ANALYSIS OF VOIP TRAFFIC IN WIRED AND WIRELESS LANS

Belqasem Salem Almontser

Abstract:

This paper is an experimental study to analyze the performance of Voice Over IP (VoIP) traffic over wired and wireless Local Area Networks (LANs) by observing VoIP Quality Of Service (QoS) parameters, these parameters comprise jitter, delay and packet loss ratio. Two widely deployed LANs technologies (IEEE 802.3Ethernet in wired and IEEE 802.11G in wireless) are chosen as the test environment to conduct this study, the test methodology adopted in this study is based on simulation of real voice traffic in both network environments, thus the study was conducted in the laboratory using real equipment. In order to evaluate VoIP traffic performance, Quality of Service (QoS) parameters on the Real-Time Transport Protocol (RTP) packet transmission under different background traffic loads are measured, File Transport Protocol (FTP) traffic is used as the back ground traffic. A set of graphs representing the jitter and delay graphs versus the RTP packets number are produced.

The test results show that at high network loads, VoIP traffic in both network environments suffers high delay and jitter values. Also it shows that performance of VoIP traffic in wireless is much worse than the performance in Ethernet under the same load conditions. In both Ethernet and wireless, it was noticeable that there is a strong linear correlation between RTP packets delay and packet loss ratio.

Introduction:

In today's world Wireless LANs becomes an essential part of many enterprises' networks, and quickly they have proven their worth. Simultaneously, many enterprises are implementing VoIP systems to have the benefit of the lower call cost and good quality service offered by this technology, the two technologies Wireless and VoIP together have founded an application called wireless VoIP.

Besides the less expensive infrastructure by the convergence of both data and voice in the same network, Wireless VoIP provides another advantage, the mobility which significantly increases the importance of both technologies; this mobility becomes more important after the arrival of new Wi-Fi phones include both cellular and Wi-Fi (dual mode) with the advantage of Wi-Fi as the inside building wireless network [3], this encourages the enterprises which are currently deploying VoIP to consider integrating WLANs into their VoIP systems, wireless VoIP allows the users to use VoIP system even in the places where the network cables are not available.

Several issues raised by the deployment of VOIP over wireless include the admission control, quality of service, system architecture and the network capacity [2]. In this paper, the quality of service issue will be considered, the study includes measuring and analyzing the performance of VOIP traffic in both wired and wireless LANs under different background traffic loads this accomplished by measuring QoS parameters such as Jitter, delay, Packet loss at different Network Loads. In this study analysis and comparison of the degradation of quality of service with respect to the increase of network load in both network environments are provided.

Methodology:

In order to ensure that the test techniques which are used in this study do provide the precise VoIP performance measurements and in order to facilitate VoIP deployment in both wired and wireless LANs, a research methodology must introduce. Many previous studies, have used network simulation software for VoIP performance evaluation, Zubairi &Zuber [10] have conducted a study to evaluate the performance of VoIP traffic on a university campus network under varying load conditions, their work is similar to this study, the difference is in the simulation environment, in [10] a network simulator software is used, also the study was conducted in wired side only.

However, on this study the test includes both wired and wireless and an empirical methodology is adopted, this methodology based on a real traffic simulation using a test environment in the laboratory with real equipment. Two widely deployed LANs technologies, IEEE 802.3Ethernet in wired and IEEE 802.11g in wireless are chosen as test environments in this study.

In order to accurately notice the quality of service degradation the delay, jitter and packet lost values results, which are obtained from the experiments are organized in the form of graphs, representing the delay or jitter values versus number of RTP packets. The average jitter and delay graphs and packet loss rate versus the network loads graphs for both wired and wireless networks are derived from different network loads results. Comparative plots for delay and jitter in both wireless and wired are produced in order to compare the performance.

1.1 Experimental Ethernet LAN Test bed:

The objectives of this study are to analyses the performance of VOIP traffic in both wired and wireless LANs, first started with the wired side, as shown in (Figure 1) the test network topology is built in structure of campus network with a hierarchy design. At the access layer two Catalyst 2950 Cisco switches are connected. A station running Cisco IP Communicator (CIPC) is connected to ALS-2 switch.

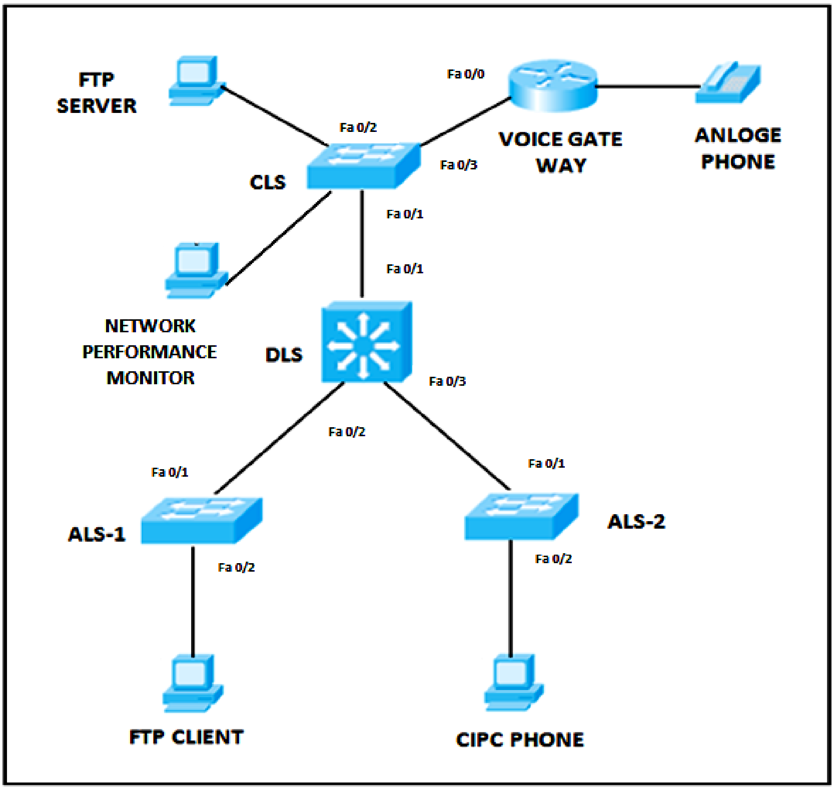


Figure: 1 Ethernet LAN Test bed

A station running File Zilla client software [4] is connected at ALS-1, at the distribution layer a Catalyst 3550 multilayer switch is used. Cisco 2811 series router represents a voice gate way configured with Call Manager Express connected to the core layer switch CLS. In order to simulate the connection to the public switched telephone network (PSTN), an analogue phone is connected to the voice gate way on the FXS port 0/2/0, which is a voice enabled port. A station running FileZilla server software [4] is connected to core layer, this represents the network file server.

The network performance monitoring is accomplished by connecting a station running wire shark utility to the interface fa0/4 on CLS switch.

1.2 Experimental WLAN Test bed:

To test the wireless side, as shown in (Figure 2), the network extend to include a wireless LAN, a Cisco 1200 access point with a default configuration is connected to the DLS switch.

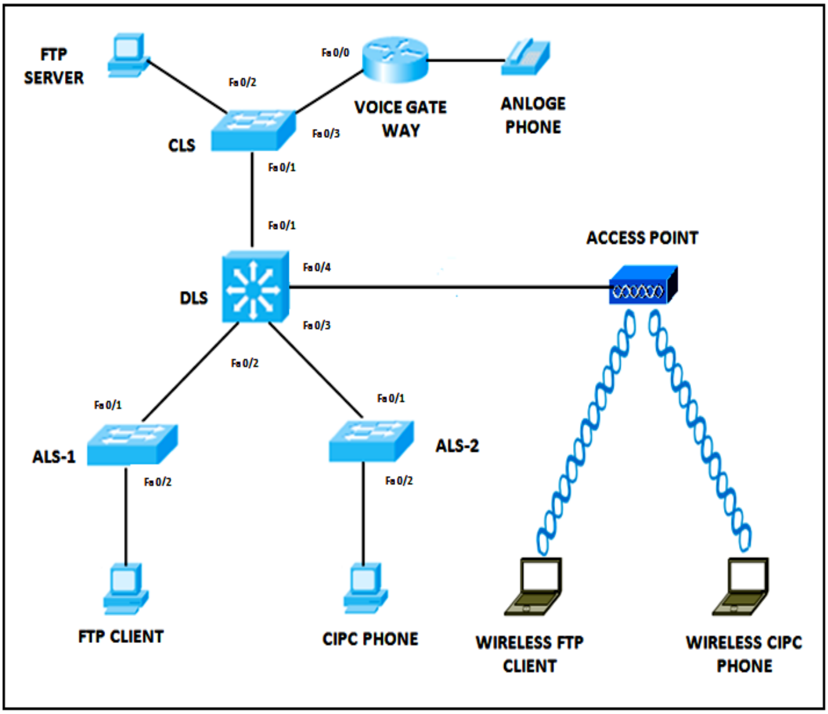


Figure: 2WLAN Test bed

The access point works in 802.11 g mode. To maintain the measurement consistency in both network environments, the same numbers of nodes are connected in both sides. Wireless station running Cisco IP Communicator (CIPC) phone is connected to the WLAN; this simulates a VOIP wireless phone. On the other hand another wireless station is loaded with FileZilla client software [4] is used torepresent FTP client, which requests a file from the FTP server and shares the wireless bandwidth with the VoIP traffic.
2.1 VOIP performance measurements in the Ethernet LAN.

In order to define the test loads, it is necessary to measure the maximum throughput that can be achieved in the network, for this purpose the network is over loaded by downloading a file from the server without any speed restriction. To precisely monitor the network load variation, bandwidth monitor software [1] installed in the FTP server and FTP client's stations. the maximum throughput was reached in the Ethernet LAN is 89 Mbps. The maximum throughput obtained in this test is divided in to a percentage scale from 10% to 100% as shown in (Table 1) with a major unit of 10%.

Load (%)	10	20	30	40	50	60	70	80	90	100
Load(Mbps)	8.9	17.8	26.7	35.6	44.5	53.4	62.3	71.2	80.1	89

Table: 1 Ethernet LAN test Loads

On the Ethernet LAN performance test, from the CIPC phone a call is generated to contact the analogue phone, while the call is active, the network load is gradually increased by limiting the down load speed on the FTP server, and at each network load level a sample of RTP packets is captured.

2.2 VOIP performance measurements in WLAN

The WLAN type is infrastructure-based network supports 802.11g configurations with 54 Mbps data rate operates in 2.4 GHz frequency band. While the theoretical throughput for 802.11g is 54 mbps. However, the maximum throughput achieved in the lab is only 21 Mbps. The test procedure used to perform the WLAN test is the same procedure has been used in Ethernet LAN test, the only difference is the way that the RTP packets captured, on the WLAN test the RTP packets traffic captured on the CIPC client wireless interface using wireshark on non-promiscuous capture mode.

Load (%)	10	20	30	40	50	60	70	80	90	100
Load (Mbps)	2.1	4.2	6.6	8.4	10.5	12.6	14.7	16.8	18.9	21

Table: 2 WLAN test Loads

The maximum throughput reached in the WLAN is 21 Mbps. This maximum throughput divide in to a percentage scale from 10% to 100% as shown in (Table 2) with a major unit of 10%.

3.1Results and VOIP performance in Ethernet LAN.

The results of this test are shown in (Table 3). For every network level load, a jitter and delay average values are cacluated, to observe how the values spreads around the average standard deviation values are alsocalculated. Jitter and delay graphs and average values versus the network loads graphs are produced.

Network Load	Voice Jitter (ms)			Packet End-to-End Delay(ms)			Packet Loss Ratio
	MAX	AVERAGE	STDEV	MAX	AVERAGE	STDEV	
0%	2.83	2.638	0.206	24.69	19.984	2.88	0.0%
10%	3.01	2.662	0.225	24.94	19.988	2.928	0.0%
20%	2.89	2.639	0.208	24.731	19.984	2.895	0.0%
30%	2.86	2.515	0.210	25.08	19.989	2.789	0.0%
40%	3.01	2.726	0.225	120.48	20.238	5.135	1.23%
50%	3.02	2.653	0.233	137.53	20.093	4.360	0.53%
60%	2.98	2.403	0.226	260.6	20.924	11.694	4.49%
70%	3.01	2.484	0.253	183.28	20.706	7.499	3.46%
80%	2.73	2.275	0.222	160.78	20.901	7.574	4.38%
90%	3.37	2.780	0.301	99.09	20.656	5.815	3.23%
100%	3.54	2.604	0.291	481.29	24.117	17.098	17.13%

Table3: Ethernet test results

3.1.1 The impact of Jitter:

When RTP Packets are transmitted over the network, the delay may change, this delay variation or Jitter occur because it depends on the times of arrival of each single RTP packet, which is related to the network load, on the situations where the network is overloaded, the right receiving of the RTP packets in the bounded time intervals is not possible [8].

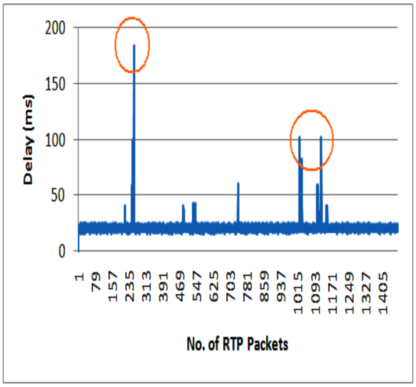
At lower network loads (0%, 10% and 20%) the average jitter is almost steady in a range between 2.63 to 2.66 ms, as the network load increases beyond 30%, the average jitter starts to change. The most obvious trend in this change which is quite different of what it is expected is the drop of the average jitter values with respect to the increase of network load, which dropped by 16 % at load 80%.

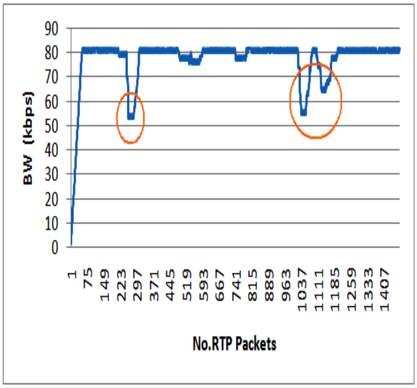
The Jitter gradually increase with respect to increase of the back ground traffic loads .In order to compare the jitter variability and how the data is spread around the mean, the standard deviation is calculated (shown in (Table 3). At load 0% where only VoIP traffic without any back ground traffic, the mean is 2.638 ms while the standard deviation is 0.206 ms.

As shown in (Table 3) the standard deviation is proportionately increases as the network load increases. The reason for this increase is the network is overloaded and receiving the RTP packets in the same time intervals becomes more difficult when more background traffic is added to the network, this delay variation has a major impact on real time application such as VoIP, because it is not possible for the receiving end to play out the RTP packet as soon as it receives it [10].

3.1.2 The impact of the delay and packet loss:

Delay and packet loss are two major factors that affect VOIP quality of service, the two factors are sometimes interrelated, the increase of the RTP packets delay with respect to the change in the background traffic load is clearly seen. The results shows that, even though the average RTP packets delay values are acceptable (below ITU recommendation 150 ms[2]), there are some RTP packets which have significantly higher delay values and it exceeded the acceptable limit.





High delay trends in Ethernet Impact of the packet loss

Figure: 3 Correlation between delay and packet loss in Ethernet

There are two reasons for the packet drop [7]. First reason is due to tail drop in the congested queues in the network devices interfaces (switches or router). The second reason is due to network congestion and line errors at one or more segments along the traffic route and because of time sensitive nature of VoIP traffic, packets are dropped at the receiver if they arrive too late to be used.

The important finding related to delay and packet loss which was observed in this study is the strong liner correlation between the delay and the packet loss. Measurement analysis and graphs to support this observation are produced, it is clearly visible from Figure 3 that whenever there is a sharp increase in the RTP packets delay, there is a strong correlation of packet loss after that.

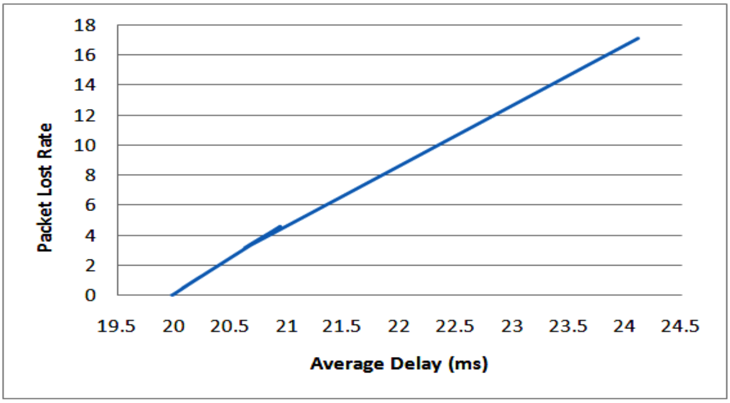


Figure: 4The Relationship between delay and Packets loss ratio in Ethernet

The average delay values versus network load are shown in (Tale 3). Also the packet loss rate values under different network loads is shown. The most obvious trends in these graphs is the simultaneous increase of the average delay and the packets drop which start at load 40%,The correlation and the strong linear relationship between the packet loss and the delay are clearly shown in (Figure 4)which represents the packet loss rate versus the average delay.

This finding matches with Roychoudhuriet all[9], where the authors conducted their experiments over the Internet in order to evaluate the effects and the correlation between delay, delay variation and packet loss; they have noticed that there is strong correlation between packet loss and delay. However, in [9] the authors mentioned that relationship is not
linear, which is different from what it is found in this study.

3.2Results and VOIP performance in WLAN:

In WLAN test, the network analyser wireshark on none promiscuous capture mode is used to collect data (jitter, delay values, andpacket loss rate). As shown in (Table 4) below , for all network loads the average jitter and delay values are calculated, inorder to observe the spread of the values around the average, standard deviation values are also calculated. For every network level load, a jitter and delay graphs are produced.

Network Load	Voice Jitter (ms)			Packet End-to-End Delay(ms)			Packet Loss Ratio
	MAX	AVERAGE	STDEV	MAX	AVERAGE	STDEV	
0%	3.11	2.628	0.234	26.55	19.988	2.947	0%
10%	2.96	2.649	0.220	26.36	19.981	2.914	0%
20%	3.99	2.493	0.444	39.88	19.998	3.440	0.07%
30%	4.62	2.979	0.523	45.2	20.055	4.232	0. 34%
40%	5.29	2.920	0.664	51.1	20.096	4.443	0.55%
50%	5.25	3.044	0.557	50.52	20.01	4.303	0.13%
60%	6.29	3.819	0.744	48.33	20.041	5.044	0.27%
70%	6.02	3.667	0.630	57.05	20.058	5.063	0.34%
80%	5.51	3.903	0.614	53.01	20.156	5.502	0.87%
90%	6.57	4.071	0.777	47.21	20.066	5.468	0.4%
100%	6.27	3.767	0.681	48.73	20.071	5.106	0.4%

Table: 4 Wireless test results

As shown in (Table 4) the packet loss values are represented in the percentage of the total RTP packets being transmitted and didn’t reach the destination. The jitter and delay values obtained from the WLAN test are organised in graphs representing the jitter and delay graphs versus the RTP packets number. The average jitter and delay values are organised in graphs representing the network load vs. the average values.

3.2.1 The impact of Jitter in wireless:

Jitter or delay variation, has significant negative effects on voice quality on WLANs [2], Jitter produced in the WLAN is the main part, because the WLAN is most probably the bottle neck in the network, the main reason of high jitter in wireless is due to the delay caused by the random channel service time,the wireless MAC protocol determine the time to be taken to transmit the frames over the WLAN [2, 8].

The dramatic increase in jitter values with respect to the increase of the background traffic loads is clearly visible in (Table4), from the information given in the (Table 4), it can be seen that at lower back ground traffic loads, the average jitter is almost steady. It is noticeable that the WLAN congestion point started at load 20%, because of the sudden increase of the jitter and delay values and the start of RTP packet lost with a value of 0.07% at this load level. The dramatic increase in average jitter values is clearly shown in the (Table 4), at load 50% average jitter increased by 16%, at load 90% the average jitter increased by 55% which is the peck value.

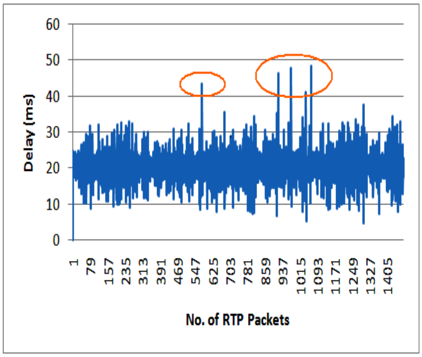
3.2.2The impact of the delay and packet loss in wireless:

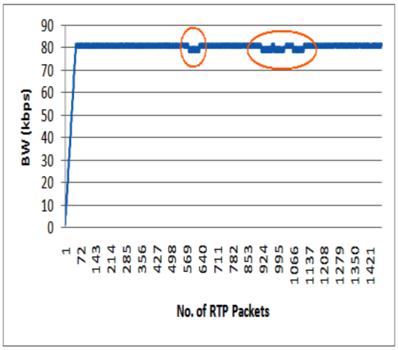
At load 0% the average delay is 19.9 ms when only VOIP traffic in the WLAN, the increase of the delay starts at load 20%, packets loss also starts at the same load, the maximum delay reached at load 80% with increase of 0.84 %.

The most obvious change observed is the fast change in the RTP packet delay, even though the change in the average delay with respect to increase in the load is no high, however, the variation of the delay between the RTP packets is clearly noticeable.

As it was mentioned that the WLAN becomes congested at lower loads compared with the Ethernet where the VoIP traffic starts struggling at load 40%. The reason behind this is because DCF which based on the carrier sense multiple access with collision avoidance (CSMA/CA) mechanism is used in 802.11 networks to control access to the radio frequency medium, DCF was not designed to accommodate a time sensitive application such as VOIP.

Guo et al [5] defined two main reasons for delay in WLANs, first due to random back-off and collision it is difficult to have full control over the exact transmission timing of the voice frames, at high background traffic loads, the collision will increase and consequently the packet delay and packet loss rate will increase.





High delay trends in Wireless **Impact of packet loss in wireless**

Figure: 5 Correlation between delay and packet loss in Wireless

The second reason they have mentioned is because all the traffic in the WLAN is best effort and there is no traffic prioritization. Therefore, large volume traffic such as FTP will consume a large proportion of the bandwidth and as result this will have significant effects on the voice traffic.

The correlation between the RTP packets delay and the packet loss which is observed in Ethernet also strongly exists in wireless. It is clearly noticeable from(Figure 5) that every time there is a sharp increase in the delay, it followed by a packet loss after that which is clearly shown in call stream bandwidth on (Figure 5). The important observation related to the correlation between delay and packet loss rate in wireless which is not clearly visible in Ethernet, is the delay threshold which determines the start of the packet loss. It is noticeable that the packet drop starts whenever the delay exceeds approximately 40ms.

The Packet loss increases sharply at load 20% where the network is overloaded with traffic. Even though the packet loss values are acceptable (generally below 3% [6]). However, when the number of RTP packets reaches the maximum level that the network can accommodate the drop packet may become extremely high.

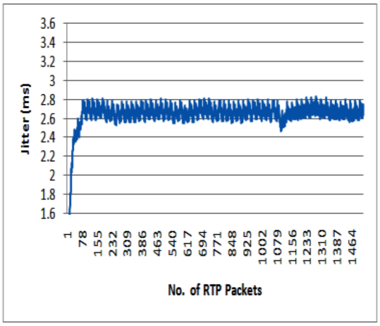
The strong linear relationship between the RTP packets delay and the packets loss rate is clearly observed shown in (Table 4), also the correlation is noticeable. Because the changes in both values trends is almost identical. From the (Table 4) it is clearly visible that it is linear relationship, the justification for this strong linear relationship is as it was previously explained, when the back ground traffic load increase, more RTP packets will arrive late to be used and will be discarded by the receiver end.

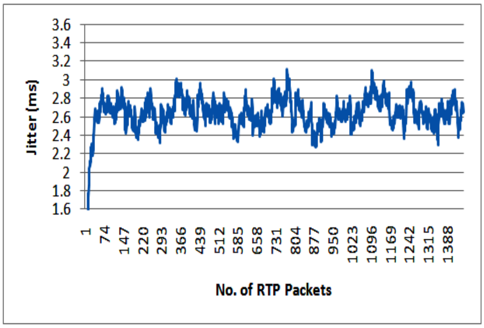
3.3 Comparison of VOIP performance between Ethernet and Wireless.

In the previous sections, analysis of VOIP traffic performance in both Ethernet and wireless provided by discussing the impact of jitter, delay and packet loss with respect to the increase of the background traffic loads. In this section comparison between VOIP performance in Ethernet and in wireless conducted. The aim from this comparison is to find out what level of quality of service wireless LANs can provide with respect to its counterparts wired LANs.

As it was shown in (Table3) the maximum network load reached in Ethernet is 89 Mbps, where only 21Mpbs maximum load achieved in wireless. To precisely compare VOIP performance in both network environments with respect to the change in background traffic loads, the comparison should be conducted under the same network loads condition. For this purpose the comparison conducted under a maximum load of 21 Mbps in both Ethernet and wireless.

3.3.1 Comparison of Jitter between Wireless and Ethernet:

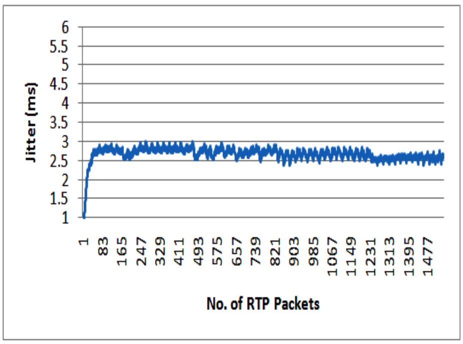




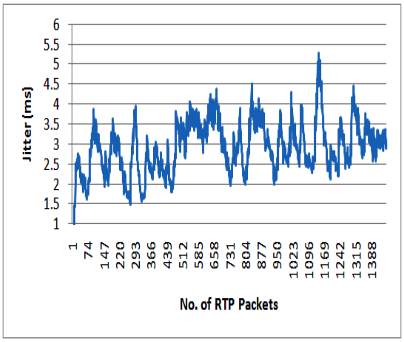
Jitter in Ethernet (Load 0%) **Jitter in WLAN (Load 0%)**

Figure: 6 Comparison of RTP packets Jitter between Wireless and EthernetWithout back ground traffic

Figure 6shows the RTP packets jitter graphs in both wireless and Ethernet where only voice traffic traverses the network. Even though there is not a big difference between the average jitter in both graphs which is 2.638 ms in Ethernet and 2.628 ms in wireless, there is obvious



difference in RTP packets jitter between the two graphs. The main difference is in the distribution of the RTP packet jitter around the average values.



Jitter in Ethernet (Load 8 mbps) Jitter in wireless (Load 8 mbps)

Figure: 7 comparison of RTP packets Jitter between Wireless and Ethernet with background traffic at load 8%

Figure 7shows the RTP packets jitter after loading both networks with the same (load 8) Mbps of large volume traffic (FTP traffic).There is a significant difference between the jitter in both graphs, the average jitter in Ethernet increased with only 0.9%. On the other hand jitter in wireless increased with 11% which is 12 times the increase in Ethernet.

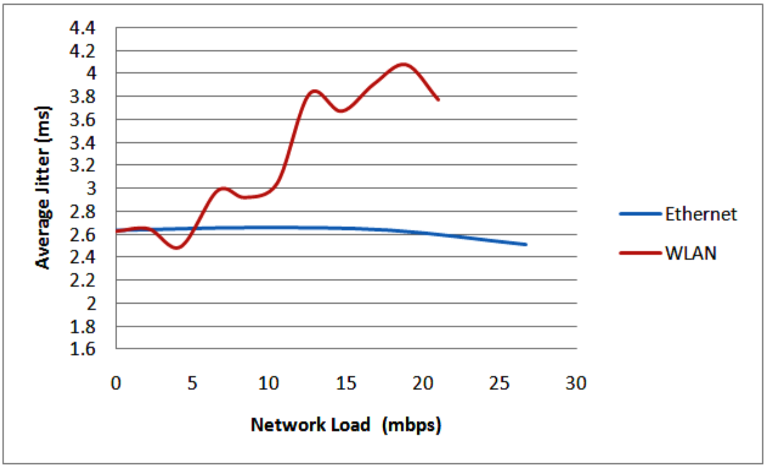


Figure: 8comparative plot of average Jitter on Wireless and Ethernet

The difference in average jitter between Ethernet and wireless under different background traffic loads is shown in (Figure 8). From the graph it can be seen that at lower network loads the average jitter is almost equal in both wireless and Ethernet. As soon as the injection of FTP traffic increased, the wireless jitter rapidly increased, it increases by 54.9%.

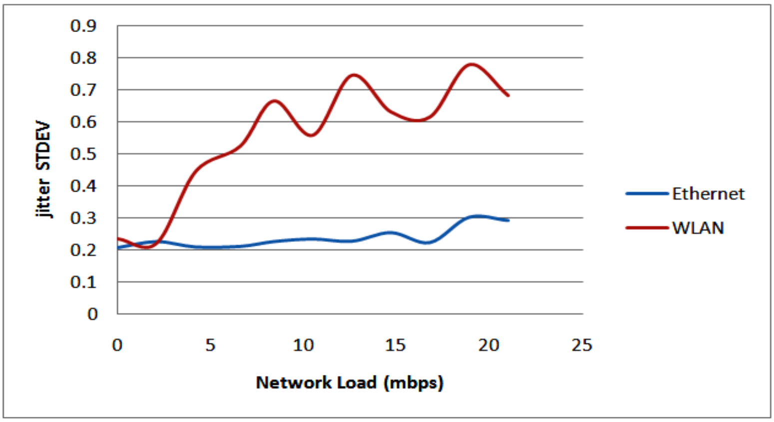


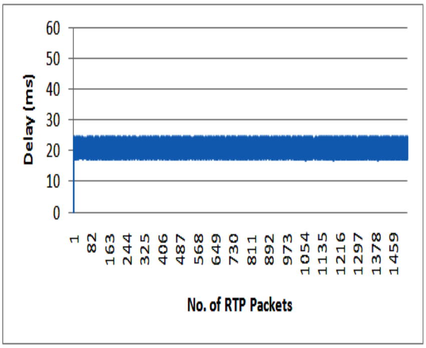
Figure: 9 comparative plot of Jitter standard deviation on Wireless and Ethernet

On the other side jitter in Ethernet remains almost steady with respect to increase in the load. To compare the spread of the RTP packets jitter values around the main average jitter value in both network environments, the jitter standard deviation graphs are produced the difference in standard deviation is clearly shown in(Figure 9).

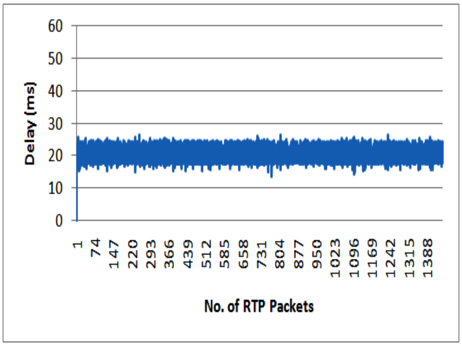
3.3.2 Comparison of delay between Wireless and Ethernet.

End to end delay and packet loss are the most important factors which have a significant impact on the quality of the voice [9], the impacts of these factors in both Ethernet and wireless were previously shown in this paper it was found that in Ethernet due to network line congestion and errors at the network segments along the RTP packets way and because of VoIP traffic time sensitivity, the RTP packets which arrive late are dropped. The evidence for this finding is the strong linear relationship between the delay and the packet loss rate.

In wireless also it is proven that there is strong linear relationship between the packets loss and the RTP packets delay. Comparing impact of delay and packet loss on RTP packets between Ethernet and wireless by presenting and discussing the delay graphs with and without background traffic on both network environments.



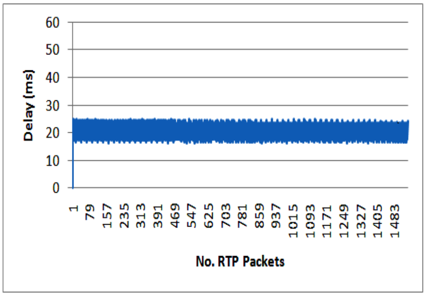
Delay in Ethernet (Load 0)

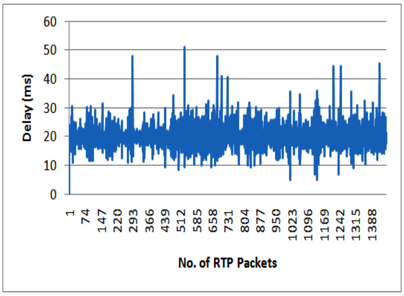


Delay in WLAN (Load 0)

Figure: 10 Comparison of RTP packets delay between Wireless and Ethernet without background traffic

Figure 10 shows the delay on both wireless and Ethernet when only the VOIP traffic in the network. On Ethernet it is clearly visible that the delay graph is moving smoothly because all RTP packets experience the same delay values. In wireless, even though no back ground traffic existed in the network, there is difference between the RTP packets delay values.





Delay in Ethernet (Load 8 mbps) **Delay in WLAN (Load 8 mbps)**

Figure: 11 Comparison of RTP packets delay between Wireless and Ethernet with Load 8 mbps

After loading both networks with the same load (8 mbps)**.**It is noticeable from (Figure 11) that there is no change in average RTP packets delay on the Ethernet, the average delay value remains the same 19.98 ms.On the other hand, there is a significant increase in the wireless side. As shown in (Figure 11), some RTP packets delay is higher than 40 ms which caused 0.55% packet loss, after loading the network the average delay in wireless increased with 0.54% and the standard deviation increased with 50%.

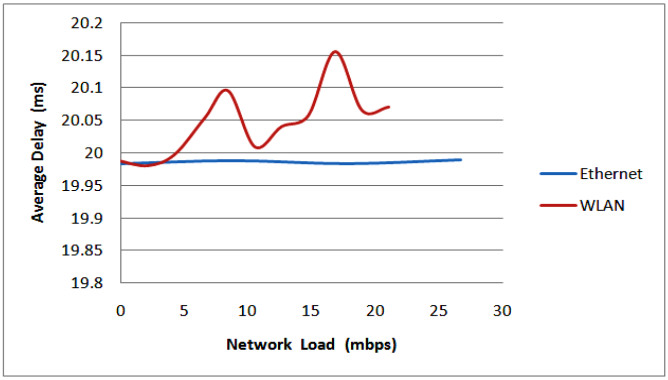


Figure 12 Comparative plot of average delay in Wireless and Ethernet

The overall delay comparison between the delay in Ethernet and wireless is shown in (Figure 12). The Ethernet average delay remains steady with respect to the increase in the background traffic loads. On the other hand as the network load increases the wireless average delay rapidly increases, it is increased by 0.84 %.

Collision and frames retransmission, voice packets will not arrive on time to be used. On both network environments there is strong linear relationship between delay and packet loss, delay thresholds which determine the packet lost is also visible especially in wireless where it was defined by 40ms. From the comparison between the performances of VoIP traffic in both networks environments it is clearly visible that the performance of wireless is much worse than in wired. The average jitter in wireless is 54.9% higher than in Ethernet and the RTP Packet delay is 0.84 % higher than in Ethernet.

Conclution:

By observing the performance results achieved from the experiments, and from the previously demonstrated analysis, it was found that, as one would expect, the best values for the QoS parameters on both network environments are obtained at 0% network load. On Ethernet ,VoIP traffic start suffering packet loss and high delay trends at load 40%, although the average RTP packet delay and jitter values are acceptable even at the high loads, however there are some RTP packets which have significantly high delay and jitter values .

The main possible reason for this high delay values and packet loss in Ethernet is due to network line congestion on the network segments along the RTP packets way, and because of VoIP traffic time sensitivity, the RTP packets which arrive late are dropped. The evidence for this finding is the strong linear relationship between the delay and the packet loss rate which was observed.

On wireless, despite the low throughput achieved, the high delay and packet loss starts relatively early, precisely at load 20%, average jitter and delay values are rapidly increased with respect to the increase of the background traffic loads. The performance is clearly worse than the Ethernet under the same load conditions. As it was explained , that the reason behind the worse performance of VoIP traffic on wireless is due to the 802.11 MAC characteristics, at the high background traffic loads it becomes difficult to control the exact transmission timing of the voice frames, because the collision will increase and consequently the packet delay and packet loss rate will be increased.

In this study it is noticeable that there is a strong correlation between packet loss ratio and RTP packet delay, this correlation is clearly visible in both Ethernet and wireless, Another noticeable feature related to this correlation is the delay thresholds which determine the packet loss, from the delay graphs it is clear that 40ms is the delay threshold which determines the start of the packet lost.

دراسة حركة الصوت عبر بروتوكول الانترنت
في الشبكات السلكية والا سلكية

بالقاسم سالم محمد المنتصر

المستخلص:

هذه الورقة عبارة عن دراسة عملية لتحليل أداء حركة اتصالات الصوت عبر الانترنت(Voice Over IP (VoIPعبر الشبكات المحلية السلكية واللاسلكية من خلال مراقبة معاملات جودة الخدمة (QoS) للصوت عبر بروتوكول الإنترنت، وتشمل هذه المعاملات كلاً من التأخير و نسبة اضطراب وفقدان حزم الصوت المرسلة عبر الشبكة. تم اختيار اثنين من التكنولوجيات المنتشرة على نطاق واسع في الشبكات المحلية (الايثرنت802.3IEEE في الشبكات السلكية وIEEE 802.11G في الشبكات اللاسلكية) كبيئة اختبار لإجراء هذه الدراسة، بالنسبة لمنهجية الاختبار التي تم اعتمادها تعتمد على محاكاة حقيقية لحركة الصوت في كلا الشبكتين السلكية واللاسلكية، وبالتالي فان الدراسة أجريت في المختبر باستخدام معدات شبكيةحقيقية.وللوصوللتقييم لأداءحركة مرور الصوت عبر بروتوكول الإنترنت، تم قياس معاملات جودة خدمة الصوتمن خلال ملاحظة حزم RTP تحت تأثير أطر مختلف منالأحمال، ودلك باستخدام حركة مرور FTP في خلفية الشبكة. وقد نتج عن هذه التجربة مجموعة من الاحصاءات والرسوم البيانية التي تمثل كلاً من التأخير ونسبة اضطراب وفقدان حزمRTP تحت تأثير احمال مختلفة .

أظهرت نتائج الاختبارات التي اجريت تحت تأثير احمال عالية، بان حركة الاتصالات عبر بروتوكول الإنترنت (VoIP) في كلاالشبكتين السلكية واللاسلكية تعاني من نسب تأخير عالية و اضطراب وفقدان لحزم الصوتRTP. أيضا أظهرت النتائج أن أداء حركة الصوت عبر بروتوكول الإنترنت في الشبكات اللاسلكية هو أسوأ بكثير من الأداء في شبكات الإيثرنت تحت ظروف الحمل ذاتها. وكان الملاحظ أن هناك علاقة خطية قوية بين تأخير حزم RTP ونسبة فقدان هذه الحزم في كلا الشبكتين الإيثرنت واللاسلكية.

References:

1. Bandwidth Monitor. (2009) [Online]. Available at: (Accessed: 10 July 2009).
2. Cai, L., Xiao, Y., Shen, X. S. & Mark, J. W. (2006) 'VoIP over WLAN: Voice capacity, admission control,QoS, and MAC', International Journal of Communication Systems, 19 (4), pp. 491 - 508 InterScience [Online]. Available at: (Accessed: 22 August 2009).
3. The enterprise is ready for wireless VoIP.Is wireless VoIP ready for the enterprise? (No date) [Online]. Available at: (Accessed: 14 July 2009).
4. FileZilla the free solution. (2009) [Online]. Available at: (Accessed: 23 August 2009)
5. Guo, F. & Chiueh, T. (No date) Comparison of QoS Guarantee Techniques for VoIP over IEEE802.11 Wireless LAN. [Online]. Available at: (Accessed: 21 August 2009).
6. NEC Unified Solutions (No date) What VOIP Requires From a Data Network. [Online]. Availableat: (Accessed: 17 August 2009).
7. Ortiz, C., Frigon, J.F., Sans, B. & Girard, A. (2008) 'Effective Bandwidth Evaluation for VoIP Applications in IEEE 802.11 Networks', Wireless Communications and Mobile Computing Conference, 2008. IWCMC '08. International, pp. 926-931 IEEE [Online]. Available at: (Accessed: 15 August 2009)
8. Rango, F. D., Tropea, M., Fazio, P. & Marano, S. (2006) 'Overview on VoIP: Subjective and Objective Measurement Methods', International Journal of Computer Science and Network Security, 6 [Online]. Available at: (Accessed: 25July 2009).
9. Roychoudhuri, L., Al-Shaer, E. & Brewster, G. B. (2006) 'On the impact of loss and delay variation on Internet packet audio transmission', Computer Communications, 29 (10), pp. 1578-1589 Science Direct [Online]. Available at: (Accessed: 1 August 2009).
10. Zubairi, J. A. & Zuber, M. (2000) Suny Fredonia campus network simulation and performance analysis using OPNET. [Online]. Available at: (Accessed: 1 August 2009).