitter and packet loss in Vo802.11 networks. Regulating QoS to the wireless broadband network is accompanied by additional network requirements.

To measure the voice quality currently available two tests to ensure metric: MOS (mean opinion score) and PSQM (perceptual speech quality measurement) (Figure 2).

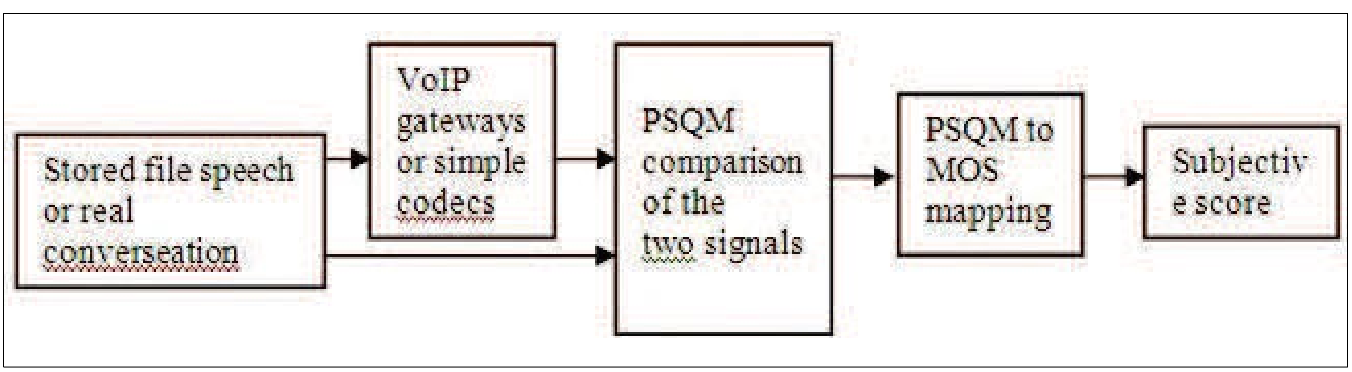

Figure 2. A Process of PSQM [2]

Investigation on IP traffic and signaling tests which were made, were carried out under the scheme of Fig. 3.

Two simultaneous calls from endpoint 192.168.1.3 to 192.168.1.4 were utilized and the results were seen within two minutes.

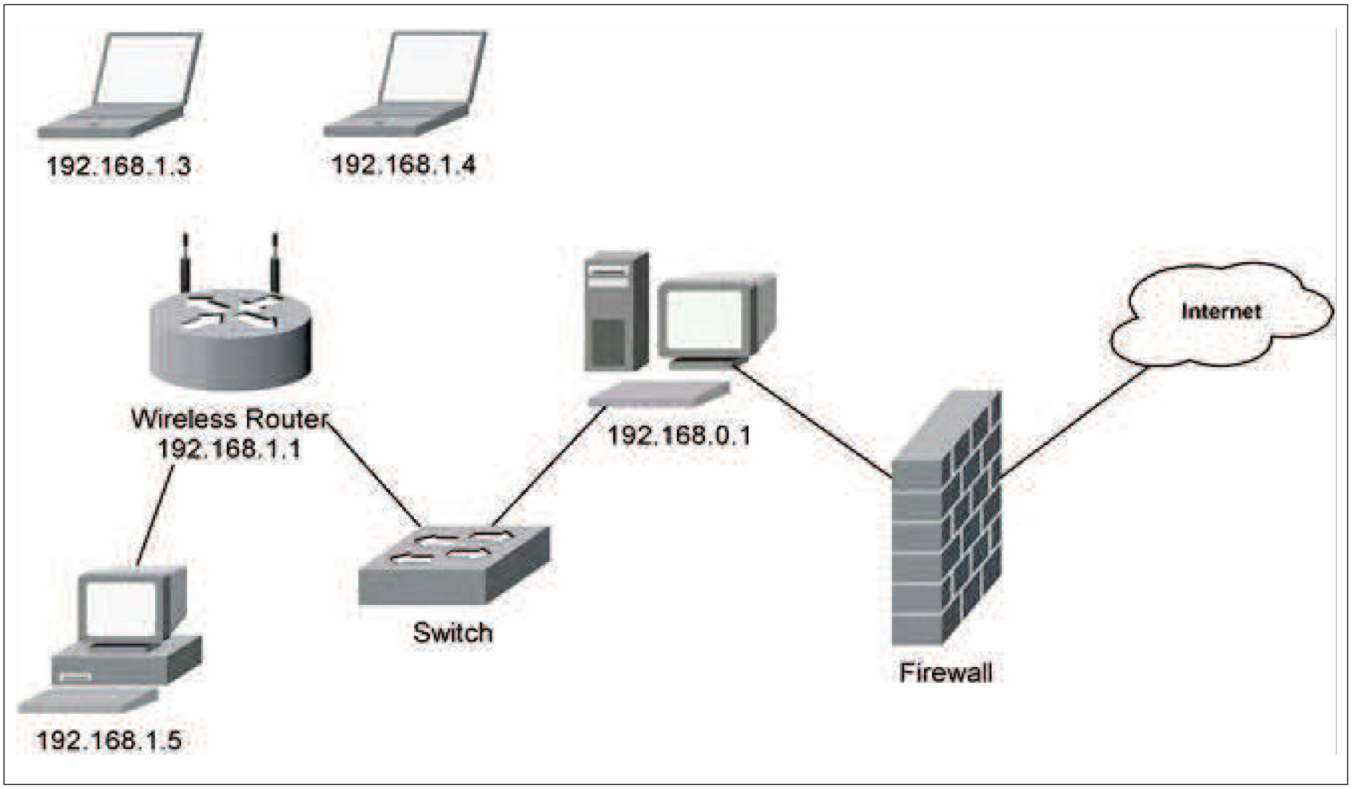

Figure 3. Network configuration

The minimum, maximum and average values of some of the most important parameters such as jitter, packet intervals and used bandwidth are given in Table 1.

Metric	**Average**	**Low**	**High**
Src Jitter (ms)	0,712	0,007	3,273
Dest Jitter (ms)	0,911	0,005	2,735
Src Packet Interval (ms)	2,431	0,009	45,001
Dest Packet Interval (ms)	2,561	0,110	33,792
Src Bandwidth (kb/s)	127,691	127,421	133,333
Dest Bandiwdth (kb/s)	127,654	127,405	137,143

Table 1. Audio Detils

The addresses of the source and the recipient along with their ports and the type of used media are given in Table 2.

Parameter	**NodeA**	**Node B**
Address	192.168.1.3	192.168.1.4
Port	40000	40000
Media Type	G.711 Alaw	G.711 Alaw
SSRC	00294823	4AE13D6C
Audio/Packet (ms)	3	3
Frames/Packet	3	3
Total Packets	79 834	79 814
Packets Lost	0	0
Longest Packet Loss Burst	20	13
Total Payload Bytes	1 916 016	1 915 536

Table 2. Type of used Media

The results in Table 3 refer to indicators of monitored calls [3].

RFactor / MOS Score are characteristics of the speech tract for each individual media stream and are assessed subjective. The software used in the research is processing by models including time-varying characteristics such as: the sequence of lost packets, discarded packets due to high jitter and others.

Burst Packet Loss Rate gives the number of the lost packets in percentages in case of unusual circumstances.

Average Burst Length shows the average duration in milliseconds of the unusual circumstances that would lead to packet loss.

Average Gap Length is the average length of the intervals in milliseconds in the event of unusual circumstances in the respective media streams.

Good Packets is the number of received packets at the two endpoints.

Lost Packets is the number of lost packets in the network for both endpoints.

Discarded Packets are the rejected packages when the delay is exceeded or earlier arrival at the source or recipient is happened.

Media Type shows the type of used media format.

A graphic expression of the quality of the audio signal is shown in Fig.4. QoS reports are given, as a result of analysis of VoIP calls over 30 seconds.

The parameters of the audio stream in graphical form are presented in Fig.5. These are number of packets per second, jitter in the

Metric	Source	Destination
Listening R Factor	93	93
Conversational R Factor	93	93
Listening MOS Score	4,195	4,195
Conversational MOS Score	4,195	4,195
P.862 Raw MOS Score	4,449	4,449
Burst Packet Lost Rate (%)	0,000%	0,000%
Average Burst Length (ms)	0	0
Average Gap Length (ms)	28411	48546
Good Packets	40005	40010
Lost Packets	0	1
Discarded Packets	2	0
Media Type	G.711 Alaw	G.711 Alaw

Table 3. Audio QoS

audio channel, packet interval of audio packages and the speed used in kbps.

The ranges of delays of the individual packets during transmission of G.711 audio stream have network delay below 10ms, and the greatest delays do not exceed 60-70ms with limit values 150ms in one direction (Figure 6).

The majority of packages have jitter below 3ms, while the largest values do not exceed 8-10ms, which again is in the norm of 20ms (Figure 7).

G.722.1: The coding system uses subband adaptive differential pulse-code modulation (SB-ADPCM) at a speed of 64kbps. The system has three modes of operation for 7 kHz audio encoding, corresponding to the bit rate which is 64 or 56, or 48kbps. For the study a speed mode 64kbps is used.

G.723.1 6,3kbps: This recommendation of the ITU, has two standard speeds - 5, 3 and 6,3kbps. The higher speed, which is used in the study, gives better quality. —J;68=0B0 The length of the voice frame is 30ms and further prediction of 7,5ms allows algorithmic delay of 37,5ms. Encoder is designed to compress a voice with good quality with reduced complexity and it is not suitable for music [4].

To assess the MOS G.711 a study was conducted with SIP signaling on a heavily loaded network (Fig. 9). Despite the large number of rejected packets and large network delay, MOS assessments for forward and reverse direction of the conversation are fully comparable with the requirements.

Data on network delays of packets, jitter, packet intervals were obtained for real SIP call. The calls have good quality – understandable, without echo and without noise.

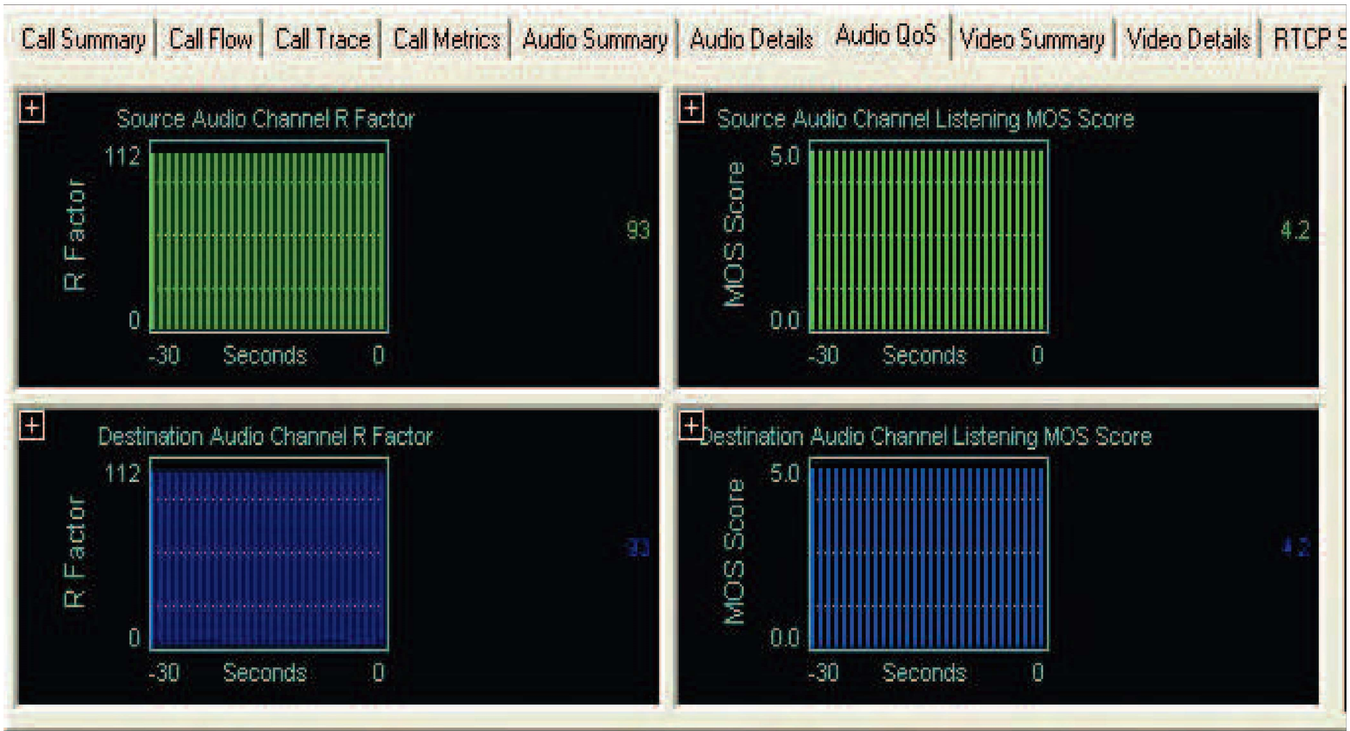

Figure 4. Audio QoS

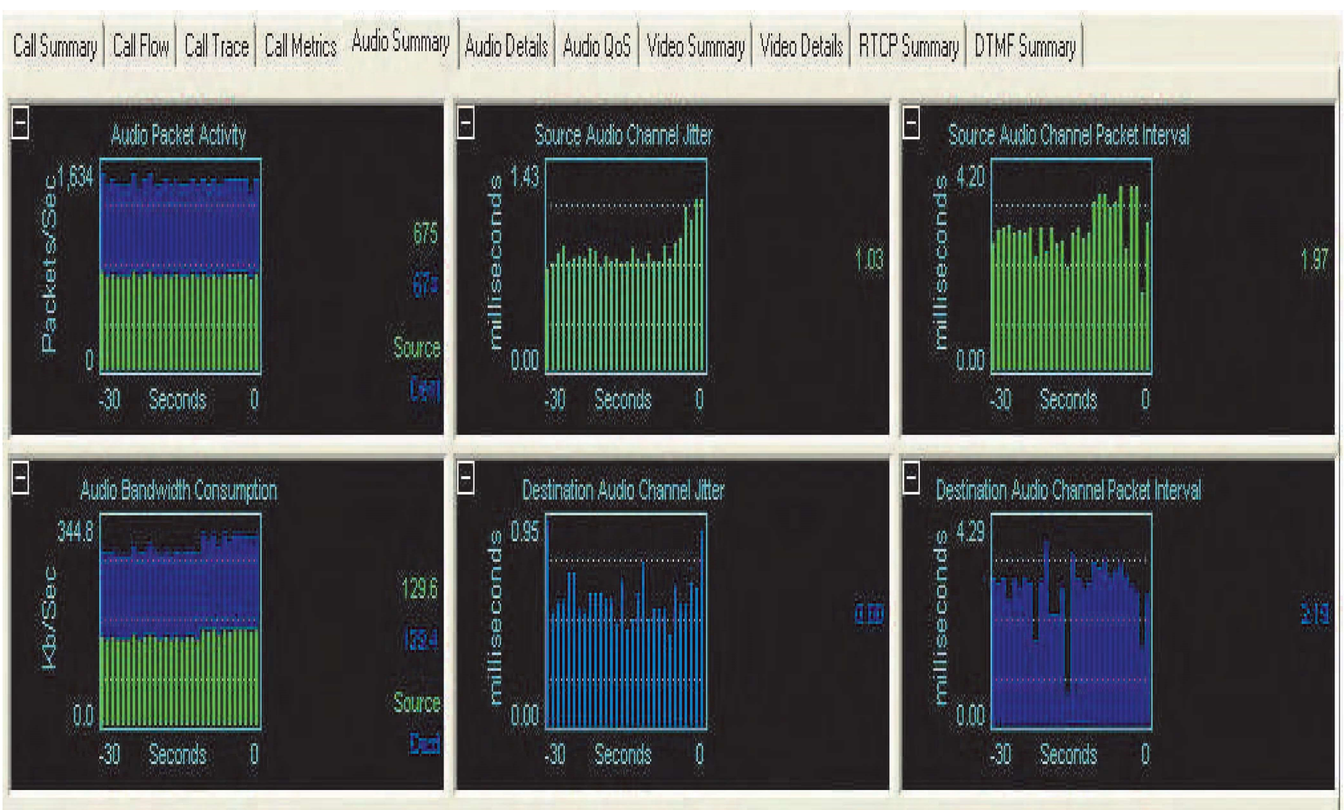

Figure 5. Graphic form of the audio stream parameters

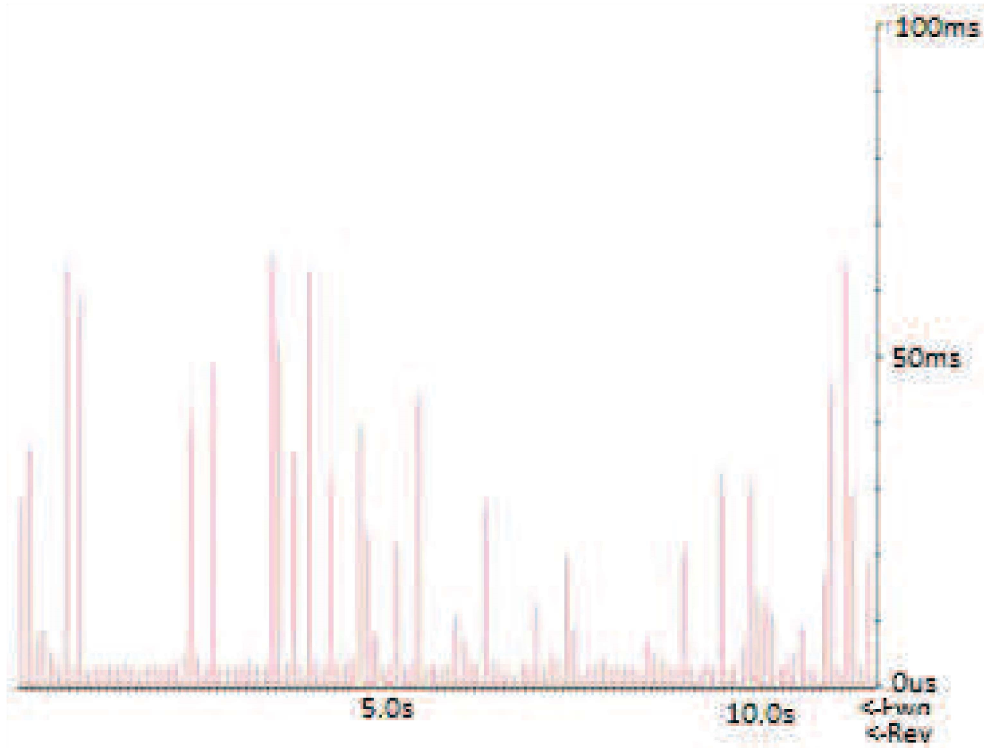

Figure 6. Delays packages using G.711

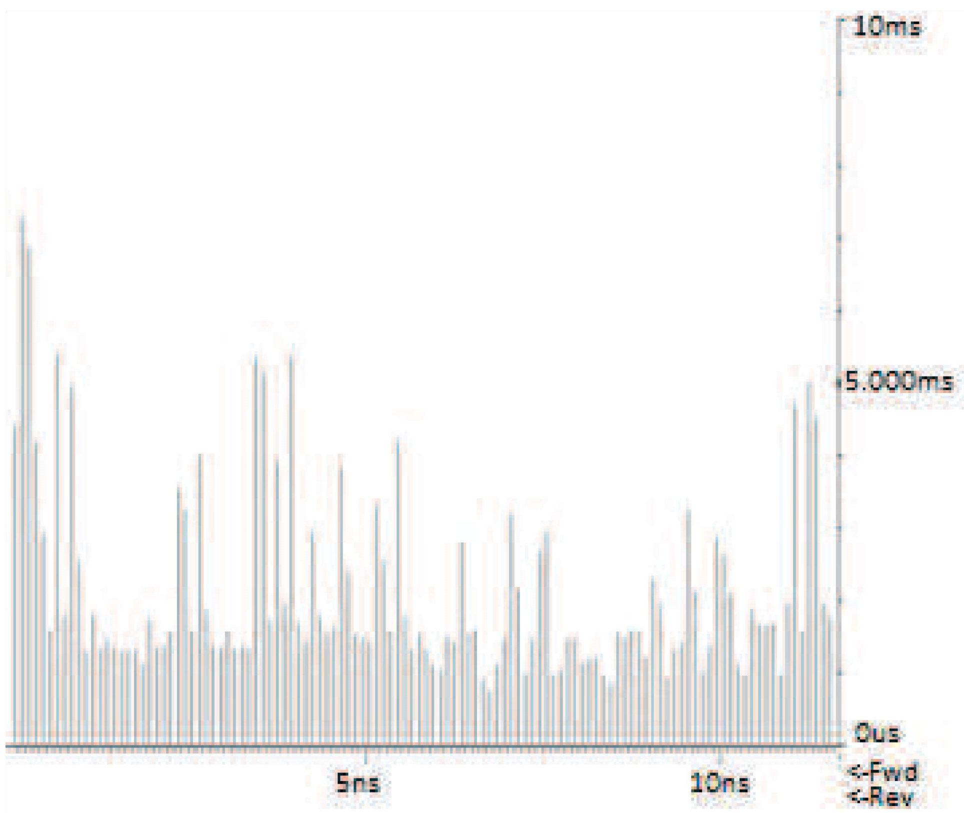

Figure 7. Jitter of the individual packages

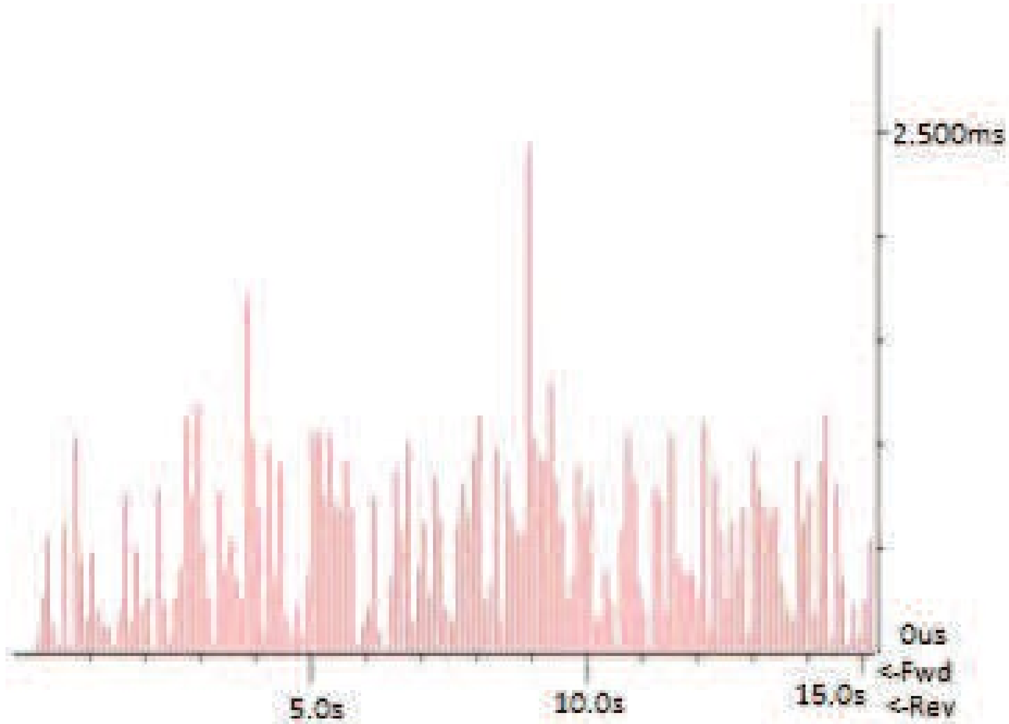

Figure 8. Delays of packets in G.723.1 6,3kbps

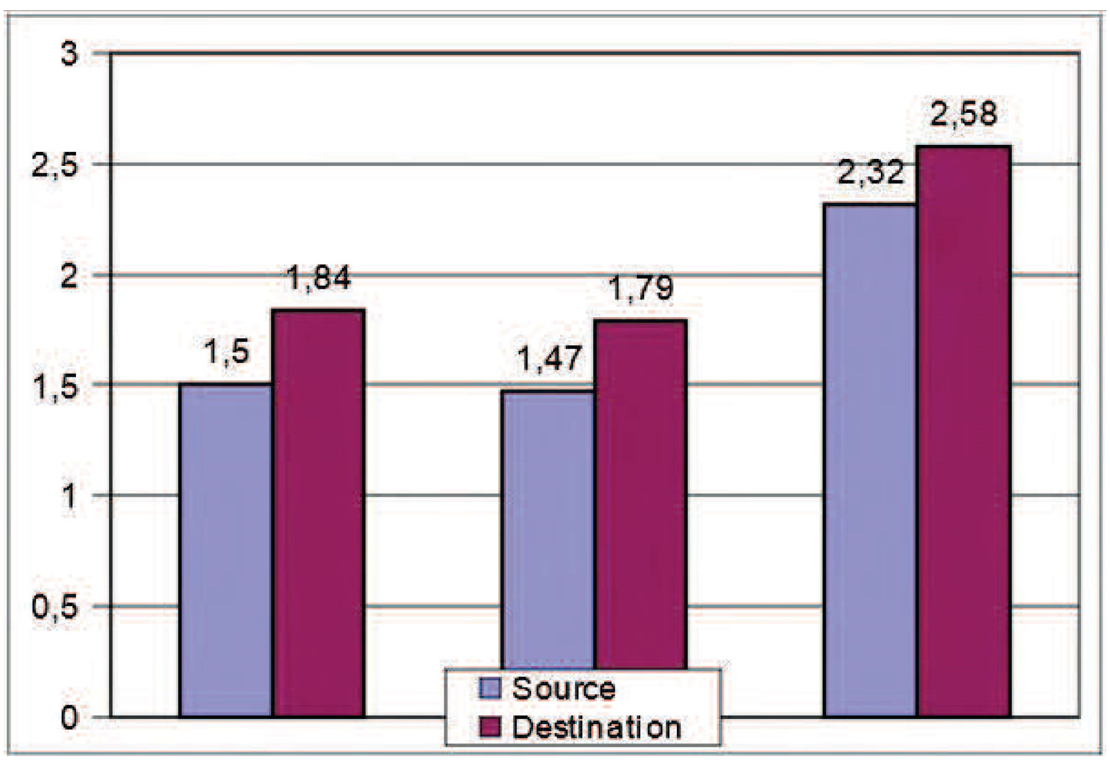

Figure 9. MOS results in loaded wireless network

3. Conclusion

The testing of the realised Vo802.11 network shows its ability to compete PSTN under comparable conditions: the use of uncompressed voice and the appropriate for the conditions protocols. The possibility of using compressed formats of the media stream while maintaining the best operational parameters gives sizable potential of the system and makes it flexible for network designers and advanced signaling provide high intelligence of the entire network. These factors make it possible to quickly build stable and reliable systems providing quality services to the end users. In the process it became clear that the disadvantages associated with the range of wireless networks and their security cannot be a real obstacle to the implementation of Vo802.11 voice transmission in real time.

The main contradictions that still remain to the fore are those related to QoS. To show in practice, in real conditions, the behavior of a system Vo802.11, a wireless tract was examined for two major signaling protocol SIP and H.323, and for the only possible at the moment transport protocol - RTP. The obtained results and the comparisons are an evidence of work together of SIP and H.323 - SIP realizing a signaling over packet switched networks, and H.323 building interface via gateways to the PSTN.

Taking into account the obtained data, concerning different aspects of the behavior of the investigated system, and recognizing that they are not used special measures to provide better QoS, such as RSVP, DiffServ, then the results obtained can guarantee the operability of Vo802.11 for voice communications.

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