Performance Evaluation of the IEEE802.11e WLAN for VOIP Communication
using OPNET Modeler

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Abstract

The existing 802.11 protocols primarily use the distributed coordination function (DCF) access method to the wireless medium. The DCF provides an equal chance to each device to access the wireless medium. When dealing with audio, video, gaming and other applications that are intolerant to bandwidth fluctuations, the fairness access provided by DCF is inadequate. The IEEE 802.11e standard is targeted at addressing these issues. In this paper the performance of the IEEE802.11e protocol for VOIP communication is analyzed using OPNET Modeler. Effect of Background traffic on Voice quality; Effect of end-to-end delay, Jitter and Packet loss on Voice quality; Effect of distance of the mobile workstations from the AP on Voice quality are analyzed in relation with the MOS values which are set to determine the quality of voice communication over WLAN.

Key Words:- Protocol, quality, voice, communication, traffic

1. Introduction

1.1 The IEEE802.11e Protocol

The existing 802.11 protocols primarily use the distributed coordination function (DCF) access method to the wireless medium. The DCF provides an equal chance to each device to access the wireless medium. When dealing with audio, video, gaming and other applications that are intolerant to bandwidth fluctuations, the fairness access provided by DCF is inadequate. The IEEE 802.11e standard is targeted at addressing these issues and contains two main sections. The first is enhanced distributed channel access (EDCA), which defines four priority levels or four access categories (ACs) for different types of packets. It doesn't, however, guarantee bandwidth, jitter or latency. The second is hybrid coordination function controlled channel access (HCCA), which guarantees reserved bandwidth for packets classified based on EDCA by using a central arbiter for the bandwidth usage.

While in the DCF all stations try to access the wireless medium with the same priority, in EDCA there are four levels of priority or ACs. The mechanism of listening to the medium and using a back-off mechanism to determine the allowed transmission time is similar to that defined by DCF. However, unlike DCF, the maximum back-off times are different for the different ACs, meaning that higher-priority ACs have a shorter maximum back-off time than lower-priority ACs. The shorter maximum back-off time allows the higher-priority AC to win access to the wireless medium more frequently than the lower-priority AC. Once a device has gained access to the wireless medium, it has the opportunity to continue transmitting for a specified transmission opportunity (TXOP). Applications or packets that share the same AC also have the same maximum back-off time and, hence, the same chance to gain access to the wireless medium. EDCA is fairly simple to implement, but cannot guarantee latency, jitter or bandwidth.

1.2 Voice over Internet Protocol (VOIP)

VoIP uses both the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) protocols for transport. The control protocols use TCP, while the data protocols and voice traffic use UDP. TCP is used for the control protocols because it is a lossless protocol; lost packets are recovered and delivery is guaranteed. Voice traffic uses UDP because guaranteed delivery is not required. The voice data is sent using Real Time Protocol (RTP) over UDP. Each RTP packet contains a short sample of the voice conversation, ranging from 10 ms up to 50 ms. The size of the RTP packet and the length of the voice sample depends on the CODEC used for the voice conversation. To send voice data over the network, it must be passed through a CODEC. Many CODECs are available for VoIP use but the most common type of CODEC is G.711, this CODEC produces a 64 Kbps stream. We will use this CODEC throughout this simulation experiment.

1.3 The SIP Protocol

The Session Initiation Protocol (SIP) is an ASCII-based, peer-to-peer application layer protocol that defines initiation, modification and termination of interactive, multimedia communication sessions between users. SIP is defined as a client-server protocol, in which requests are issued by the calling client and responded to by the called server, which may in itself be a client for other aspects of the same call. SIP is not dependent on TCP for reliability but rather handles its own acknowledgment and handshaking. This makes it possible to create an optimal solution that is highly adjusted to the properties of VoIP.

2. The Simulation Experiment

Appropriate scenarios for testing a voice quality over an IEEE802.11e protocol were depicted. By generating a VOIP traffic service at each workstation, simulations were made to evaluate the performance of voice calls in each scenario.

2.1 Tools used in the experiment

2.1.1 OPNET Modeler:

The in-built MOS measuring tool to evaluate the VOIP performance were used.

2.1.2 MOS (Mean Opinion Score) & R values

The MOS is a representation of the quality of human speech. To determine MOS for a specific configuration, MOS rates a voice quality from 1 to 5, 1 being the worst, and 5 being the best quality. The MOS of a specific configuration is the arithmetic mean of the individual MOS values as recorded by the listeners. MOS is a very useful means of measuring voice quality, as it allows for easy comparison of voice call quality from one test to the next. R-factor is another tool that can be used to measure voice quality over a given network. When using MOS values, call quality is measured on a scale of one to five, with one having the lowest call quality and five the highest. R-factors use a scale of zero to 100, where zero represents the lowest quality and 100 the highest. (Fig. 1)

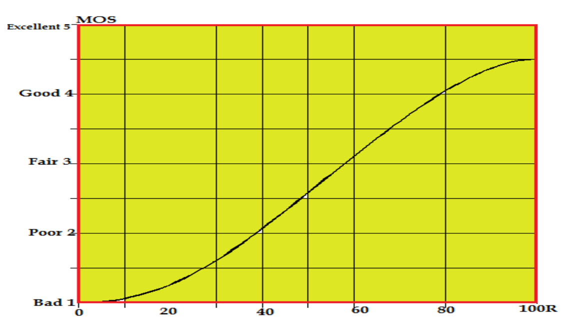


Fig.1 Relationship between MOS and R-factor

2.2 Configuring Call Quality Requirements

The report that could be generated by the OPNET VOIP assessment provides information about the call quality in the network, but also evaluates whether the call quality meets the performance requirements you specified. For evaluating call quality in the network, OPNET has a facility to configure the following: The thresholds for rating a call to be of poor, acceptable, or good quality. The allowable amount of calls with poor call quality in a network with acceptable voice performance

The VoIP Readiness Assessment lets you configure call quality requirements in terms of MOS values or R-factors. For both measures, you specify the requirements in the same way: by specifying the boundaries of the range of values for poor, acceptable, and good call quality. During the assessment, the software computes the quality of a call and determines whether that call quality is poor, acceptable, or good based on the values you configured in the Service Level Criteria screen. Because of this, the report depends greatly on the values you configure in the assessment wizard

2.3 Experimental Test-bed

The scenario shown bellow (fig. 5) were depicted to make the simulation

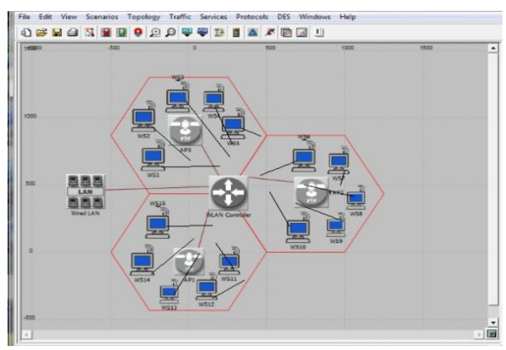


Fig.5 Scenario depicted to make the simulation

Scenario Description

- Three Access points connected to a central WLAN controller; Five mobile workstations connected at each access point with a roaming capability enabled and different parameters of the IEEE802.11e were configured that are appropriate to this simulation experiment.
- Wired LAN with six computers connected to WLAN through the WLAN Controller. VOIP traffic service with background data traffic is implemented at each mobile workstation and the different attributes are set (see the procedure in this document for setting up the VOIP traffic service)

Simulation experiment performed:

- To evaluate the real performance of WLAN for VOIP service as a result of Delay, jitter and packet loss, a constant Bit Error Rate (BER) were introduced at each mobile workstation and the effects on voice quality were measured in terms of MOS.
- To see the effect of background traffic on voice quality, by enabling and disabling the EDCF attribute of the IEEE802.11e protocol, the voice quality were simulated by incorporating background traffic over the network and voice quality were measured in terms of MOS.
- Effect on call quality as more simultaneous voice calls made over the WLAN were simulated and results were obtained in relation with MOS. This will help us later to determine the maximum number of supported calls that can be made over the IEEE802.11e protocol.
- To see the effect of distance of the workstation from the access point, Mobile workstations were configured to move away from the access point and voice quality results were obtained in relation with the MOS value.

2.4 Simulation Results

2.4.1 Effects of Background traffic

2.4.1.1 Scenario Used

The scenario above (fig.5) were used

2.4.1.2 Assumptions

- No hidden node problem
- No interference signal
- All workstations are assumed within the working range of the access point (5m from the AP) for better signal strength.
- Roaming capability of the workstation is disabled
- The same type of voice traffic is configured at each workstations
- The default Constant bit rate (CBR) background traffic is used

2.4.1.3 Simulation Procedure

1,3,4,6,7,8,9 and 10 number of voice calls were generated separately with and without background traffic then average MOS values were generated for each of the voice calls made by enabling and disabling the QoS parameters of the ieee802.11e protocol.

2.4.1.4 Results and discussion

From the output of the simulation result, a graph is plotted (fig. 7) to compare and contrast the effect of background traffic on the performance of a voice quality with enabled and disabled QoS parameters of the ieee802.11e protocol.

A total of 24 reports like shown on the sample report figure (Fig.6) were generated by OPNET. The results obtained are tabulated as shown in Table.1 so as to plot the graph on fig.7.

When only one, two, three or four workstations are communicating simultaneously, call quality When only one, two, three or four workstations are communicating simultaneously, call quality is good in all tests. When number of workstations connecting to the network is increased, the average MOS with background traffic drops below 3. This is because the background traffic has a greater effect on the voice call quality as the numbers of workstations are increased.

The quality of calls with background traffic decreases further with increased number of workstations while the calls with no background traffic remain roughly consistent in quality until we reach the maximum number of call supported by the ieee802.11e protocol. At 7 number of calls at a time, even with no background traffic, call quality decreases (MOS = 2.5). This is because of increased packet loss due to retransmissions as well as increased delay and jitter levels.

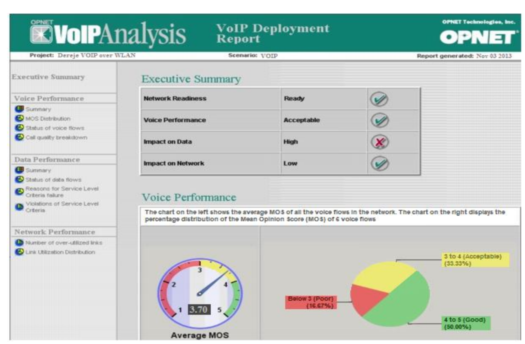


Fig. 6 Sample of the VoIP deployment report (Average MOS value for 6 simultaneous voice calls made with background traffic and QoS parameters enabled)

Table1. Average MOS value for simultaneous voice calls made with background traffic

No. of Simultaneous Voice Calls made at a time	1	3	4	6	7	8	9	10
Average MOS (With background traffic and QoS enabled)	4.2	4.2	4	3.7	3.2	3	2.8	2.4
Average MOS (With background traffic and QoS disabled)	3.8	3.8	3.6	3	2.5	2	1.5	0.3
Average MOS (without background traffic and QoS enabled)	4.5	4.5	4.5	4	3.6	3.6	3.4	3.2
Average MOS (without background traffic and QoS disabled)	4.5	4.5	4.2	3.8	3.6	3.2	3	2.6

From the graph, it is clearly seen that: With EDCF and HCF attributes of 802.11e disabled i.e with no QoS, that is, with all traffic having the same priority, call quality with background traffic drops off significantly but without background traffic and with enabled QoS of 802.11, the voice quality is dramatically improved. This is because EDCF and HCF attributes of 802.11e protocol is giving higher priority for Voice communication than other types of data traffic over the network.

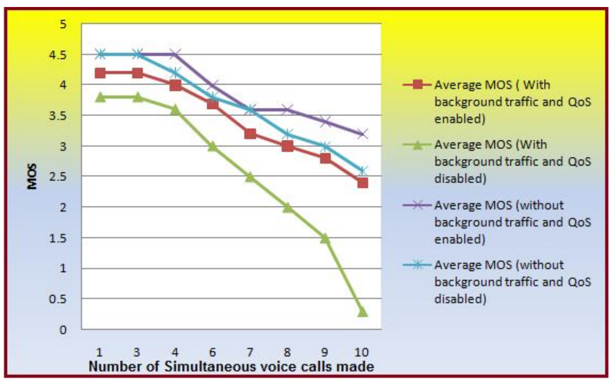


Fig.7 No. of calls versus MOS with and without background traffic and With QoS enhancement and without Qos

As it is clearly seen from the graph (Fig.7), it is possible to make 10 simultaneous voice calls with 3 MOS value under background traffic with QoS enabled (which is acceptable in our case) but if you see the MOS value to make 10 simultaneous voice calls with no QoS, it has the value of 0.3 (which is poor in our case).

2.4.2. Effect of end-to-end delay, Jitter & P. loss

2.4.2.1 Scenario Used

The scenario above (fig.4) were used

2.4.2.2 Assumptions

- No hidden node problem, No interference signal
- All workstations are assumed within the working range of the access point (5m from the AP) for better signal strength.
- Roaming capability of the workstation is disabled
- The same type of voice traffic is configured at each workstations
- The default Constant bit rate (CBR) background traffic is used
- QoS parameters are enabled

2.4.2.3 Simulation Procedure

1,3,4,6,7,8,9 and 10 number of voice calls were generated separately with background traffic and different values of bit errors which are added in the network (OPNET has the facility to add constant Bit Error Rate (CBER)) then average MOS values were generated for each of the voice calls made respective to the different values of the CBER.

2.4.2.4 Results and discussion

Most VoIP quality performance is closely related and affected by end-to-end delay, jitter, and packet loss. A two-way conversation is very sensitive to delay and jitter, but it can tolerate some degree of packet losses, depending on the error-resilience of the codec used. ITU has recommended that one-way end-to-end delay should be no greater than 150 ms for good voice quality, and up to 400 ms for acceptable voice quality, with an echo canceller. Packet loss is also a major source of impairment in VoIP systems. A voice quality is considered acceptable only when the packet loss rate is less than 2%.

To evaluate the effect of delay, packet loss and jitter on voice performance over WLAN, we introduced a constant Bit Error Rate (BER) values over the network depicted on the scenario (Fig.4). We assume that the channels between all pairs of nodes are subject to this BER value. Theoretically, poorer channel conditions will lead to higher BER values, which would cause an increase in per-packet delay that would definitely degrade the numbers of voice calls that can be made at a time and the quality of the voice as well.

From the simulation, we observe that for BER less than or equal to 10^{-5}, the packet error rate and the delay is so low that the difference in capacity between such a channel and an error-free channel is almost negligible, therefore the network can support a number of simultaneous voice calls (7 in our case) with MOS value of 3 and above (Fig. 8 and Table.2). For BER greater than or equal to 0.01, the voice call that can be supported by the network reduced drastically due to the very high packet error rate (For our case less than 2.1 MOS value to make 3 and less than three calls at a time) (Fig. 9 and Table 3)

From this simulation we can conclude that a higher bit error rate (BER) resulted from end to end delay, jitter and packet loss, would degrade the voice quality and affects the number of quality voice calls that can be made over the WLAN network. When the BER value varies from 0 to 10^{-5}, the delay incurred on the WLAN is only from 7ms to 33ms and 8 simultaneous calls can be made with a MOS of 3 which is in the range of good quality voice in our assumption.

Table.2 Average delay, MOS and number of voice calls for BER = 10^{-7}

BER = 10^{-7}								
Number of Simultaneous voice calls made	1	3	4	6	7	8	9	10
Delay (ms)	7	8	11	17	20	23	27	33
Average MOS	4.2	4.2	4	3.7	3.2	3	2.8	2.4

<u>Note:-</u> The Delay and average MOS values for 0, 10^{-6} and 10^{-5} BER values are almost Identical with the above values, therefore we omitted these values for simplicity purpose.

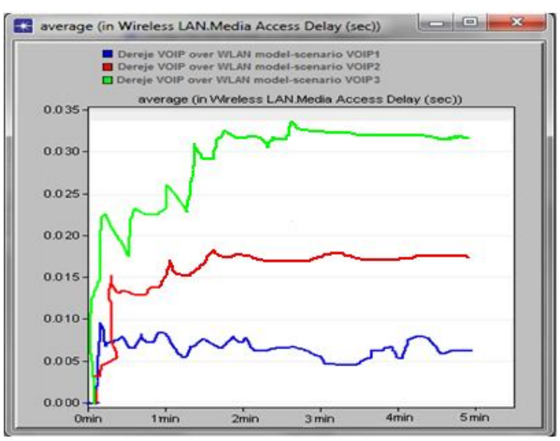
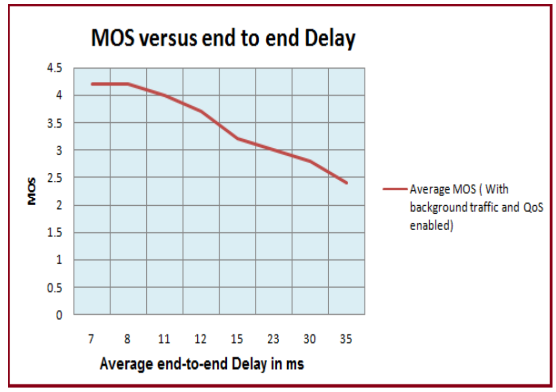


Fig 8 (a)



(b)

Fig. 8 a & b Sample average delay and MOS values for the BER value of 10^{-7} making 1, 6 and 10 simultaneous voice calls at different simulation time

Table3. Average delay, MOS and number of voice for different values of BER

BER	0.0001	0.001	0.01	0.1
Average Delay (ms)	82ms	300ms	400ms	550ms
MOS	3.6	2.8	2.1	1.2
No. of voice calls made at a time	3	3	3	3

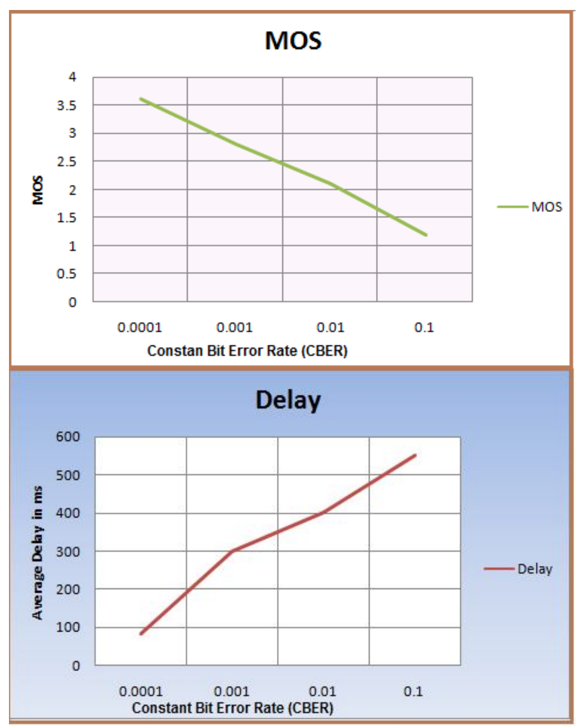


Fig. 9 Sample average delay and MOS values for different BER values greater than 10^{-5}

2.4.3 Effects of distance from the AP

2.4.3.1 Scenario Used

The scenario above (fig.5) were used

2.4.3.2 Assumptions

- No hidden node problem
- No interference signal
- All workstations are subjected to move within the working range of the AP (in our case from 0 to 45m)
- Roaming capability of the workstation is enabled
- The same type of voice traffic is configured at each workstations
- The default Constant bit rate (CBR) background traffic is used
- QoS parameters of ieee802.11e protocol are enabled

2.4.3.3 Simulation Procedure

All workstations were subjected to 5,15,25,35 and 45 meters from the AP at different simulation time and MOS measurements were taken by making 1,3,4,6,7,8,9 and 10 number of voice calls respective to each distance values.

2.4.3.4 Results and discussion

As it can be clearly seen from the graph (Fig.6), when the distance from the AP to the workstation increases, the link adaptation mechanism degrades the physical mode. As a consequence, the available throughput above the MAC layer is reduced and less simultaneous voice calls are possible. In this case there is a high degradation of the capacity of the network to support more number of voice calls with increased distance of the workstations from the AP. According to the simulation result, only one voice call with MOS value of 4 is possible by the ieee802.11e protocol if all workstations are 45m away from the AP. Contrary to this, the simulation result shows as it is possible to make more than 5 simultaneous calls with a MOS value greater than 3 if the workstations are within 25m distance from the access point..

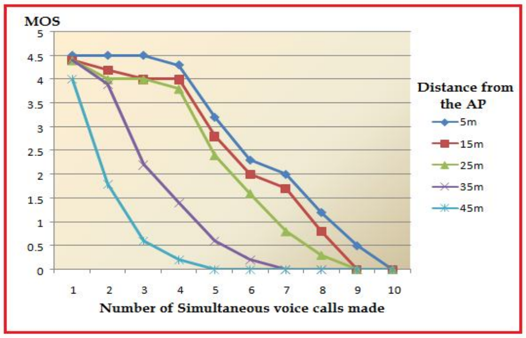


Fig. 10 MOS values vs. number of simultaneous voice calls and the effect of the distance to the AP

3. Summary and Conclusion

Evaluating the effects of different WLAN parameters on voice communication quality were the main objective of this paper. The effects of network parameters like: - effects of background traffic on voice quality; effects of end to end delay, jitter and packet loss on voice quality; effects of client roaming on voice quality: effects of distance of the mobile workstations from the AP on voice quality; were given emphasis and studied in detail. We proved as these WLAN parameters has a tremendous impact on the performance of the WLAN protocol, therefore it is a wise idea again to look for an appropriate strategies to resolve the problems caused by this network parameters so that a quality voice communication could be achieved over the IEEE802.11e wifi protocol.

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