ENSC 835: COMMUNICATION NETWORKS
CMPT 885: SPECIAL TOPICS: COMMUNICATION NETWORKS

DEPLOYMENT OF MOBILE AND FIXED VIDEOCONFERENCING OVER AN EXISTING IP INFRASTRUCTURE

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FINAL PROJECT

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1. ABSTRACT

Over the last decade there has been a significant increase in demand for delivery of interactive multimedia content to the end user from voice to video. At the same time, the growth of popularity of wireless access technologies has prepared the consumer to expect an increase in bandwidth and the quality of user experience over wireless links every year. As a result, with the popular acceptance of personal wireless devices in the business environment and everyday life, and the necessity to communicate, there is a strong need for reliable interactive content transmission. This project concentrates on simulation of the deployment of videoconferencing over a typical office network that includes both wired and wireless links using OPNET. Two wireless LAN standards are tested—802.11b and 802.11g. The maximum number of video and voice calls is determined, showing the capacity of the whole network. In addition, separate tests are applied directly to wireless clients, measuring the limitations of each the wireless standards. Metrics as wireless load, throughput and drop rate, bit and packet rate, as well as router queue size are used to analyze the behavior of the system.

2. INTRODUCTION

2.1. Videoconferencing Background

From the first satellite network that came to life in 1970 in Hawaii (ALOHA) to the 21st century, the history of the Internet has shown an explosive growth and popularity. With the Internet's traffic increasing every year and its infrastructure expanding, the demand for better and faster network connection keeps on rising. So does the demand for services such as interactive video and voice communications. When one combines two of these, it is called videoconferencing (also known as videoteleconferencing).

Videoconferencing has gained its popularity throughout 1990s, when more efficient compression algorithms have been developed and the IP based videoconferencing became possible. As a result, that made the technology available to the general public at a reasonable price. In 2008 one can find a variety of commercial and free-for-all open source applications to install on a personal computer or a set of computers belonging to an enterprise.

2.2. Motivation

Videoconferencing as a service and idea has a number of benefits in our society—from personal to commercial. As a personal service the advantages are obvious—looking at a caller during the conversation allows conveying much more useful information like face expressions. These can not only be fun, but also allow a better communication experience, especially for people who are separated by large distances.

In the business world, frequent travel between sites can be cut down if the quality of technology achieves the goals of meetings by creating natural video and voice links. As a result that saves money and time by decreasing the operational cost of a company.

As well, in office or industrial environments with an already existing deployed network across the premises, the addition of point-to-point videoconferencing support can be accomplished with minimal costs. In this case, there is an interest in simulating the network with a new type of traffic before investing time and money into hardware and software.

OPNET simulator has been chosen for this project because it is a popular industry-proven tool. Version 12.0 provides interactive voice and videoconferencing modules that allow implementing the simulation without model and code modifications.

Even though there is a number of papers dedicated to the study of videoconferencing support over wireless links, at the time of this project, no existing OPNET models have been identified; thus, creating further motivation for the project.

2.3. Scope

The application of videoconferencing has proven itself useful in various scenarios—from office and industrial environments connecting devices via wired and/or wireless links to personal communications via portable devices. In this paper we study the deployment of videoconferencing application over the currently existing Wireless LAN and LAN infrastructure. A simple yet typical case of a two floors business is simulated.

A typical scenario can include one plant/warehouse floor with wireless clients and an office floor with wired desktops. Such an application can be found in industrial plants where the communication between factory personnel is essential. Example applications are:

3. Plant maintenance (i.e. chemical, power, etc)—personnel with PDAs reporting to office
4. Production environment—warehouse, equipment maintenance with personnel reporting back to office
5. Office with two floors and a mix of wired and wireless clients

The main interest of the project is to determine the capacity of the network that contains both wireless (802.11b/g) and wired links, and to support videoconferencing calls. This is done by observing the traffic and the mismatch between sent and received videoconferencing packets at the wireless router and single clients. Throughput, load, drop rate and queue sizes at wireless router are used to determine the health and limits of the wireless LAN, which prove to match the typical values reported by other studies.

This paper does not take into account the background wireless traffic with the assumption that wireless devices are strictly used for videoconferencing. The wired network itself has a typical background traffic resulting from ftp, http, data base access, web cache and NAT.

OPNET simulator is used to measure the total videoconferencing traffic exchanged between wired and wireless clients by combining following:

- Fixed and mobile workstations
- Two fixed workstations
- Two mobile workstations

IT Guru Academic Edition is used to verify the capacity of strictly wireless LAN by limiting the number of wired workstations to one; thus, eliminating the first two cases above.

Default OPNET's wireless channel represented by a 14 stage pipeline has been used. Thus, the free space propagation model has been utilized.

2.4. Related Work

Our project reuses the IP infrastructure presented [1]. The OPNET model that was used to verify the analytical results of [1] has been provided by K. Salah via email and modified to meet the purposes of our study. The main interest in K. Salah's work for this project was in the model that demonstrated a proven way to measure video and voice traffic over wired links. Voice and video coding parameters, as well as packet sizes and typical IP traffic have been suggested.

The similar simulations have been performed to measure the capacity of IP network with VoIP application deployed [2]. Figure 1 shows a typical network topology with necessary network components. The study performs a thorough analysis of the network conditions with VoIP traffic and a typical background traffic originating from the Internet browsing, e-mails, file transfer and database access in a small enterprise office consisting of 3 floors. A method to determine the capacity of the network to support voice calls under these conditions is presented. Simulation optimization techniques and assumptions that allow to save time are presented.

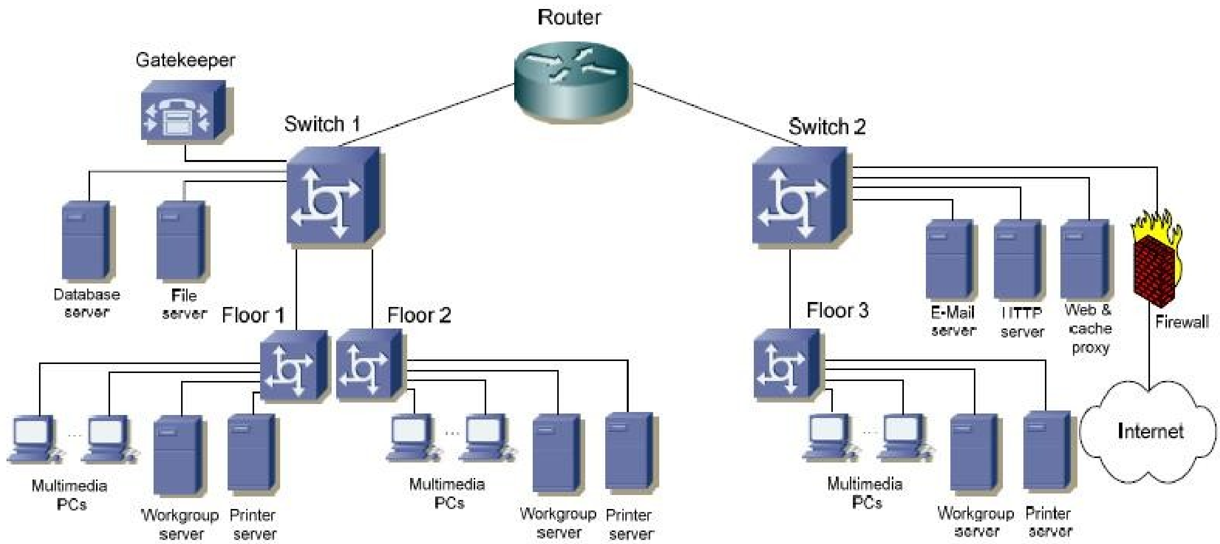


Figure 1: Network topology with necessary network components [2]

In [3], an ad-hoc 802.11b testbed is used to experimentally measure the maximum number of wireless clients receiving 400 kbps video and 128 kbps audio streams simultaneously. It was demonstrated that under these conditions, up to 8 clients can be supported. 9th client overloads the wireless LAN and decreases the performance of other clients. The similar performance measurements have been done for 802.11g in [4] to support multi-player games that combine audio and video traffic and [5] measuring throughput of UDP traffic over 802.11g wireless LAN.

The authors of [6] emphasize the differences between unicast and multicast video streaming and propose near optimal coding solutions for wireless LANs, specifically 802.11b. Wired and wireless channels behave very differently from each other and need a special care when video streaming is considered. An important feature specific to WLANs is in the way ACKs are sent. UDP unicast frames are sent asynchronously in order to hide the lossy wireless environment from the transport layer, while multicast frames are not acknowledged to minimize the ACK explosion due to poor link conditions. The study explains the modeling of WLANs based on Packet-Erasure Model to abstract away the physical and link layers from the the rest of the system. In our project, we consider point-to-point videoconferencing connections. Algorithms presented in [6] can optimize the video stream; however, have not been considered in this paper.

3. Model

3.1. Modeling the Wireless Network

In our study, we use the similar network infrastructure as presented in [1]. There are two floors. One is representing an office with wired desktops (floor 2) and another one—a plant's floor with a wireless LAN (floor 1). Clients of floor 1 are connected via a wireless router to the Floor 1 Switch that in turn is connected to the main router via Switch 1 as in Figure 2. Floor 2 has wired desktops that are represented by two LAN workstations—for videoconferencing and background traffic generation respectively. Background traffic is described in the following section. Floor 2 Switch is connected to the main router via Switch 2. The network contains a number of servers running typical enterprise applications such as FTP, database, mail, web, web cache and NAT.

Figure 2 shows the OPNET model used for our project.

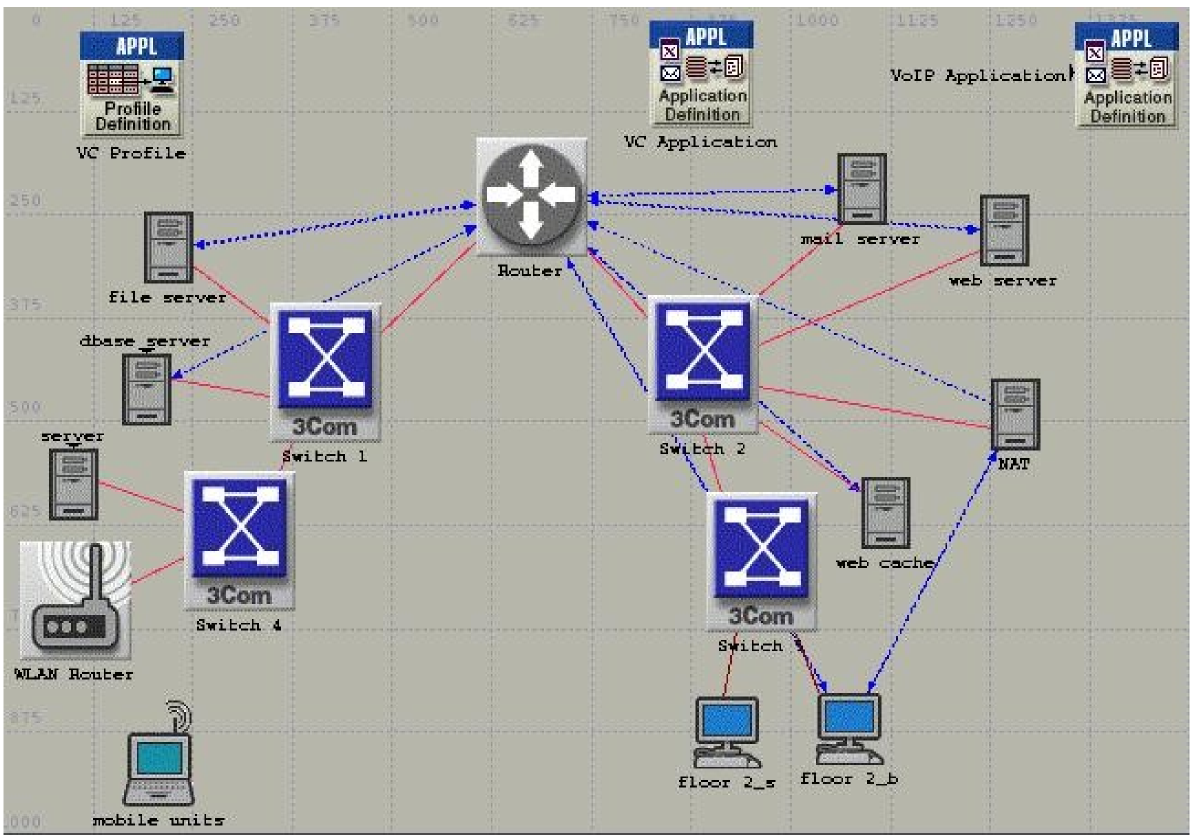


Figure 2: OPNET model of the organization of the network with wireless terminals

3.2. Application and Profile

There are two types of applications used by our model. The first one is videoconferencing (VC Application) that uses the following parameters:

- Type of service: Interactive Multimedia
- Frame Interarrival Time Information: 30 frames/sec

- Frame Size Information: variable, loaded from script
- Symbolic Destination Name: VC_dest
- The other settings are left default

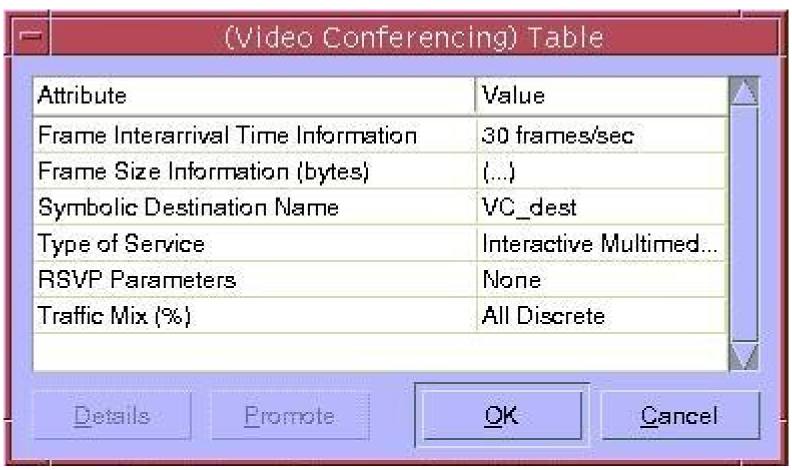
The size of a video packet has a variable length. A file with 42972 packet size entries ranging from 65 to 1518 bytes and collected empirically as specified in [1] is used for simulation. [1] is using frames with the size of 1344 bytes based on statistical analysis of the distribution for video bit rate calculations. Please refer to the Generating Traffic section below.

The second application (VoIP Application) configures voice as follows:

- Encoder Scheme: G.711
- Voice Frames per Packet: 5
- Symbolic Destination Name: VoIP_Dest
- Type of Service: Best Effort

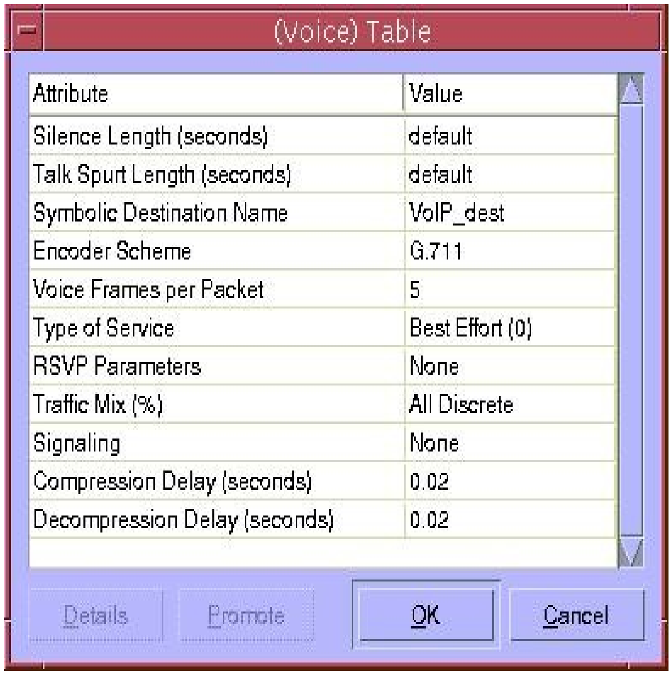
OPNET's default configuration for voice application uses 32 8-bit samples per second, while G.711's standard specifies 160 Bytes packets. Thus a factor of 160/32 = 5 is required.

Figures 3 and 4 present the configuration tables for videoconferencing and voice respectively.



Attribute	Value
Frame Interarrival Time Information	30 frames/sec
Frame Size Information (bytes)	(...)
Symbolic Destination Name	VC_dest
Type of Service	Interactive Multimed...
RSVP Parameters	None
Traffic Mix (%)	All Discrete

Figure 3: Video Conferencing Application table



(Voice) Table

Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	VoIP_dest
Encoder Scheme	G.711
Voice Frames per Packet	5
Type of Service	Best Effort (0)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02

Details Promote OK Cancel

Figure 4: Voice Application table

The main purpose of our project is to determine the capacity of the network to support videoconferencing calls that consist of both video and voice calls. We achieve this by slowly increasing the number of calls while measuring sent, received and dropped traffic at the wireless router. When the capacity of the wireless link is reached there is a mismatch between the number of sent and received packets. Hence, the profile that we are using includes both of the applications and is configured to start every 2 seconds. This allows us to use a single wireless and single desktop workstation models of OPNET, while imitating a dynamically increasing number of workstations.

The videoconferencing profile (VC Profile) runs VC and VoIP applications and is shown in Figure 5. The following parameters have been used:

- Profile Name: VC_user and VoIP_user
- Start Offset: 10 sec
- Repeatability: 2 sec
- Number of Repetitions: Unlimited
- Repetition Pattern: Concurrent
- Operation Mode: Simultaneous
- Start Time: 60 sec

As a result, our profile starts at 70 seconds from the start of simulation. The new profile is loaded every 2 seconds, creating 2 video and 2 voice calls. Note that by default, the background IP traffic flow starts at 40 seconds, allowing 30 seconds gap before videoconferencing traffic. The offset of 10 seconds has been chosen for debugging reasons and kept through the project.

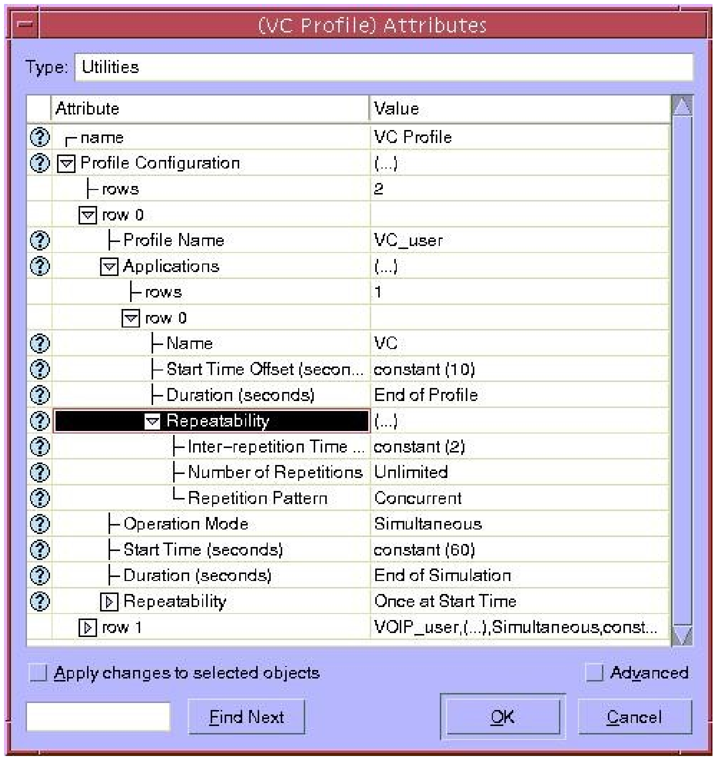


Figure 5: Videoconferencing profile

3.3. Generating Traffic

There are three types of traffic flowing in the network: IP flow, video and voice. Table 1 shows the IP flow traffic between network elements, the configured distributions, intervals, and traffic defined in bits/sec and packets/sec as suggested by [1]. Note that the packet sizes exchanged between different types of services is kept constant as seen from the last column.

Server	Type	Interval (s)	bits/sec	packets/sec	bits/packet	Bytes/pkt
File server->Router	Uniform	0-86400	1900000	158.3	12002.53	1500.32
Router->File server	Uniform	0-86400	1900000	158.3	12002.53	1500.32
Db server->Router	Uniform	0-86400	2300000	191.7	11997.91	1499.74
Mail server->Router	Uniform	0-86400	2300000	191.7	11997.91	1499.74
Web server->Router	Uniform	0-86400	2200000	183.3	12002.18	1500.27
NAT->Router	Uniform	0-86400	2300000	191.7	11997.91	1499.74
Web cache->Router	Uniform	0-86400	2600000	216.7	11998.15	1499.77
Floor 2b->Router	Uniform	0-86400	2400000	200	12000	1500
Floor 2b->NAT	Uniform	0-86400	1200000	100	12000	1500
NAT->Floor 2b	Uniform	0-86400	1200000	100	12000	1500

Table 1: IP Flow Traffic

As mentioned in the application configuration section, the frame rate for video traffic is chosen to be 30 fps and the packet size 1344 bytes. That creates a 320 kbps video stream each way (i.e. 30 pps) or 640 kbps both ways (i.e. 60 pps).

The bit rate for voice, as specified by G.711 is 90.4 kbps. The IP overhead of 66 bytes/voice_packet. At 30 fps, that creates 15,840 bps to the total of 106.2 kbps for voice packets (i.e. 90.4 kbps + 15,840 bps) each way.

4. Simulation Study

4.1. Scenarios

The are two main scenarios that are used to simulate videoconferencing application in 802.11b and 802.11g wireless LANs respectively using OPNET. One additional scenario is used to verify the results of 802.11b using IT Guru Academic Edition. The differences between different wireless physical layers are emphasized and the results are compared to other sources. Table 2 below summarizes the scenarios, physical layers and the specific names used in their respective models. For the verification methodology used in Scenario 3, please refer to the Verification section.

OPNET Simulations

Scenario	Physical Layer	Scenario Name
1	802.11b	2_wlan_80211b_bg_2_flr
2	802.11g	3_wlan_80211g_bg_2_flr1

IT Guru Simulations

Scenario	Title	Scenario Name
3	802.11b	7_H323_Cloud_8usr1

Table 2: Scenarios

4.2. Simulation Runs

Each scenario has been run multiple times to make sure the results are consistent. The longest simulation run was for scenario 2 with the peak time of 1 hour, 58 minutes for 4 minutes of real time.

5. Simulation Results

5.1. Total Statistics at Wireless Router

Figure 6 and Figure 7 show the traffic sent and received (in pps and bps respectively) by all clients within the network that are using videoconferencing profile (I.e. wired and wireless clients on both floors). The statistics includes all of the calls established between (1) wired and wireless clients (I.e. video call from the plant to the office), (2) only wired calls (I.e. calls established between the office staff, and (3) only wireless calls that are made between the plant's personnel.

The video conferencing traffic starts at 70 seconds as expected. Every two seconds 2 new video and 2 new voice calls are established. For example, at 72 seconds, there are 4 video calls or 60*4 = 240 pps as estimated earlier. At 1 minute 38 seconds mark, the mismatch between the transmitted and received number of packets is observed. Two seconds before that the network was still performing well. This is the point at which the capacity of the network under giving conditions has been reached. To estimate the number of calls we can use the following simple calculation: 2calls + 2calls*((1m * 60s/m + 36s – 70s)/2s) = 28 videoconferencing calls.

Figure 8 represents the load, throughput and dropped data at a wireless router in bits per second. Similar to the previous figures, the load and throughput start increasing gradually from 70 seconds until the capacity of the network is reached. At 1 minute 38 the first mismatch is observed between the traffic submitted by the application layer to the MAC layer (I.e. load) and the traffic received by the MAC layer (I.e. throughput). Throughput reaches 4 Mbps and stays constant, which is the normal average throughput for 11 Mbps 802.11b WLAN. Our findings agree the experimental results reported in [3].

The size of the queue at the wireless router is shown in Figure 9. As the capacity of the physical layer is reached, the packets are first stored in the buffer before being dropped. It is obvious from the figure that the buffer size starts increasing at 1 minute and 38 until the overflow occurs.

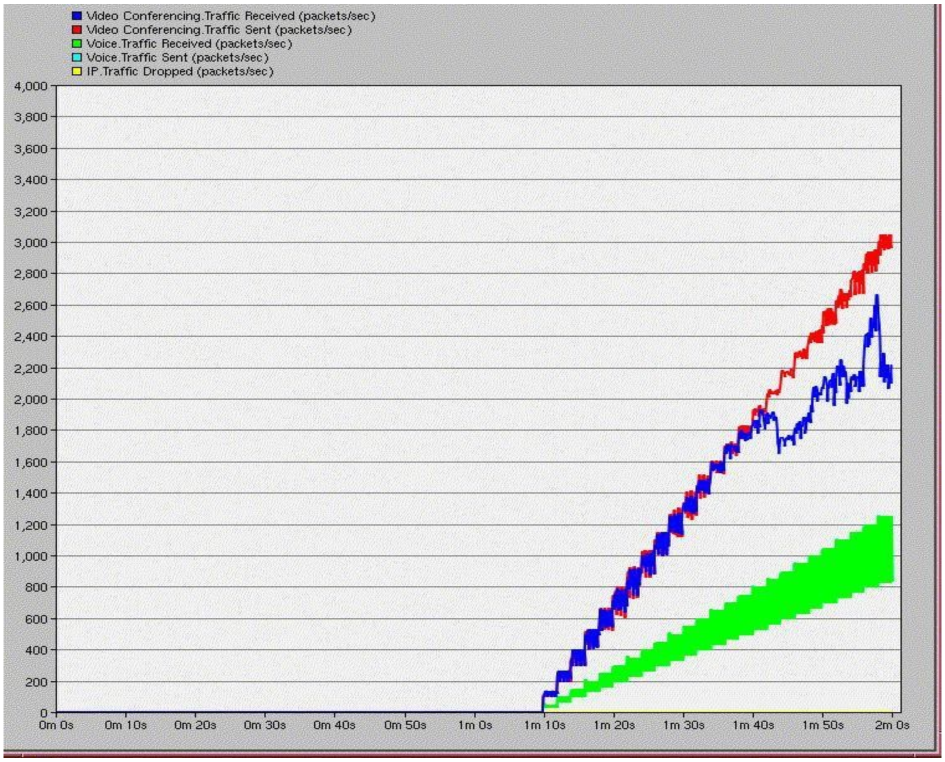


Figure 6: 802.11b Videoconferencing traffic (packets/sec)

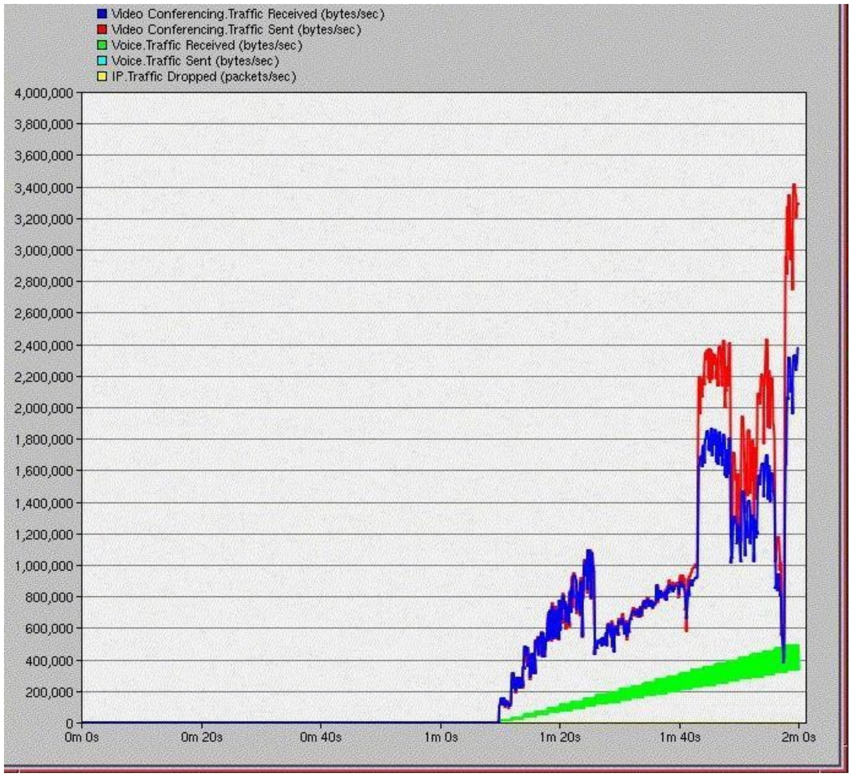


Figure 7: 802.11b Videoconferencing traffic (bits/sec)

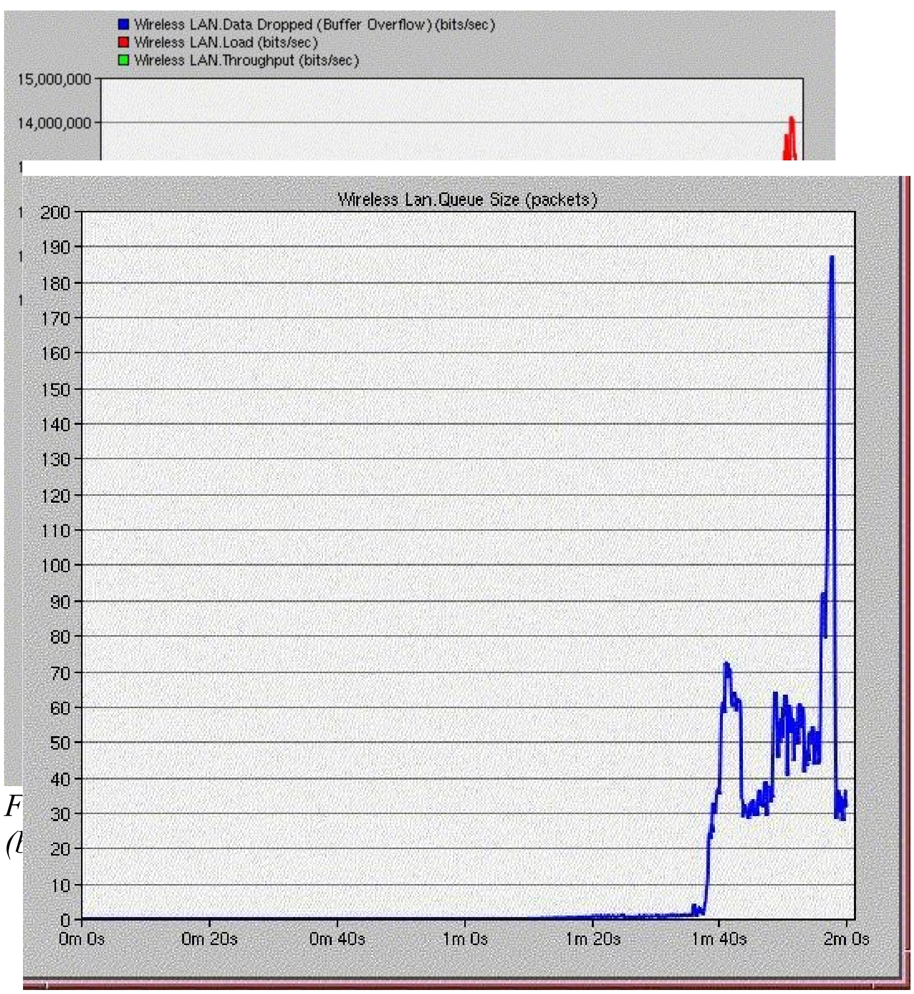


Figure 9: 802.11b WLAN's Queue Size (packets)

Similar results are observed for Scenario 2 (802.11g). The videoconferencing profile is loaded at 70 seconds from the beginning of the simulation. Every 2 seconds two calls are established (i.e. one wireless and one wired). This time, as observed from the traffic sent and received in packets per second and bits per second in figures 10 and 12 respectively, the mismatch occurs at 2 minutes 48 seconds. It takes a loner time for 802.11g to reach its capacity as expected due to the higher throughput (54 Mbps theoretical). Figure 13 shows that the average throughput of 802.11g of 25 Mbps is reached.

The maximum number of videoconferencing calls for scenario 2 is 98 using the similar approach as in scenario 1 (i.e. 2calls + 2calls*((2m * 60s/m + 46s – 70s)/2s) = 98)

Figure 11 shows the queue size in packets which consistently increases starting from 2 minutes 48 seconds.

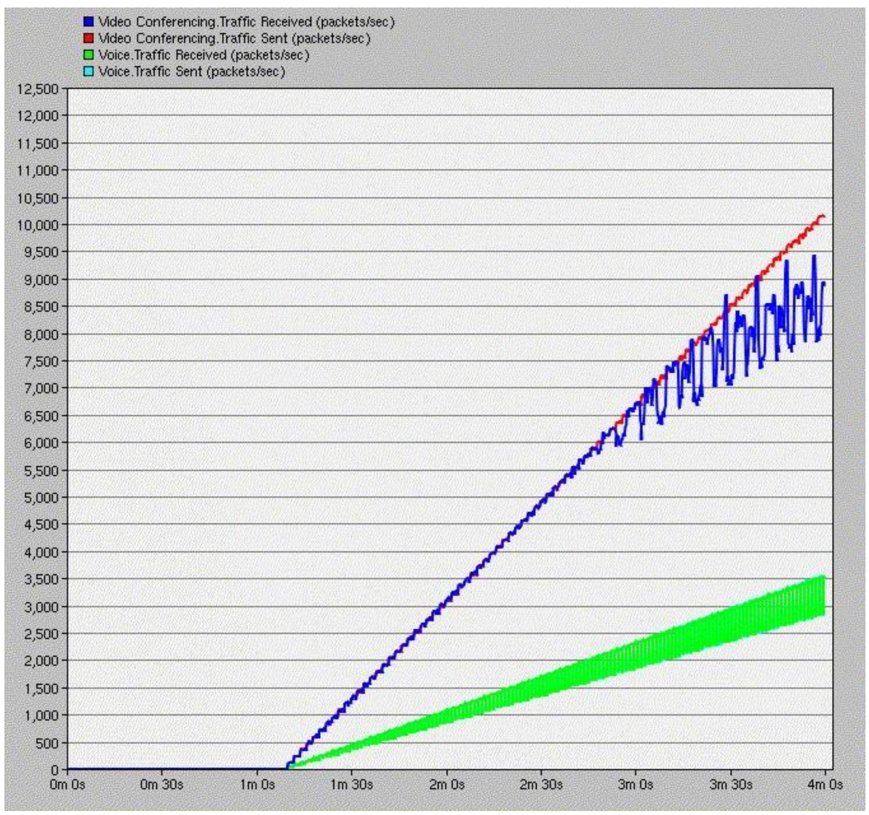


Figure 10: 802.11g Videoconferencing traffic (packets/sec)

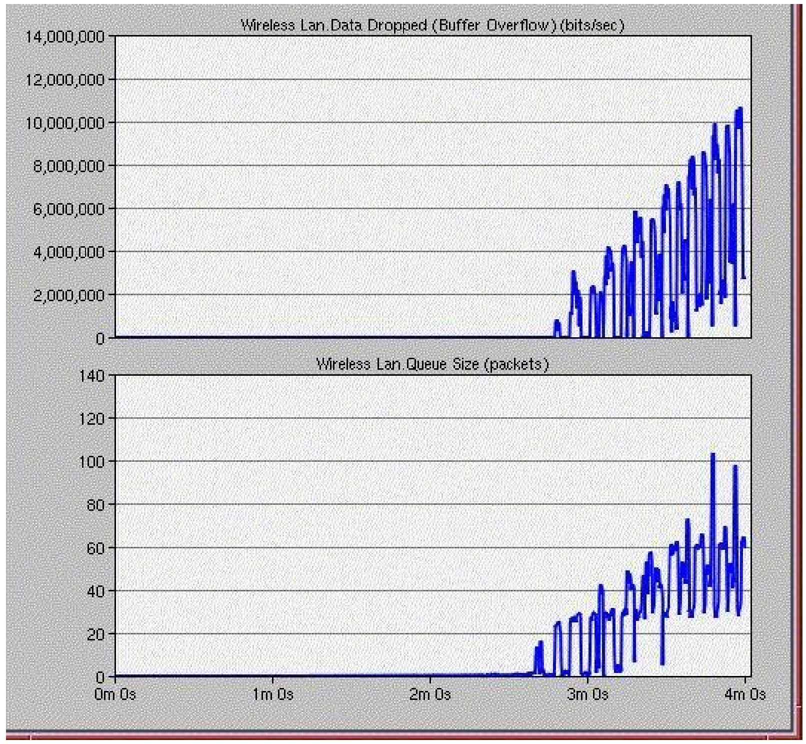


Figure 11: 802.11g WLAN's Queue Size (packets)

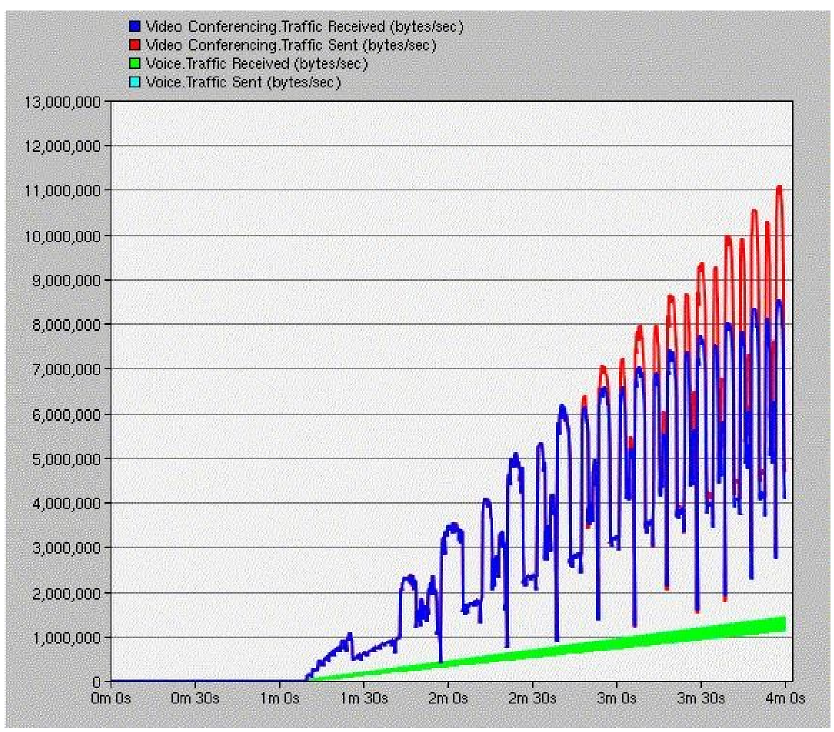


Figure 12: 802.11g Videoconferencing traffic (bits/sec)

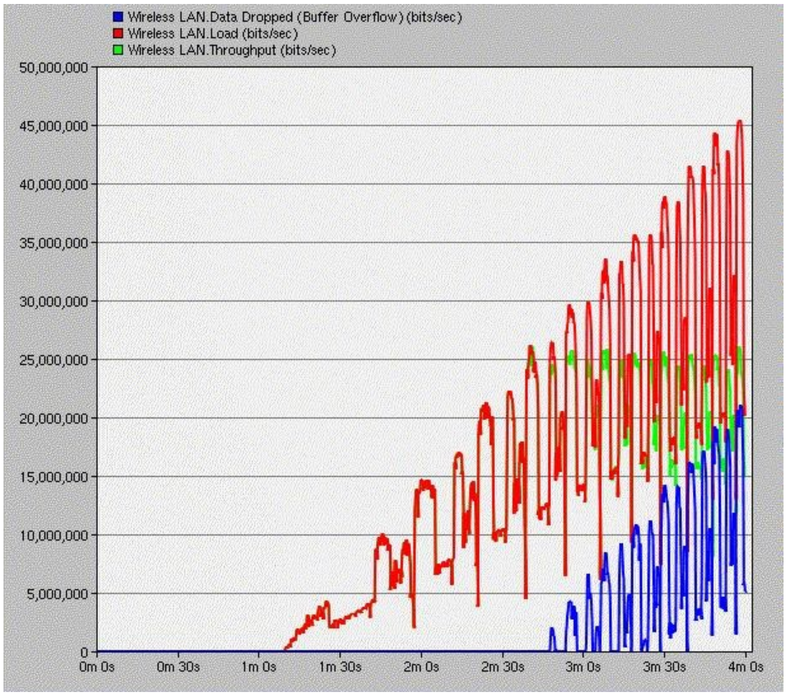


Figure 13: 802.11g WLAN's Load, Throughput and Data dropped (bits/sec)

5.2. Results Verification

In the first two scenarios we have demonstrated maximum number of videoconferencing calls that can be supported by the example deployed network. The average throughputs of 802.11b and 802.11g have been shown to match expected as measured by other studies. The capacity that we measured included the calls that are were established not only between the wired and wireless clients, but also between the wired clients only. The third scenario is used to demonstrate the calls that are established through the wireless router only; thus, allowing us to measure the capacity of the wireless LAN to support videoconferencing calls. IT Guru Academic Edition has been used. Please refer to the figures 14 and 15.

Instead of increasing the number of calls every 2 minutes, a wireless subnet was created and wireless clients are added manually. It was found out that 8 wireless clients and 1 wired workstation were performing well without any packet drops. However, once the 9th wireless client has been added, the mismatch has been observed as in the first two scenarios. Thus, 9 videoconferencing calls can be established over the wireless LAN connection. Please refer to the figures 16 and 17. Similar results can be obtained for 802.11g.

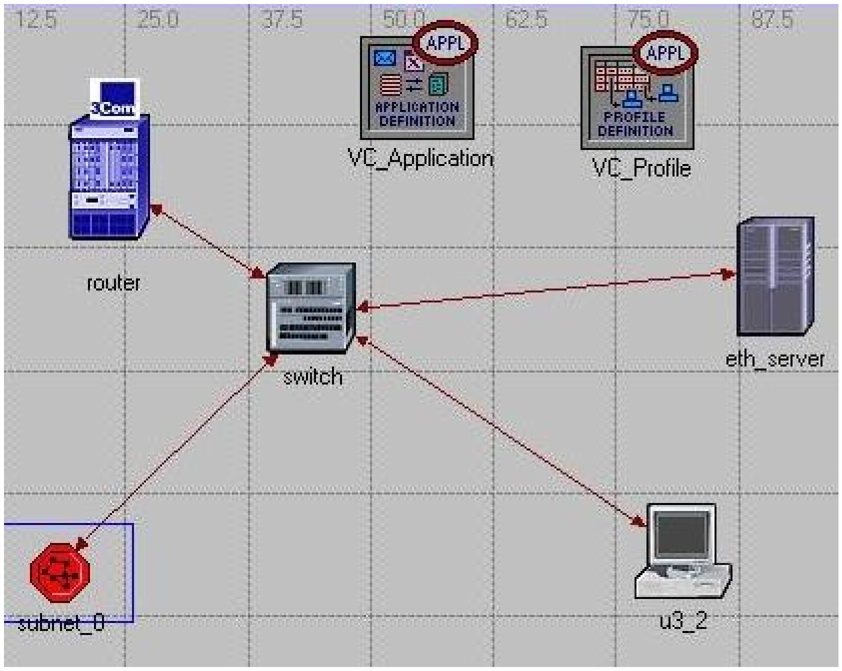


Figure 14: 802.11b Network Topology

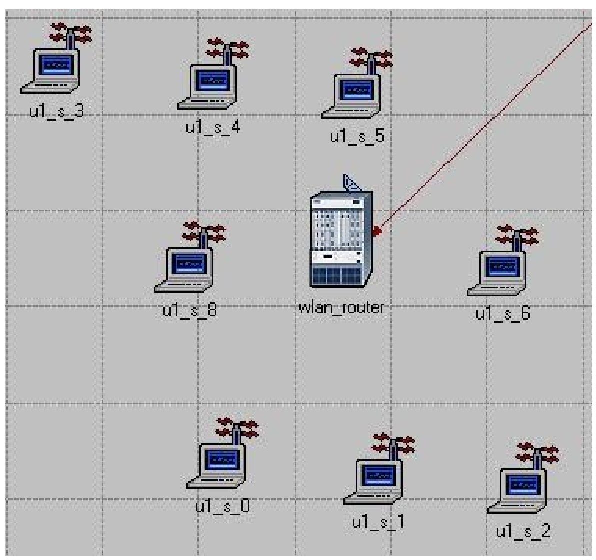


Figure 15: Subnetwork with 8 wireless clients

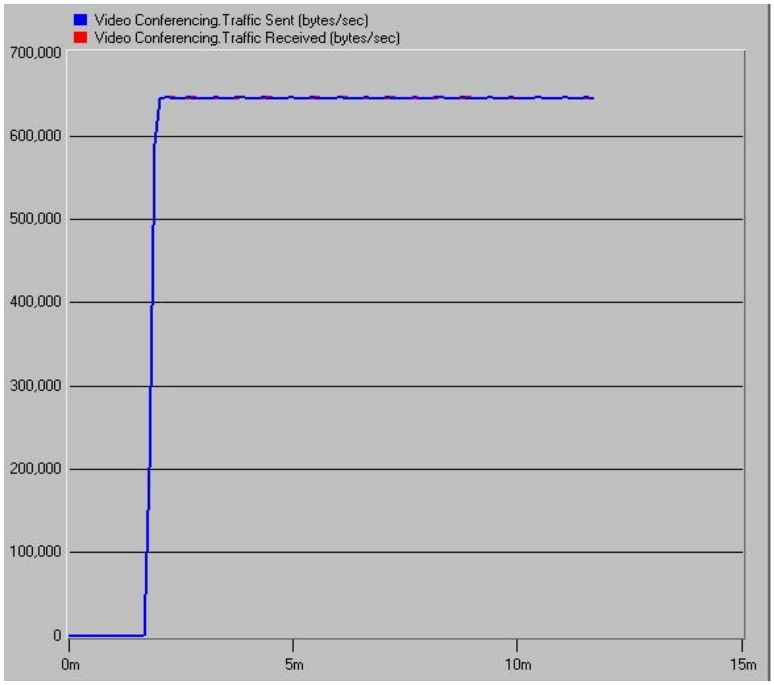


Figure 16: 802.11b Videoconferencing traffic sent and received (bytes/sec) – wireless 8 clients

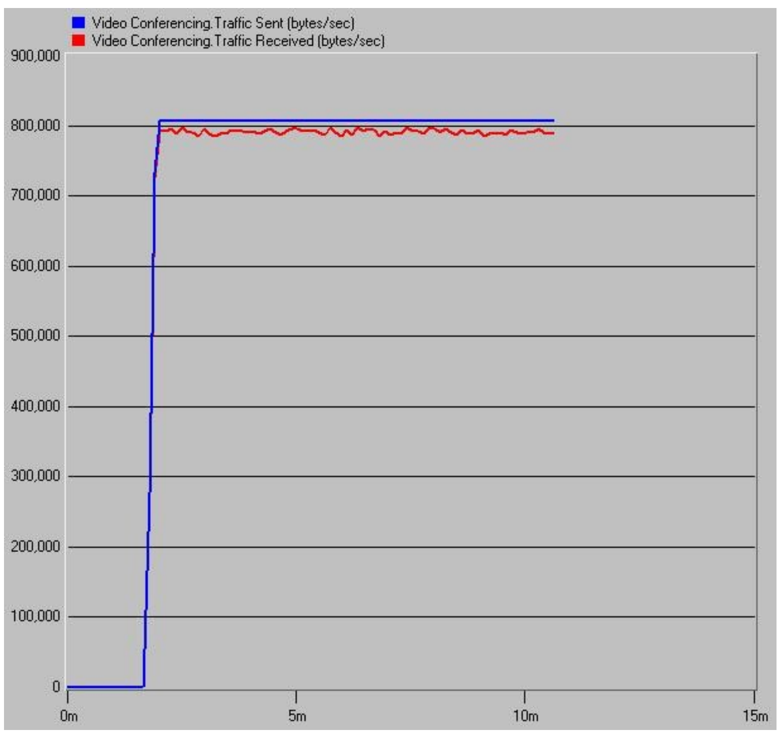


Figure 17: 802.11b Videoconferencing traffic sent and received (bytes/sec) - 9 clients

6. DISCUSSION AND CONCLUSIONS

In this project we have demonstrated how to simulate the deployment of videoconferencing application over an existing network. A two-floor typical office network has been used. The number of videoconferencing calls that can be established in the chosen network has been calculated. Wireless router's traffic sent and received, as well as load, throughput, the number of dropped packets and queue delays have been analyzed and proven to agree with each other. Average throughput for 802.11b and 802.11g has been determined and confirmed with the results found by other authors. Verification of our own work has been done using IT Guru Academic Edition. Detailed discussions and analysis can be found in the previous section.

7. REFERENCES

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