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# Performance Evaluation of Multimedia Streams Over Wireless Computer Networks (WLANs)

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***Abstract***

*The field of WLAN has grown significantly in the last three decades. An interesting usage of computer networks is in offices and educational institutes were increased to exchange streams of multimedia data. These privately-owned networks are known as Local Area Networks (LANs) which come under the category of small scale networks but researchers introduced the possible usage of wireless technologies for interconnection. This has resulted in the emergence of wireless LANs (WLANs). In the world of information system, performance is a very important subject. It consists of the work which the computer WLAN system is supposed to do, and how do we judge whether is being done well. To achieve performance in the field of WLAN, certain features must be available and operational in the design of the WLANs. Performance must also be considered when configuring and implementing the WLAN systems. The modeler of such WLAN systems must follow a systematic approach in evaluation, measurements, and analytical or simulation modeling of such WLAN system. This paper presents a comparison and an evaluation of transmitting multimedia streams over three different WLANs scenarios by using OPNET simulator*

## 1. Introduction

Universal access to multimedia information (multimedia streams) is now the principle motivation behind the design of next-generation computer and communications networks. Furthermore, products are being developed to extend the capabilities in all existing network connections to support multimedia traffic. This is a profound paradigm shift from the original analog-voice telephony network developed by the Bell System and from the packet-switched, data-only origins of the Internet [1]. The rapid evolution of these networks has come about because of new technological advances, heightened public expectations, and lucrative entrepreneurial opportunities.

Many researchers and manufacturing agencies interested in the transmission of multimedia streams over networks. By multimedia, we mean data, voice, graphics; still images, audio, video, and we require that the network support the transmission of multiple media, often at the same time. Two observations can be made at the outset. The media to be transmitted, often called sources, are represented in digital form, and the networks used to transmit the digital source representations may be classified as digital communications networks, even though analog modulation is often used for free space propagation or for multiplexing advantages. In addition to the media sources and the networks, we will find that the user terminals, such as computers, telephones, and personal digital assistants (PDAs), also have a large impact on multimedia communications and what is actually available.

The rest of paper is organized as follows. In section II, we introduce brief description of the multimedia communication problem. Computer networks are described in section III. Performance engineering and evaluation methodology is found in section IV. Simulation results are discussed in section V. Finally, we conclude this paper in section VI.

## 2. Multimedia Communication Problem

The development here breaks the multimedia communications problem down into the components [2] shown in Figure 1. Components shown there are the sources, the source terminal, the access network, and the destination terminal. This categorization allows us to consider two-way, peer-to-peer communications connections, such as videoconferencing or telephony, as well as asymmetric communications situations, such as broadcasting or video streaming. In Figure 1, the source consists of any one or more of the multimedia sources, and the job of the Source Terminal is to compress the source such that the bit rate delivered to the network connection between the source terminal and the destination terminal is at least approximately appropriate. Other factors may be considered by the source terminal as well. For Example, the source terminal may be battery-power-limited device or may be aware that the destination terminal is limited in signal processing power or display capabilities. Further, the source terminal may packetize the data in a special way to guard against packet loss and aid error concealment at the destination terminal. All such factors impinge on the design of the source terminal. The access network may be reasonably modeled by a single line connection, such as a 28.8 Kbits/s modem, a 56 Kbits/s modem, a 1.5 Mbits/s Asymmetric Digital Subscriber Line (ADSL) line, and so on, or it may actually be a network that shared capacity, and hence have packet loss and delay characteristics in addition to certain rate constraints. The backbone network may consist of a physical circuit switched connection, a dedicated virtual path through a packet-switched network, or a standard best-effort transmission Control Protocol / Internet Protocol (TCP/IP) connection, among other possibilities. Thus, this network has characteristics such as bandwidth, delay, and packet loss. The delivery network may have the same general set of characteristics as the Access network, or one may envision that in a one-to-many transmission that the Delivery Network might be a corporate internet. Finally, the destination Terminal may have varying power, mobility, display, or audio capabilities.

The source compression methods and the network protocols of interest [2] are greatly determined by international standards, and these standards can be adapted to produce the needed connectivity is a challenge. The terminals specified less by standards and more by what users have available now and are likely to have available in the near future. The goal is clear however ubiquitous delivery of multimedia content via seamless network connectivity.

## 3. Computer Networks

We focus in this section on every thing between Source Terminal and the Destination Terminal in Figure 1. Two critical characteristics of networks are transmission rate and transmission reliability. The desire to communicate using multimedia information affects both of these parameters profoundly. Transmission rate must pushed as high as possible, and in the process, transmission may suffer. This becomes even truer as we move toward the full integration of high-speed wireless networks and user mobility. Networking is one of the most interesting disciplines of communication and computer science. It enables individual users who utilize several platforms and share multimedia information with others from their current locations. Users cannot enjoy with this service, if they alter their locations. In the following section, we will introduce a Infrastructure–Based Network that we use one it in our paper.

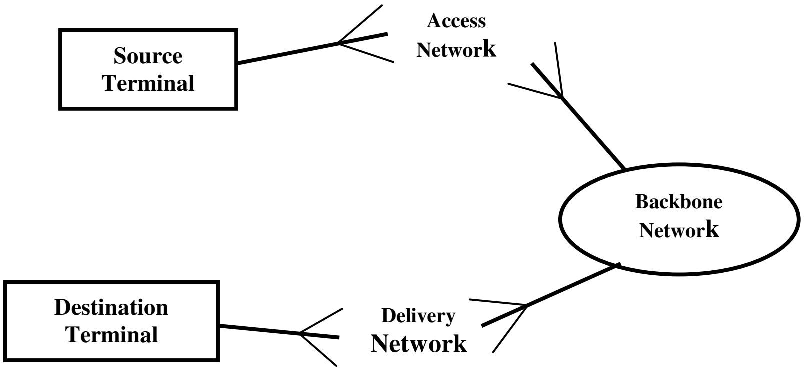


Figure 1. Components of a multimedia communication network

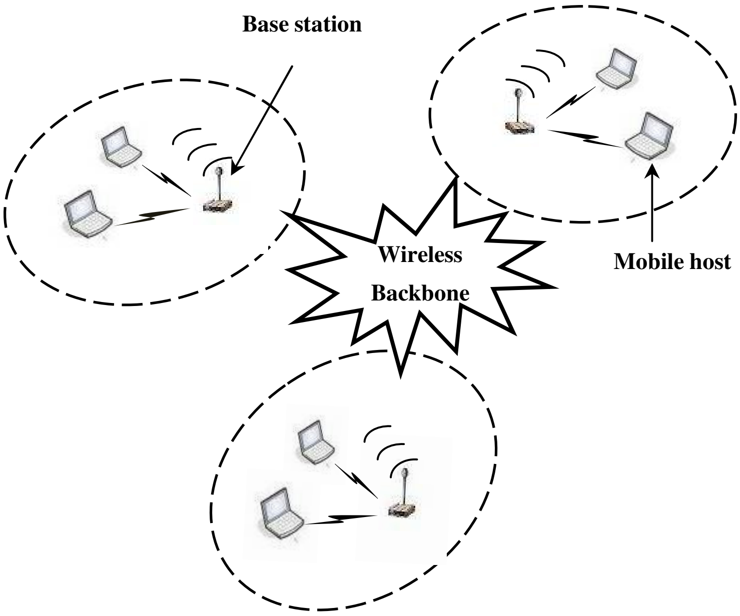


Figure 2. An Infrastructure-Based Networks

### 3.1. Infrastructure – Based Networks [3]

It consists of two parts: fixed wireless part and a dynamic wireless part as illustrated in Figure 2. The fixed wireless part is typically a hierarchy of wide area and local area wireless networks, used as the backbone networks. The wireless backbone connects between special switching wireless nodes called "Base stations". A base station is a computing device equipped with a wireless interface that enables it to connect directly to a mobile host within the coverage area of the base station.

A mobile host within infrastructure networks connects to and communicates with the nearest base station that is within its communication range. A base station uses the fixed wireless network to connect to the other base stations. Hence, base stations act as an interface or gateway (i.e. router or relay) between the dynamic wireless part and the fixed wireless part. As the mobile host travels out the range of one base station and into the range of another base station, a handoff occurs from the old base station to the new base station. Then, the mobile host is able to continue communication seamlessly throughout the network as illustrated in Figure 2. Good examples of infrastructure-based network type are wireless cellular networks and wireless local area networks (WLANs) [3].

## 4. Performance Engineering and Evaluation Methodology

It has been a common practice in many systems (such as Multimedia streams over WLANs) projects to consider performance requirements only at the final stages of the software development process. As a consequence, many systems (Multimedia streams over WLANs) exhibit performance failures that leads to delays and financial losses to companies and users [4]. Performance engineering analyzes the expected performance characteristics of a system during the different phases of its lifecycle. Performance engineering 1) develops practical strategies that help predict the level of performance a system can achieve and 2) provides recommendations to realize the optimal performance level. Both tasks rely on the following activities that form the basis of a methodology.

- Understand the key factors that affect a system's (Multimedia streams over WLAN) performance.
- Measure the system (Multimedia streams over WLAN) and understand its workload.
- Develop and validate a workload model or simulator that captures the key characteristics of the actual workload.
- Use the models or simulators to predict and optimize the performance of the system (Multimedia streams over WLAN).

Performance engineering can be viewed as a collection of methods for the support of the development of performance-oriented systems throughout the entire lifecycle [4]. The phases of the system lifecycle define the workflow, procedures, actions, and techniques that are used by analysts to produce and maintain IT systems (WLAN). Figure 3 provides an overview of a performance-based methodology for system design and analysis. This methodology is based on models that are used to provide QoS assurances to IT systems. These models are: workload model, performance model, availability model, reliability model, and cost model. At the early stages of a project, the information and data available to develop the models are approximate at best.

For instance, in the requirement analysis stage, it is difficult to construct a detailed workload model, specifying the demands placed by transactions on the system's resources. However, as the project evolves, more system information becomes available, reducing the error margin of the model's assumptions and increasing confidence in the estimated predictions of the future system's performance [4]. In summary, the workload and performance models evolve side by side with the system project throughout the stages of its lifecycle. Various types of workload and performance models can be used during the system lifecycle. Basically, they differ in three aspects: complexity, accuracy, and cost. Figure 3 illustrates the spectrum of the models for the different phases, ranked from least to most complex and costly. It is expected that as complexity and cost increase, the refined models becomes more accurate. At the low end of the Figure3.5, one may use back-of-the-envelope calculations as a simple, easy-to-use technique, with a relatively low level of accuracy. At the high end, detailed models of running systems are used that are accurate but, are also costly and complex. Also Figure 4 gives an overview of the main steps of the quantitative approach to analyze the performance of a system.

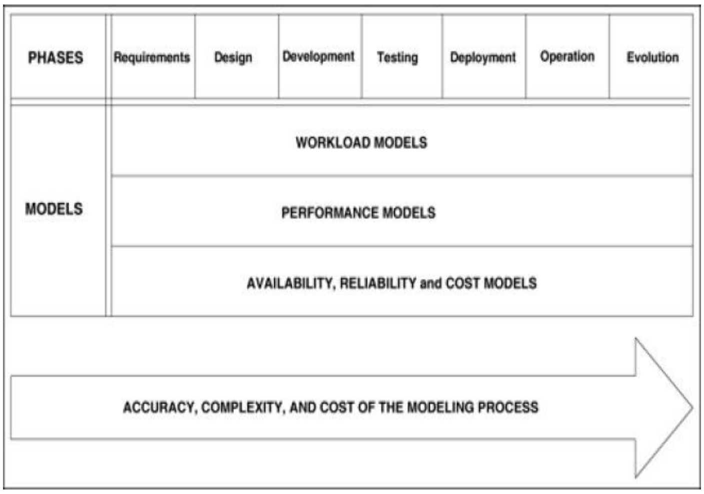


Figure 3. Modeling process

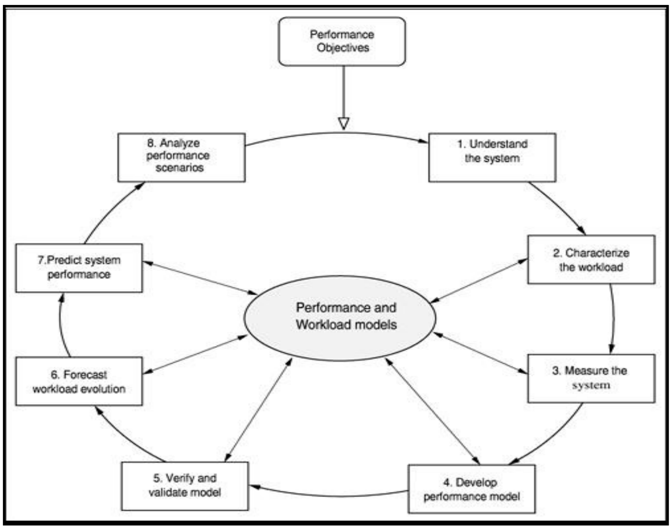


Figure 4. A model-based performance engineering methodology

### 4.1. Performance Evaluation

### A. Simulation Environment and Methodology

To study performance of transmitting Multimedia streams over LANs, OPNET simulator [5] was used. Our simulation models a network of fixed 8 Access Points (AP) and number of workstations varied as 5, 8, and 10 hosts per each AP with in 3000 m x 3000m area. All APs and hosts are connected via one central bridge. Radio propagation range of each host was 450 meters and for each AP was 900 meters. Channel capacity was 2 Mbits/Sec. The IEEE 802.11 Distributed Coordination Function (DCF) [6] is used as the MAC protocol. The Free-Space propagation model [3] is used at the radio layer. In the radio model, we assumed that the radio type was Radio-Capture. Constant Bit Rate (CBR) model is used for data flow and each data packet size is taken as 512 bytes. We use the same parameters for every experiment unless otherwise specified. We have used the following metrics [7] in evaluating performance.

- **End to End Delay**: which is calculated as the average difference between the time each data packet is transmitted by a source entity and the time is received by a receiver entity, and then averaged over the total number of receiver entities.
- **Traffic Received (bits/sec)**: This metric concerns with evaluating the total average of data packets that received by each receiver over all receiver entities per one second.
- **Data Dropped (bits/sec)**: It measures the average of data packets that dropped due to their propagation through network model layers (i.e. overflow of buffers) and due to their fails to reach to their receiver entities (failure of all retransmissions until retry limit). It calculated as the total size of higher layer data packets (in bits/sec) dropped by all the WLAN.
- **Delay (sec)**: it represents the end to end delay of all the packets received by the wireless LAN MACs of all WLAN nodes in the network and forwarded to the higher layer. This delay includes medium access delay at the source MAC, reception of all the fragments individually, and transfers of the frames via AP, if access point functionality is enabled.
- **Load (bits/sec)**: it represents the total load (in bits/sec) submitted to wireless LAN layers by all other higher layers in all WLAN nodes of the network. This statistic does not include the bits of the higher layer packets that are dropped by WLAN MACs upon arrival and not considered for transmission due to a) insufficient space left in the higher layer packet buffer of the MAC, and b) their large sizes that exceed the maximum data size allowed by the standard for WLAN MACs to accept for transmission.
- **Media Access Delay (sec)**: it represents the global statistic for the total of queue and contention delays of data packets received by all WLAN MACs in the network from higher layer. For each packet, the delay is recorded when the packet is sent to the physical layer for the first time. Hence, it also includes the period for the successful RTS/CTS exchange, if this exchange is used for that packet.
- **Throughput (bits/sec)**: it is calculated as the number of data bytes delivered to receiver entities per each second. It is important to figure out the total efficiency of over all WLAN environments.

### B. Simulation Results

#### B.1 Scenarios

In these simulation runs we implement three different scenarios with different number of hosts per each AP. Firstly, we use 5 hosts per each AP. OPNET of 5 hosts per each AP is shown in Figure 5. Secondly, 8 hosts per each AP are implemented as shown in Figure 6 and at last simulation run, 10 hosts are used as in Figure 7. In the three scenarios 8 APs are used and only one central bridge is used to connect them. The purpose of these runs is to measure different metrics that stated at **section V.A**, and evaluate the overall network performance especially network efficiency and reliability [8] (i.e. how the network operate at heavy circumstances) and scalability (i.e. how the network can operate when the number of hosts increases).

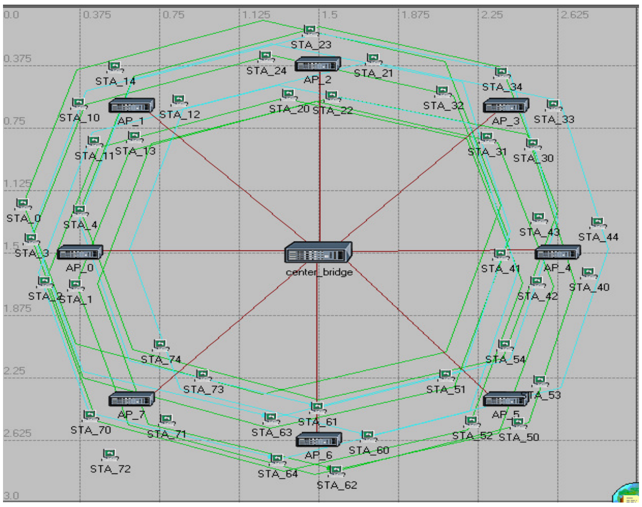

Figure 5. OPNET implementation of 5 hosts per each AP

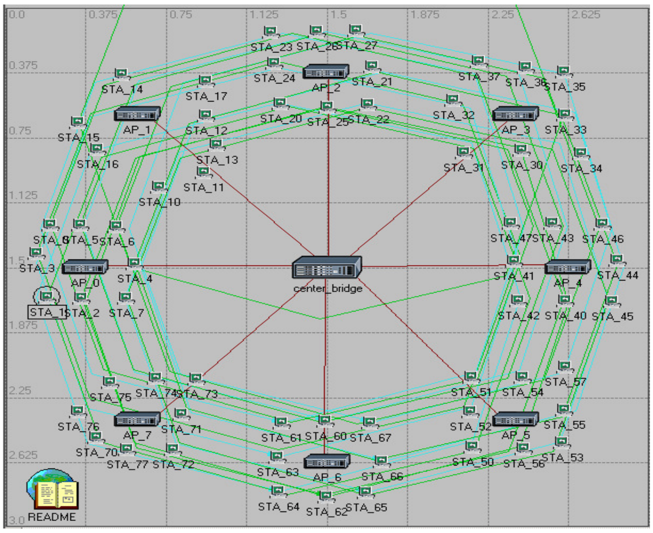

Figure 6. OPNET implementation of 8 hosts per each AP

### B.2 Results and Analysis

In the following sections, metric results that taken from OPNET simulator will be figured out. Discussion about that results will be stated to show how increasing in number of users will affect on all the network performance (efficiency and scalability issues).

### B.2.1 End to End Delay Results

**Figure 8** shows OPNET results for the three previous scenarios. **Figure 8** shows the ratio of average End to End Delay as a function of simulation run time that used in our experiments. First scenario (5 hosts per each AP) achieves the lowest End to End Delay. It is a natural situation because small number of hosts that applied per each AP and because small numbers of hosts leads to a small propagation of control and data packets and then reduces collisions, congestions, and buffer overflow problems. So End to End Delay, between sending data from a source entity until reach to its receiver entity, will be reduced. On the other side, when number of hosts per each AP increases (e.g. 8 hosts per AP and 10 hosts per AP), it increases End to End Delay of delivering data packets to their receivers. The previous situation can come up due to increase of possibility of collisions, congestions, and buffer overflow occurrences. The previous End to End Delay can be traced back to limited bandwidth that generally found in WLAN. It increases interference of radio waves and then increasing possibility of losing data packets. It leads to enforce TCP protocol to resend these data packets and leads to send more control packets to build tunnels between sources and receivers.

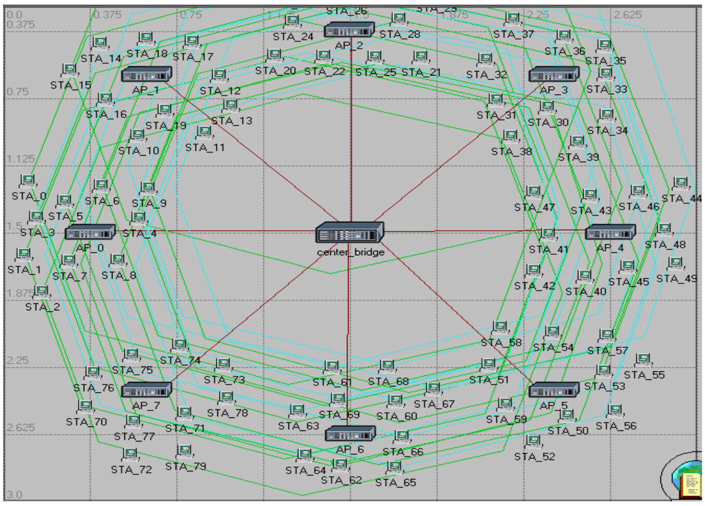

Figure 7. OPNET implementation of 10 hosts per each AP

All the previous limitations lead to a huge number of data and control packets that propagate over the entire wireless network. In Figure 9, we averaged results of each scenario as alone. It shows average End to End Delay as a function of number of users per each AP. It shows that increasing number of users per each AP will yield to monotonically increasing in End to End Delay.

### B.2.2 Traffic Received Results

The average of data packets that received by each receiver per a second as a function simulation run time is shown in Figure 10. Each scenario indicates that average value of data packets that received by each receiver tolerate slightly around a certain value. For example, when hosts are five, its averaged traffic received ranged around 35 packets/second. At 8 hosts per AP the previous value is ranged around 55 packets/second. At 10 hosts per AP, traffic received value is tolerated around 72 packets/second. It is indicated that when number of user's increases, it leads to increase of receivers and sources, so more data packets will be reached to their receivers.

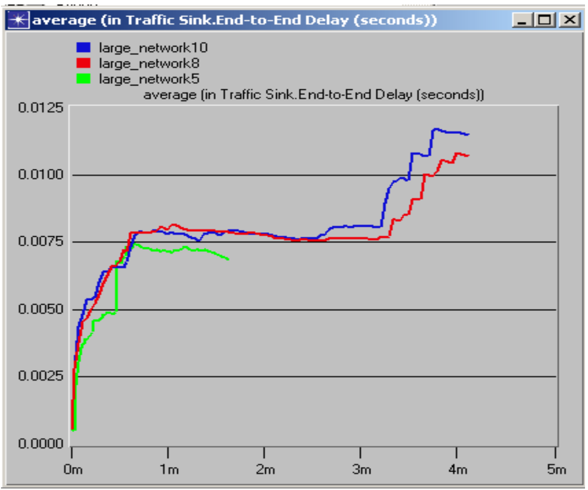


Figure 8. End to End Delay as a function of simulation run time

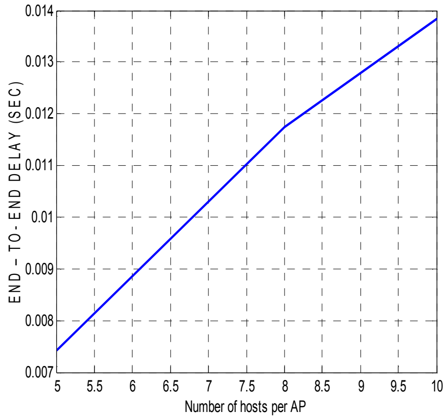


Figure 9. End to End Delay as a function of hosts per AP

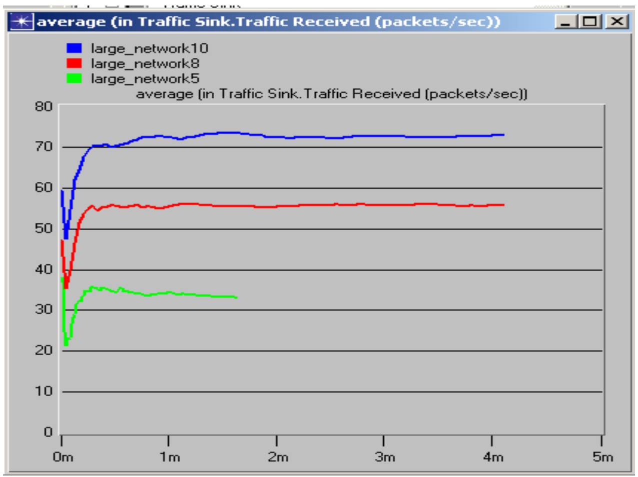


Figure 10. Traffic Received as a function of simulation run time

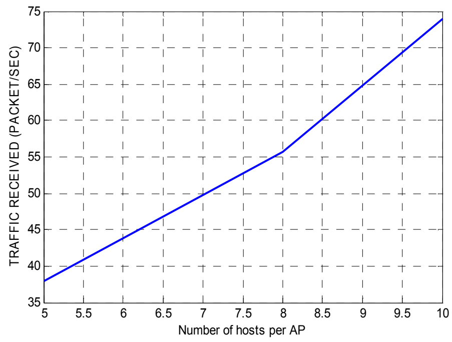


Figure 11. Traffic Received as a function of hosts per AP

Figure 11 shows the average traffic received by each receiver as a function of number of users per each AP. It is evident that average value of data packets reached to each receiver increases as the number of hosts increases.

### B.2.3 Data Dropped Results

Data Dropped metric is so important when evaluating a WLAN system. It shows how a WLAN can operate when number of users increases (scalability issue). Figure 12 shows the relation between percentage of data dropped per each second and the simulation run time. When data dropped percentage is increased, it can be considered as a serious mark that the WLAN system's function is affected negatively by increasing of number of users. Firstly at 5 hosts per AP, this percentage is low compared to other scenarios (8 hosts per AP and 10 hosts per AP). There are many reasons that cause that degradation in WLAN system. More users lead to more sending of control and data packets. So the limitations that stated in End to End Delay and others as continuous movement of users and centralization mechanism that exist due to use of APs and central bridge.

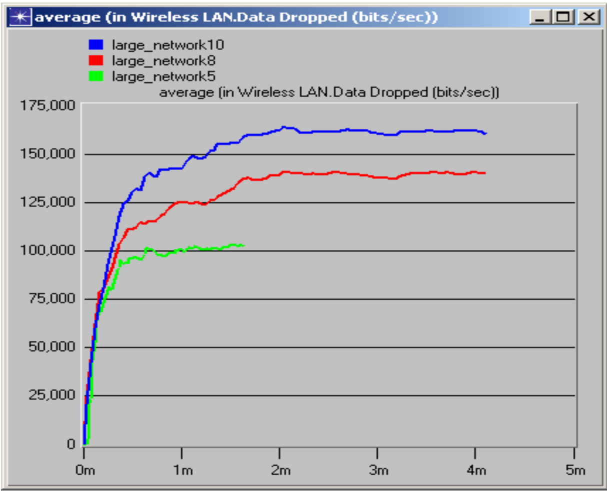


Figure 12. Data Dropped as a function of simulation run time

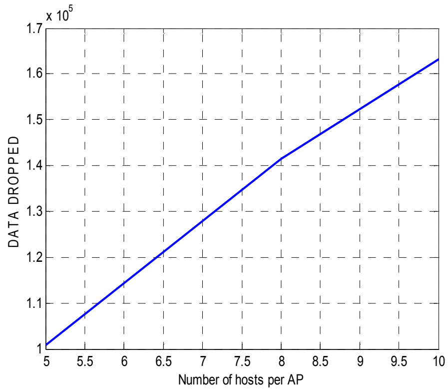


Figure 13. Data Dropped as a function of hosts per AP

Centralization mechanism means that AP is the only one that receives all packets (control or data) from its users and resends them to their receivers. It acts as a central gateway (router). This means that if this AP fails, the connection between users is lost. AP suffers from excessive number of packets that go in and out from it which lead to traffic jam around it. This excessive traffic jam leads to ease of data packets that dropped. The previous reasons cause that percentage of data dropped increases as number of users increase. Figure 13 figures out the data dropped percentage as a function of number of users. It confirms the previous described results.

### B.2.4 Delay Results

It is a metric that used to determine delay between MAC layers. It concerns with measuring the delay when data frames are divided into packets and then sent until they reach to their destinations and then assembled at their destination MACs. Figure 4.10 shows average delay as a function of simulation run time. In Figure 14, 5 hosts per AP achieve the lowest delay. That low delay is found due to using small number of users in over the entire network. Network performance is reduced due to increasing number of users because delay becomes larger. It is evident from Figure 14 when applying 8 hosts per AP and 10 hosts per AP. MAC concern with dividing and representing any data frames that come from upper layers into packets with a specific sequence numbers. It also responsible about assembling that received packets according to their sequence number. When number of users increase in the network, it increases number of data packets that sent over the entire network. It causes many data packets to be dropped and then receivers' MAC wait until all data packets received completely. So the delay metric become larger as users increases. When we apply simulation run time 1 and 2 minutes, we found that all scenarios increase slightly. It can be traced back to the fact that network is still can deal with large number of data packets. But when applying 3 and 4 minutes, curves of both 8 hosts and 10 hosts per AP increases sharply indicating that number of data packets in network become so large and then possibility of data dropped occurrence become high and then it increases delay until all packets reach to their destinations' MAC. Figure 15 shows the relation between delay and number of users. If Figure 14 and 15 are compared, they confirm that when number of users increases, it increases delay.

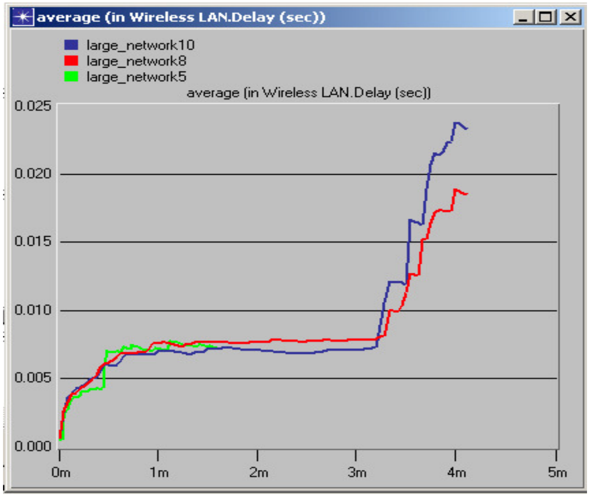


Figure 14. Delay as a function of simulation run time

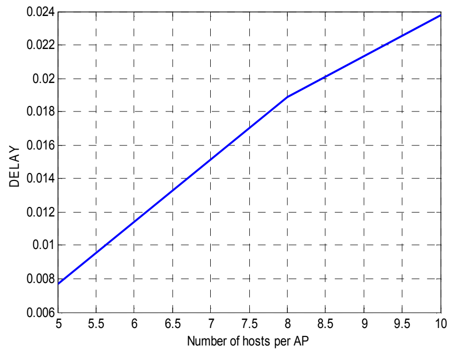


Figure 15. Delay as a function of hosts per AP

### B.2.5 Load Results

To study the impact of data load (i.e. the number of data packets that transmitted at each second), OPNET figures out the previous scenarios in Figure 16. Figure 16 depicts the average load over the entire network as a function of simulation run time. Generally, more number of users leads to more data packets transmitted over the entire network which increases network load. As shown in Figure 16, load of 5 hosts increases until reach to 750,000 data packets/sec. at 8 hosts per AP, it reaches to 1,200,000 packets/sec while it reaches to 1,500,000 packets/sec at 10 hosts per AP. It is confirmed before that increasing in data packets over the entire network, will affect negatively in efficiency and scalability issues over the entire network. It leads to problems of buffer overflow, collisions, congestion, ease of data dropped, and etc. Figure 17, which shows the relation between load and number of users, makes another confirmation for the previous results.

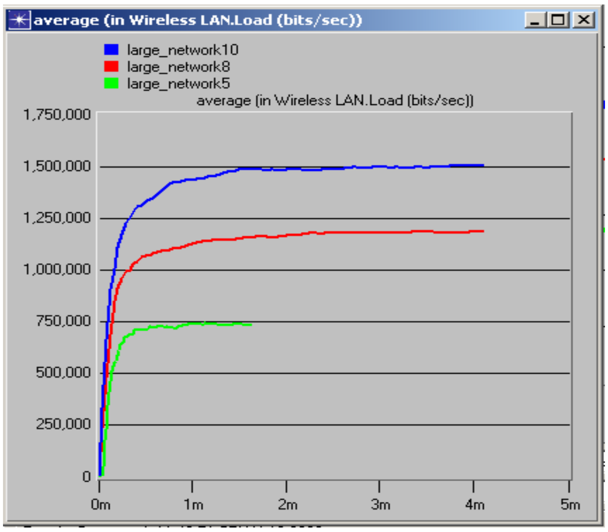


Figure 16. Load as a function of simulation run time

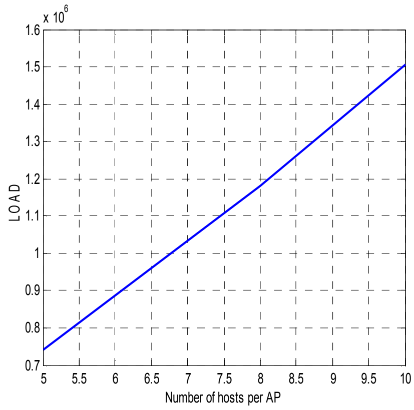


Figure 17. Load as a function of hosts per AP

### B.2.6 Media Access Delay Results

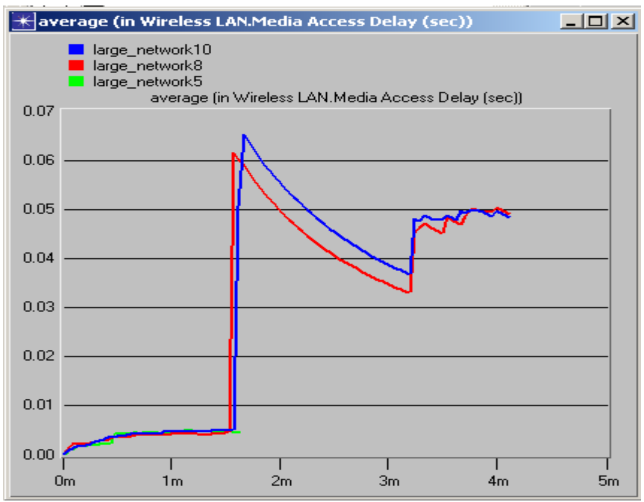


Figure 18. Media Access Delay as a function of simulation run time

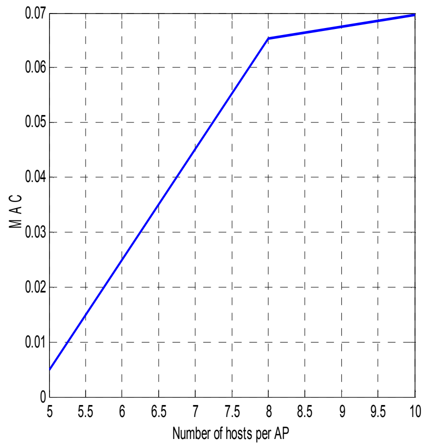


Figure 19. Media Access Delay as a function of hosts per AP

The main characteristic of WLAN is contention based property of its medium access. It means that when any source needs to send its data packets, it firstly listens to the medium. It cannot send data packets until be sure that one medium channels are free, so each source fight to take a free channel and send its data through that medium channel. When a source cannot find a free channel, it will wait until can reserve one. The previous situation increase what is called "medium access delay"? Medium access delay becomes worst when the number of users over the entire network increases as shown in Figure 18. Increasing of users will increase number of sources that can wait for a free medium channel. Also limited bandwidth of WLAN makes a huge of medium access delay problem. For example in Figure 18, 10 hosts per AP suffers from high medium access delay when it compared to 8 hosts per AP and 5 hosts per AP. 5 hosts per AP achieves the lowest percentage of that delay. Figure 19 depicts the average media access delay as a function of number of users. From that figure we can conclude that any increase of number of users cause an increase in that delay and then increase degradation of all the network performance.

### B.2.7 Throughput Results

This metric is the most interesting issue when measuring and evaluating any network especially WLAN. It tries to conclude how all devices and protocols that applied in WLAN act efficiently to deliver the desired data packets to their receivers. Throughput concerns with the actual data packets that received by all receivers per one second. Figure 20 depicts throughput as a function of simulation run time. In Figure 20, 5 hosts per AP scenario achieves nearly 600,000 total actual data packet delivery. 8 hosts per AP scenario (i.e. increase number of users by 3 per each AP) reaches to 950,000 delivered data packets while at 10 hosts per AP scenario, data packet delivery achieves 1,200,000 packets. It is evident that as number of users increases, also number of receivers increases which lead to the fact that throughput increases. Figure 21 shows the relation between average throughput as a function number of hosts per each AP. The Figure shows that increasing in number of users will yield to monotonically increasing in throughput.

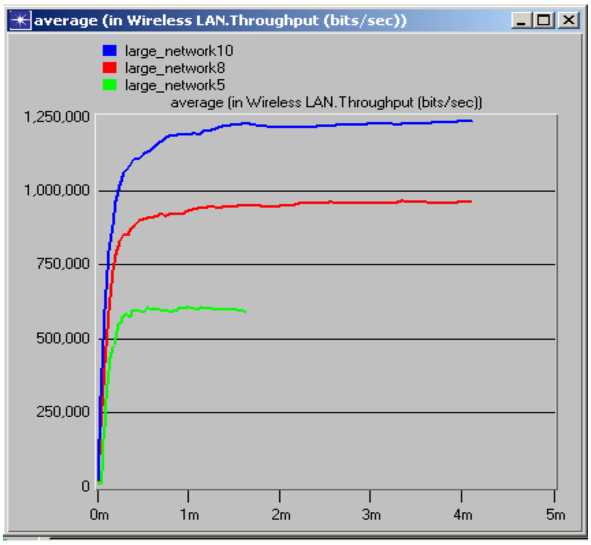


Figure 20. Throughput as a function of simulation run time

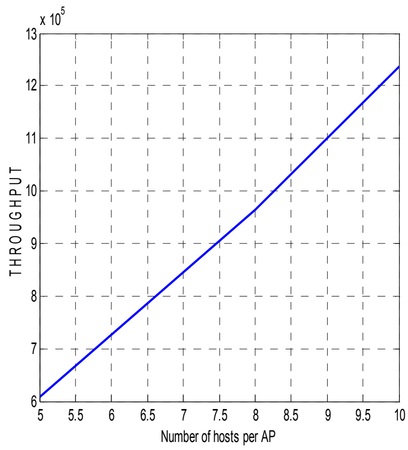


Figure 21. Throughput as a function of hosts per AP

## 5. Conclusion

In this paper, systematic steps are followed to evaluate the performance of WLAN environment by exchanging multimedia streams data over WLAN devices especially when number of hosts over the entire network is increased. Three scenarios of WLAN system are specified (5, 8, and 10 hosts per each AP). The metric that we used in the evaluation are determined also like End-to-End delay, traffic received, data dropped, delay, load, media access delay, and throughput. And the goals of this paper are to monitor the scalability and efficiency issues of the entire WLAN system when number of hosts increased.

Simulation results are conducted and analyzed to sum up the performance of WLAN system. Simulation results show that increasing in number of hosts per each access point, will cause increase in End to End delay by 42% at 8 hosts scenario and by 64% at 10 hosts scenario over 5 hosts scenario. Also Traffic Received increases by 18% and 36% at 8 hosts and 10 hosts respectively. Data dropped increases by 37% and 58% but load has increasing by 43% and 75% at each 8 and 10 hosts. Media access delay has a percentage 6% and 6.5% which increased over 5 hosts scenario. Finally, throughput increases by 38% and 65% in each 8 and 10 hosts than 5 hosts per each access point. Simulation results sum up that when number of users increases, a degradation of network performance will occur. This degradation can be traced back to huge number of data and control packets that propagate over the entire network. Huge number of packets causes occurrence of collisions, buffer overflow, congestion, and ease of dropping data packets which affect negatively on network performance.

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