Simulation and Analysis of QoS Video Conference through IMS Interworking Network – UMTS

Haider M. Turki AL Hilfi and Emad Alnawafa
Dep. Telecommunication, Faculty of Electronics, Telecommunication & Information Technology
University 'Polytechnic' of Bucharest, Romania
Email: hi.1977@yahoo.com; emad_nawafah@yahoo.com

Abstract —In this article, video conferencing over IMS - UMTS network is simulated using OPNET modeler 14.5 software and analyzed for the required QoS. The model used is the Integrated Services (IntServ) model to observe IMS and Differentiated Service (DiffServ) architecture to observe the relationship between IMS and UMTS. The simulation is run for different combination of users (8, 16 and 32) users with different simulation time such as (200, 400 and 600) seconds. From the results, it is observed that IntServ architecture produced the largest average value of 26.754 Mbit /s for the throughput, E2E delay of 233.178 ms, and jitter of 0.084 ms. The DiffServ model produced a throughput of 8,595 Mbit / s, E2E delay of 164,249 ms, and jitter of 0.599 ms. Packet loss is found to be less than 0.95% in both the architectures. It is inferred that according to ITU-T G-114 standard, the average performance parameter values in both the models are acceptable.

Index Terms—IMS, UMTS, IntServ, DiffServ, Video conference.

I. Introduction

The proposed work is a development of research "High Quality of Service Video Conferencing over IMS" that has been done previously by [1], [2]. In the study, various performance metrics such as delay, jitter, and packet loss is analyzed for video conferencing service and compared for two different architectures on IMS networks [3]. The main focus of this study is to analyze interworking between IMS and UMTS for video conferencing and voice services in terms of QoS parameters such as delay, jitter, packet loss, and throughput. The scenarios used in this research are Integrated Service IntServ and DiffServ. IntServ architecture is used to analyze the QoS of IMS network, whereas DiffServ model analyzes the relationship between IMS and UMTS. As mentioned earlier, the simulation is run for different combination of users (8, 16 and 32 users) with different simulation time such as (200, 400 and 600) seconds.

The results will then be compared with ITU-T standards. In the literature, three types of QoS models are often used, namely best-effort service, integrated service, and differentiated service. Best-Effort Service is used to do all the effort in order to send a packet to a destination. This method does not guarantee that the packet will reach the intended recipient. IntServ provides service-level applications through E2E network parameter negotiation, whereas DiffServ provides a set of classification tools and queuing mechanisms against protocols or applications with certain priorities over different networks [3].

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QoS parameters are used to determine the quality of a network in terms of delay, jitter, throughput, and packet loss. Delay is the total time required for the information or data to propagate from sender to receiver on a network. Delay variation is the interval of arrival between packets at destination. Throughput represents the total arrival of a successful packet until a goal over a given time interval is divided by the duration of the time interval. Packet loss is the number of packets lost during the delivery process. The following is a standardized value of service quality based on ITU-T G.114 [4], [5].

II. Research Methods

A. Parameters of Research

In this research, the simulation is done using IntServ and DiffServ architectures. Services to be observed are video conferencing services on IMS networks and interworking between IMS and UMTS. The performance is evaluated in terms of parameters namely delay, jitter, packet loss, and throughput. The study was conducted with three different times, (200, 400 and 600) seconds with different number of users (8, 16 and 32)users.

Table I shows the concept of data retrieval used in this study.The research method used is experimental method that is by simulation of IMS network and interworking IMS - UMTS with IntServ and DiffServ scenario using OPNET Modeler 14.5.

B. IMS Architecture

The IMS framework composes of many layers such as Transport Layer, Service or Application Layer, IMS Layer and the User Layer. The IMS layer will act as a control layer or center between Service Layer and Transport Layer as shown by Fig. 1. The Access layer of different access technologies will be attached in the Transport Layer. The user end device (UE) will be located at the User Layer. The signaling and service management are needed to connect between users across different access technologies at the Transport layer through the IMS layer [12]. Then the IMS will allow users to access services layer without the limitation of different access technologies of the users.

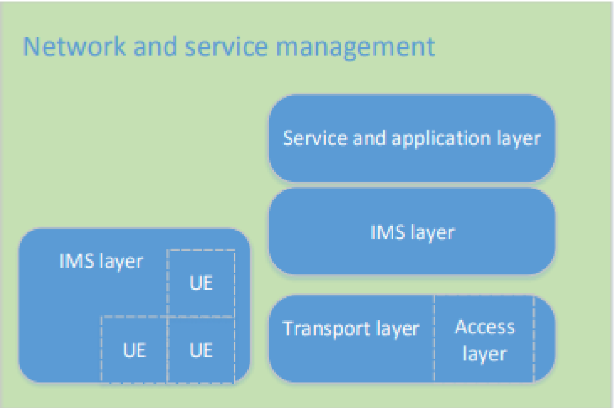

Fig. 1. Simplified IMS Framework [12].

TABLE I. CONCEPT OF DATA RETRIEVAL

Scenario	IntServ	DiffServ
Parameter	Throughput, E2E Delay, Jitter, and Packet Loss	Throughput, E2E Delay, Jitter, and Packet Loss
Number of Users	8, 16, and 32 users	8, 16, and 32 users
Simulation Time	200, 400, and 600 (Sec)	200, 400, and 600 (Sec)

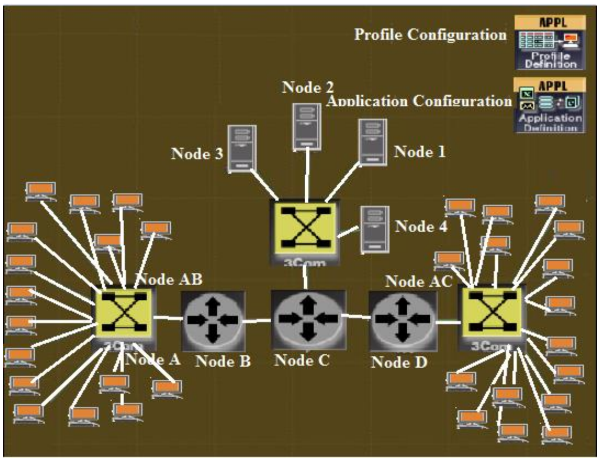

Fig. 2. IMS network topology on IntServ Scenario.

a) Intserv scenario model

The configuration of the IntServ scenario topology design is as follows.

1. Application Configuration.

Application components are used to define the services used. In this research, the service used is video conference service. Video conferencing services are real time services with video and voice traffic classes.

b) DiffServ scenario

DiffServ scenario will create a topology between different networks of IMS and UMTS. Fig. 3 shows the IMS and UMTS network topologies in OPNET Modeler 14.5 using 32users.

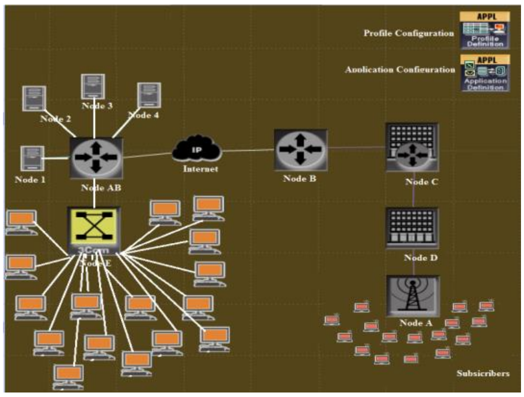

Fig. 3. IMS network topology - UMTS on Diffserv Scenario.

In the DiffServ model, the same configuration settings will be used as mentioned earlier in the case of profile configuration, SIP proxy server configuration and user configuration.

2. Application Configuration.

In the DiffServ scenario, ToS is replaced by using the DS Code Point option in which the code is used to prioritize the service. The Application configuration is shown in the ToS section. Type of service in the DiffServ scenario for video services is AF43 because the service used is video conferencing services where video conferencing services are prioritized after voice service. While voice using EF indicates that voice is prioritized rather than video service.

III. RESULTS AND DISCUSSION

A. IntServ Scenario

From the results it is observed that IntServ scenario results in throughput values of 1.486 Mbit / s, 15,083 Mbit / s, and 26,754 Mbit /s for (200, 400 and 600) seconds respectively. From the results, it is observed that IntServ architecture produced the largest average value of 26.754 Mbit /s for the throughput, E2E delay of 233.178 ms, and jitter of 0.084 ms. The DiffServ model produced a throughput of 8,595 Mbit / s, E2E delay of 164,249 ms, and jitter of 0.599 ms. Packet loss is found to be less than 0.95% in both the architectures. It is inferred that according to ITU-T G-114 standard, the average performance parameter values in both the models are acceptable. In this study, the longer the simulation time the greater the value of throughput generated.

This is because packets are sent from sender to receiver. From the simulation results, when compared between 16 users and 32 users on 400 and 600 seconds time respectively, it can be inferred that more the number of users, smaller the throughput value. In the case of 200 seconds simulation time, if the numbers of users are more, the throughput is greater. The situation may be due to the instability of the network because the traffic in this study is generated at 70 seconds while the simulated time is taken for 200 seconds. Fig. 4 depicts a graph that illustrates the change in the throughput value of a video conference service.

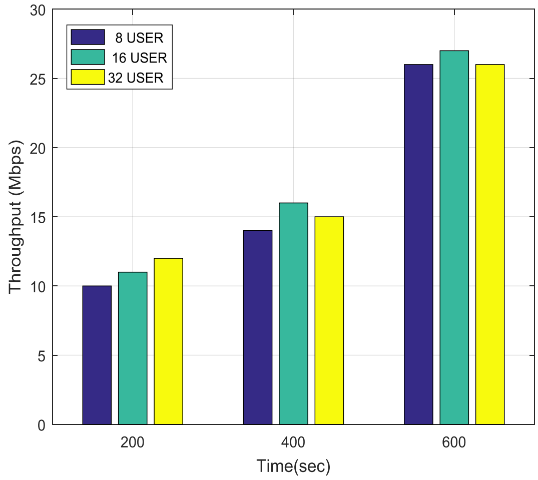

Fig. 4. Throughput on the IntServ Scenario.

E2E Delay estimation for a duration of 200 seconds on video conferencing services with different users produced acceptable results that fulfill the standards and is included in good category because the resulting values are less than 150 ms. The average value for 200 seconds simulation is 144.684 ms. For simulation time set to 400 seconds, the resulting value is found to be more than 150 ms and hence included in the category of enough, that is between 150 to 400 ms. The average value for 400 seconds simulation is 196.472 ms. Likewise for 600 seconds, each user group obtained an average value of more than 150 ms but not more than 400 ms so it is being included in the category enough and still acceptable. The average value for 300 second simulation is 233.178 ms. from these results it can be inferred that the communication process for video conferencing services on IMS network can still run well.

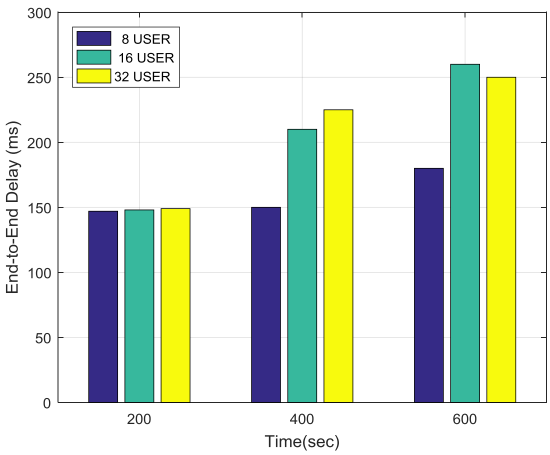

Fig. 5. E2E delay on the IntServ Scenario.

In this study, it is inferred that for video conferencing services the performance metric E2E delay for different simulation time increases with the number of users. Longer the simulation time, the greater the end-to-end delays. The greater the value of the resulting delay will cause the resulting throughput smaller. This can be seen in a 600 second simulation using 16 and 32 users, where the delay in 32 users is greater than 16 users so that the throughput of 32 users is smaller compared to 16 users case. Fig. 5 shows a graph that can illustrate the value of end-to-end delay for video conferencing services.

The jitter value evaluated through simulations considering different run times of (200, 400 and 600) seconds using (8, 16 and 32) users were included in good category, whereas jitter values less than 20 ms are included in either category. From these results it can be concluded that the communication process for video conferencing services on the IMS network can still run well when using 32 users because the resulting jitter value were included in either category.

Jitter occurs due to the difference in arrival time between packets at the recipient side. The greater the value of jitter the greater the value of the resulting delay. Communication process will be better if the resulting jitter value is small. Fig. 6 shows a graph depicting a change in the value of jitter in the IntServ scenario.

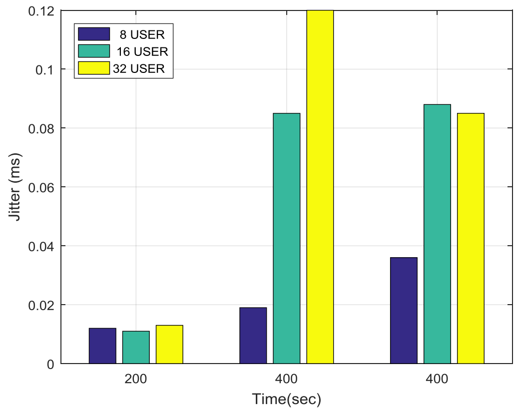

Fig. 6. Jitter for IntServ Scenario

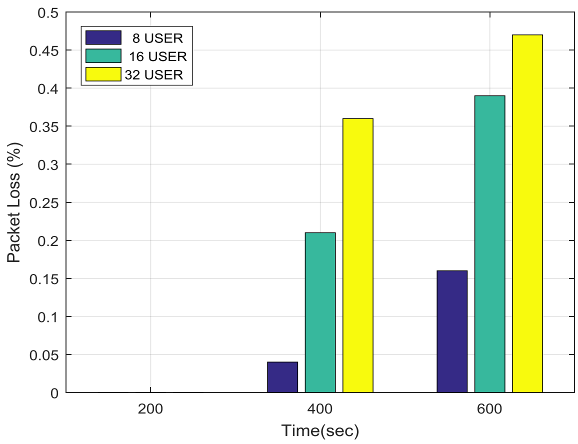

Fig. 7. Packet Loss on the IntServ Scenario.

The packet loss value less than 1% is included in either category. From these results it can be said that the communication process for video conferencing services on the IMS network can still run well when using 32 users because the packet loss value is less than 0.95%, not much data is lost. Fig. 7 shows a graph depicting the change in packet loss values in a video conferencing service.

In Fig. 8 for a time of 200 seconds, the packet loss value is 0.001486% when using 8 users, 0.001584% when using 16 users, and 0.001489% when using 32 users. It can be inferred that there is significant changes in the performance for different simulation times

TABLE II: THROUGHPUT, E2E DELAY AND JITTER AND FINALLY PACKET LOSS ON INTSERV SCENARIO

Throughput Value			
Simulation Time	*Throughput* (Mbit/s)		
	8 user	16 user	32 user
200 Sec	0.973	1.492	1.992
400 Sec	13.448	16.438	15.663
600 Sec	26.442	27.357	26.464
E2E Delay Value			
Simulation Time	Rata-Rata *Delay* (ms)		
	8 user	16 user	32 user
200 Sec	143.294	144.287	146.529
400 Sec	152.718	209.888	226.586
600 Sec	186.855	264.198	248.285
Jitter Value			
Simulation Time	Rata-Rata *Jitter* (ms)		
	8 user	16 user	32 user
200 Sec	0.00319	0.00217	0.00436
400 Sec	0.02816	0.09843	0.11487
600 Sec	0.04558	0.09532	0.09998
Value of Packet Loss			
Simulation Time	Rata-Rata *Packet Loss* (%)		
	8 user	16 user	32 user
200 Sec	0.001486	0.001584	0.001489
400 Sec	0.036456	0.227379	0.382818
600 Sec	0.188783	0.398655	0.483898

B. *DiffServ Scenario*

It is observed that the throughput in the case of DiffServ architecture for 200 seconds is 0.259 Mbit/s, for 200 seconds it is 3.588 Mbit/s and for 300 second the average value is 8,595 Mbit / s.

From the values it is inferred that, longer the simulation time, the greater is the value of throughput. This is because more packets are sent from sender to receiver. From the simulation results, it can be depicted that when moving from 16 users to 32 users throughput value goes down. Fig. 8 shows a graph illustrating the change in throughput values in the DiffServ scenario.

E2E Delay in DiffServ scenario for simulation run times 200 seconds with usage of (8, 16 and 32) user got result which still fulfill standard and included in good category because value yield less than 150 ms. The average for 200 second simulation is 125.245 ms. For a 400 seconds simulation, the average obtained is 153,958 ms. As the resulting value exceeds 150 ms, this case is included in the category enough and still acceptable. For 600 seconds simulation a value of more than 150 ms but less than 400 ms is achieved, hence it is in the category, sufficient and acceptable. The average for a 600 second simulation is 164.249 ms.

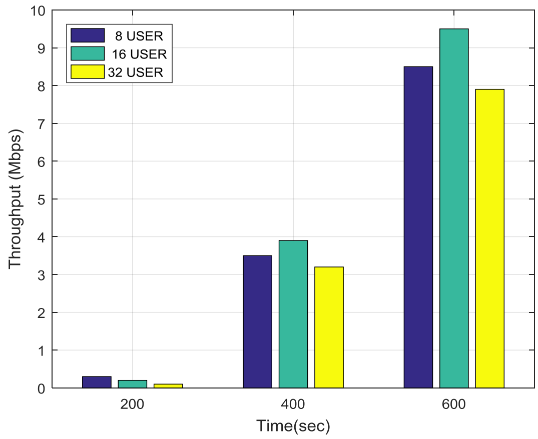

Fig. 8. Throughput on the DiffServ Scenario.

From these results it can be said that the communication process for video conferencing services on the IMS - UMTS network can run with good quality because the resulting E2E delay is still acceptable. Figure 9 shows a graph that illustrates the E2E delay for video conferencing services in the DiffServ scenario.

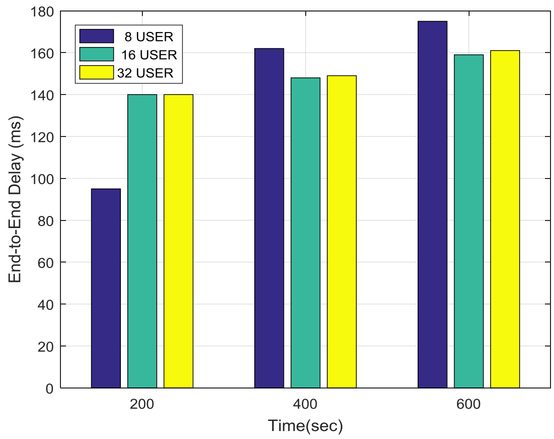

Fig. 9. E2E on the DiffServ Scenario

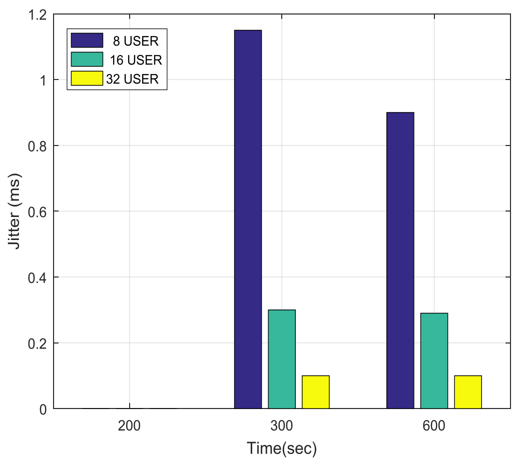

Fig. 10. Jitter on the DiffServ Scenario.

The evaluated results for simulation time (200, 400 and 600) seconds using (8, 16 and 32) users are included in either category because the resultant value is not more than 20 ms. From these results it can be said that the communication process for video conferencing services on IMS - UMTS network can run well because the resulting jitter value is acceptable and included in either category. Fig. 10 shows a graph illustrating the change in jitter value in the DiffServ scenario.

From Fig. 10 it can be inferred that, the jitter value simulation time 200 seconds when using 8 users produced a value of 0.0000098 ms, using 16 users resulted in 0 ms, and with 32 users is 0.0000019 ms. On comparing these results with 400 seconds and 600 seconds simulation time, there is a significant difference noted as seen in the graph. Differences in value occur due to differences in the amount of data packets in the case of 400 seconds and 600 seconds.

TABLE III: THE THROUGHPUT, E2E DELAY AND JITTER AND FINALLY PACKET LOSS VALUE OF THE DIFFSERV SCENARIO.

Throughput Value			
Simulation Time	**Rata-Rata *Throughput* (Mbit/s)**		
	8 user	**16 user**	**32 user**
200 Sec	0.394	0.348	0.333
400 Sec	3.595	3.488	3.678
600 Sec	8.878	8.358	8.547
E2E Delay Value			
Simulation Time	**Rata-Rata *Delay* (ms)**		
	8 user	**16 user**	**32 user**
200 Sec	94.161	140.743	140.832
400 Sec	162.822	148.787	150.271
600 Sec	173.975	156.482	162.288
Jitter Value			
Simulation Time	**Rata-Rata Jitter (ms)**		
	8 user	**16 user**	**32 user**
200 Sec	0.0000098	0	0.0000099
400 Sec	1.178	0.311	0.089
600 Sec	0.999	0.386	0.099
Packet Loss Value			
Simulation Time	**Rata-Rata Packet Loss (%)**		
	8 user	**16 user**	**32 user**
200 Sec	0.0023	0.0009	0.0008
400 Sec	0.9953	0.3128	0.3155
600 Sec	0.2963	0.3516	0.3462

The packet loss value for simulation time (200, 400 and 600) seconds using (8, 16 and 32) users are included in good category. The packet loss value less than 0.95% is included in either category. From the table each user has a difference value that is not too significant. The difference in value is not influenced by the number of users but more difference is noticed when the simulation time is increased. These results indicate that the communication process for video conferencing services on IMS - UMTS network can run well because the resulting packet loss value is less than 0.95% so there is not much data loss.

Fig. 11 shows a graph depicting the change in packet loss values for video conferencing services in the DiffServ scenario.

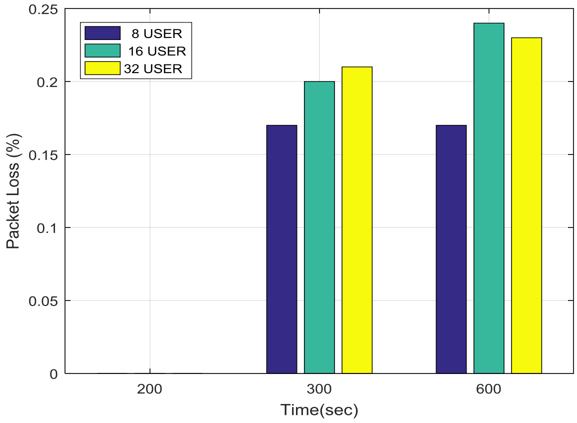

Fig. 11. Packet Loss on the DiffServ Scenario.

In Fig. 11 for simulation time 200 seconds, the packet loss value is 0.0011% when using 8 users, 0.0008% when using 16 users, and 0.0006% when using 32 users. On comparison, the results corresponding to 400 seconds and 600 seconds, shows significant results.

IV. CONCLUSION

The simulation results for IntServ and DiffServ architectures shows better results. This is demonstrated by the feasibility of IMS - IMS network and IMS - UMTS network which is marked by obtaining the value of QoS parameters in accordance with ITU-T G.114 standard. So the communication process by using video conferencing services on the network can run well. From the results, it is observed that IntServ architecture produced the largest average value of 26.754 Mbit /s for the throughput, E2E delay of 233.178 ms, and jitter of 0.084 ms. The DiffServ model produced a throughput of 8,595 Mbit / s, E2E delay of 164,249 ms, and jitter of 0.599 ms. Packet loss is found to be less than 0.95% in both the architectures. It is inferred that according to ITU-T G-114 standard, the average performance parameter values in both the models are acceptable.

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Haider Mohammed Turki Alhilfi was born in Basra, Iraq, in 1977. He received the B.S. degree in communication engineering from the 'Al-Furat Al-Awsat Technical University AL Najaf, Iraq, in 2009 and the M.T. Degree in wireless communication engineering from the University (SHIATS) Science and Technology, India, 2014. He is currently a Ph.D. student in Department of Telecommunications Engineering, University "Polytechnic" of Bucharest, Bucharest, Romania. His research interests include wireless network, wireless sensor networks, security and key management of wireless networks and wireless Ad-hoc networks.

Emad S. Alnawafa was born in Madaba, Jordan, in 1986. He received the B.S. degree in Computer Engineering from the Ablaqa' Applied University, Al-Salt, Jordan, in 2009 and the M.S. degree in computer engineering from the Jordan University of Science and Technology, Irbid, Jordan, 2013. He is currently a Ph.D. student in Department of Telecommunications Engineering, University "Polytechnic" of Bucharest, Bucharest, Romania. His research interests include wireless network, wireless sensor networks, security and key management of wireless networks and wireless Ad-hoc networks.