Deploying IPTV Services Over Proposed Network for Yemen Universities

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Abstract

Recently, organizing Internet Protocol Television (IPTV) services over existing IP networks became an active subject in the research. An analytical approach for deploying these services in a network designated for the higher education sector in Yemen will be presented. This network will interconnect, mainly, the Yemeni universities for offering some advanced information technology services, such as e-Learning e-Library and video on demand (VoD).

The Multiprotocol Label Switching (MPLS) is an advanced technology that enables IP networks to support traffic engineering efficiently. It speeds up packet forwarding and improves the network scalability by using multicast label switching to forward IP multicast packets at layer 2 with minimal forwarding at layer 3. A mechanism for supporting IP multicast in MPLS will be deployed for the propsed network. Also, an approach to measure the number of users to be supported by this network will be introduced.

Keywords: *MPLS*, *IP multicast*, *traffic engineerin, IPTV*.

1. Introduction

Internet Protocol Television (IPTV) is a set of multimedia services that are distributed throughout an IP network, where the end user receives video streams through a set-top-box (STB) connected to a broadband connection. IPTV is often combined with services of VoD, Pay Per View (PPV), Network-based time shifting and Subscription Services [1][2]. These services must possess an adequate level of quality of service, security, interactivity, and reliability. From the perspective of the provider, IPTV includes the video acquisition, processed video, and video secure distribution on the IP network infrastructure. Researchers have been looking forward for multicast technology instead of unicast technology. By using a multicast communication, the deployment of multimedia application can be facilitated greatly [3]. In multicast communication, only one copy of data packet is sent using a single address for the entire group of receivers. The two most important benefits of using multicast are: it reduces the load on the server as servers send only one packet instead of multiple packets to the receivers, and it reduces the overall bandwidth consumption as only one data packet is transmitted wherever possible. So, in order to provide resilient IPTV service proper multicast mechanism is essential [4][5][6].

Deployment of IPTV services over existing IP networks is very limited in scope and has become one of the intense interest subjects in the research arena. We present an analytical approach for deploying these services in the proposed network. We are considering MPLS and IP-Multicast techniques to deliver such IPTV streaming video. Our approach predicts the maximum number of IPTV streaming video sessions that can be sustained by the proposed network. Protocol Independent Multicast-Sparse Mode (PIM-SM) is used to deploy IPTV streaming video, which is the popular multicast protocol used. The simulator OPNET 14.5 modeler is used for simulation purposes in order to validate results obtained by performing analysis works [7][8].

2. Analytical Approach

Here, the analytical approach to deploy IP-Multicast IPTV services over a university network is presented. The actual number of users that the network can sustain is limited by two metrics. These metrics are the available bandwidth and delay. In order to determine the maximum number of users that can be supported by the modeled network while maintaining the Video delay constraint, we develop an algorithm that basically determines the network capacity in terms of multicasting video sessions that can be sustained by computing the number of users based on delay and bandwidth [9].

3. Proposed Algorithm

The algorithm presents the overall computed maximum number of multicasting video sessions based on the video delay constraint. It takes as input the number of users denoted by U, the inter-arrival time that indicates each generated video session is arrived after constant time denoted by T_{ai}, the time of the last successful video traffic denoted by two parameters: *min* for minutes and *sec* for seconds. Finally, the start time with time offset that indicates the time at which the video session starts. This time is denoted by T_{st}, so the formula that can be used to compute the maximum number of IPTV sessions that the network can sustain is $U_N = U + [(min*60+sec-T_{st})/T_{ai}]*U$. The algorithm steps are shown in Fig 1.
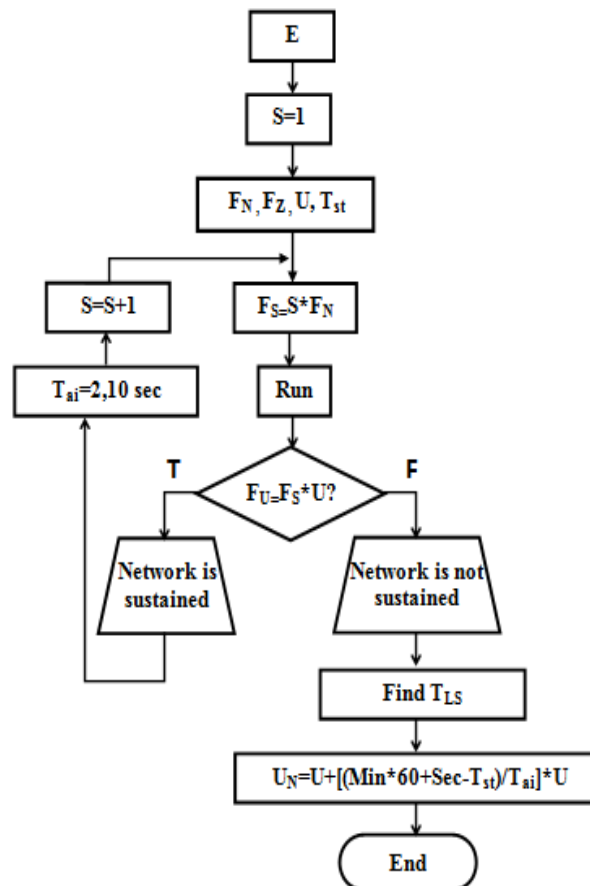

Fig 1: Algorithm to compute maximum number of multicasting video sessions

Inputs:
1. E: Network elements.
2. s: Number of sessions. Initially, set to s=1.
2. F_N : Number of frames that can be delivered in a second.
3. F_Z : Frame size in bytes.
5. U: Number of users.
6. T_{st} : Start time + time offset.
7. T_{ai}: Inter-arrival time.
8. F_S : Number of frames that can be sent by the sender.

Processes and Outputs:
9. Run the network and get the time of the last successful video traffic in minutes (min) and seconds (sec).
10. Calculate the number of frames that can be received by the receiver (F_U).
11. Check whether $F_U=F_S * U$,
12. If Yes, then the network is sustained and go to loop; take arrival time T_{ai} 10 or 2 seconds and increase S by one.
13. If No, then the network is not sustained; and find T_{LS}.
14. Calculate the maximum successful video sessions (or users) that can be sustained by a network $U_N=U+[(min*60+sec-T_{st})/T_{ai}]*U$.

4. OPNET Simulation

The OPNET Modeler is a commercial discrete-event simulator. It provides a global environment to model, simulate, and evaluate performance of all models of commercially available network elements, and has various real-life network configuration capabilities [10]. This makes the simulation of real-life network environment close to reality.
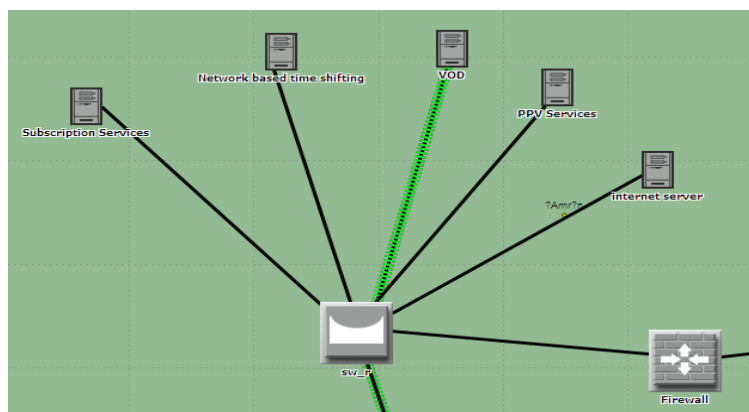
5. Modeling The Network

Our simulation modeling for the case study is shown in Fig 2. In OPNET, many vendor specific models are included in the pre-defined components libraries. However, the specific devices that we need are ppp_server_adv servers in the main centre network and university networks, ethernet_wkstn_adv for receivers in the main center and departments of the university colleges, eight ethernet2_slip8_gtwy routers for LERs and LSRs of the MPLS backbone network. Also, there is need for three CS_7206_6s_a2_ae8_f4_tr4_slip16_adv routers for multicast backbone network in every university, CS_C3512XL_ae12_adv_46_ upgrade switch to connect the servers of the main centre network, CS_C3512XL_ae12_adv switch for branching the college to its departments, CS_2948G_ae48_ge2_adv switch for connecting the devices of the receivers in the department, PPP_SONET_OC3 cables for connections between the LSRs,PPP_DS3 cables for connections between LERs and custom routers and for connections between LERs and LSRs and also to connect the servers with its switch router in the main centre, two 1000BaseX_adv cables for multicast backbone connections, 100BaseT_adv cables for college connections. Fig 2 shows the components of the proposed overall network, which is limited here to four main sites. Exactly, the four sites represent the main data-center, Yemen center for information technology in higher education (YCIT-HE) in the capital, Sana'a; and the data centers of the three main public universities (the University of Sana'a, the University of Aden, and the University of Hadramout).
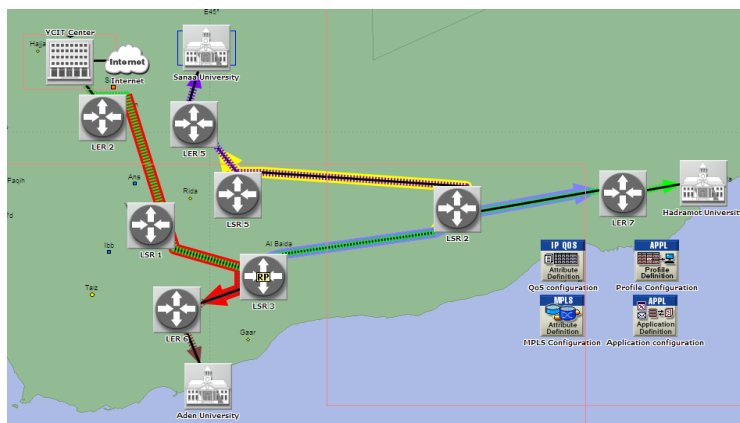
Fig 2: OPNET model of the proposed IPTV network for Yemen universities

Fig 3 illustrates the OPNET model for the main data centre (YCIT-HE data centre), where some of the essential hardware/software components exist. Mainly, central servers, routers, switches, firewalls, services, and systems are involved. From the other hand, Fig 4 shows a part of the network inside one of the universities contained in this network; namely the University of Aden.

Fig 3: OPNET model of the YCIT-HE Centre network
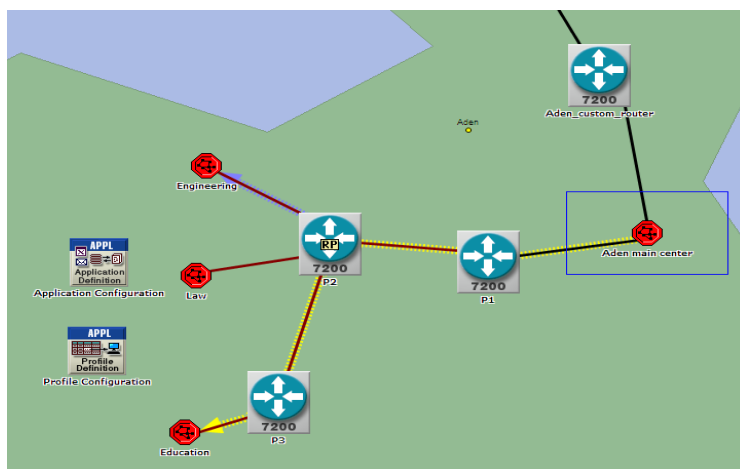
Fig 4: OPNET model of the network of some colleges of Aden University

6. Experiments

As we know, in a multicast technique, the sender is responsible for sending the packets to all receivers. So, the head end server is responsible to run the profile. To study how MPLS technology increases the throughput of the main backbone network and thus increasing the traffic received in the receivers of the data centers of the universities, we configure two scenarios:

1. The first scenario consists to deploy the main multicast backbone network with MPLS.
2. The second scenario is for deploying the network without MPLS.

After that, we want to study how many users can be sustained by a university network. For this reason, we configured five scenarios.

3. The third scenario consists of deploying IPTV channel over the network. We configure this scenario where every five seconds two users are added. This shows how to add session to the two users, and from this scenario we can't reach the maximum number of users can be supported.

4. The scenarios four to seven are configured in the following table:

Scenario	Inter-repetition time [sec]	Number of Frames per second	Frame Size (bytes)
4	10	30	1300
5	2	50	1300
6	10	30	1300
7	2	50	1300

These scenarios are configured every 10 and 2 seconds based on the number of frames that may be 30 or 50 frames, see Fig 5(a,b), and frame size equals to 1300 bytes where we add a session concurrent to 2 users, then add other session to other two users, and so on [11]. Fig 6 (a,b) shows the repeatability for these scenarios.
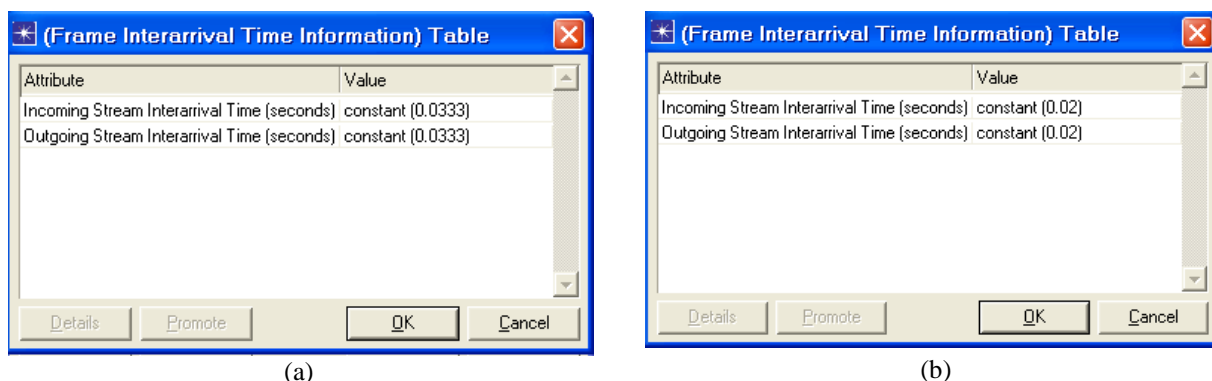

(a)   (b)

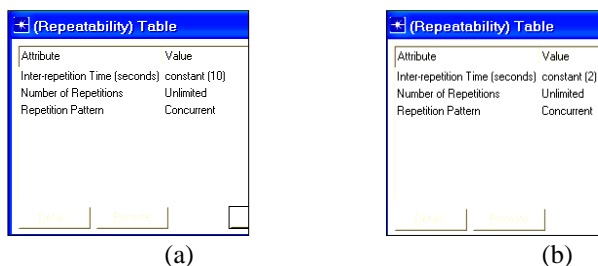**Fig 5: Configuration to set the Number of frames (a) 30 Frames/sec, (b) 50 Frames/sec.**

(a)   (b)

Fig 6 : Configuration of the Inter-repetition time in the scenarios 4 to 7
(a) repetition every 10 sec, (b) repetition every 2 sec

7. Simulation Results

In this section, the simulation results are presented by deploying the IPTV services over the network that is composed of the main MPLS backbone network and university networks based on IP-Multicast. From Fig 7, the results of the scenarios 1 and 2 show the throughput is increased with MPLS and so the traffic received is increased.
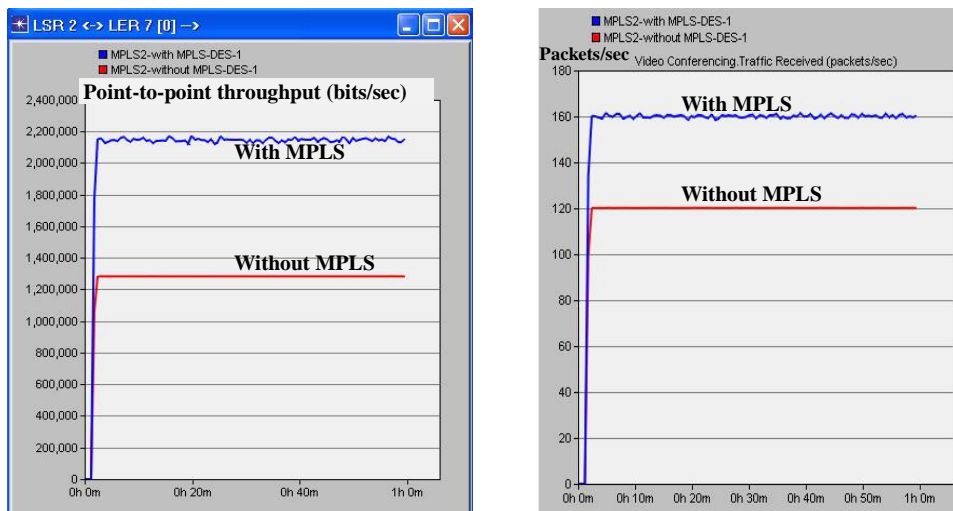

Fig 7: (a) The point-to-point throughput from (b): The traffic received of the receiver of
LSR2   to LER7 for both scenarios   a university data-center for both scenarios
1 and 2.   1 and 2

Our approach was to measure at the end of the simulation how many users can be sustained by this network. We modeled two scenarios. The third scenario was with a profile modeled to add two users in serial. Hence, this senireo adds the session to the same number of users(two users only). We can't reach the maximum number of users that can be sustained. The profile in this scenario is modeled to start the traffic at 110 seconds of the simulation run with two users are added. This happens by sending the head end server session to the Backbone router, then the Backbone router replicate this session and send it to support two users. After that, every 5 seconds a new session added to the same two users.

Fig 8 (a,b) shows the two users added in the traffic received and the traffic sent one session added from the receivers of the departments of the colleges and from the head end server of the data-center of the university respectively then every 5 seconds session is added.

**(a)**   **(b)**
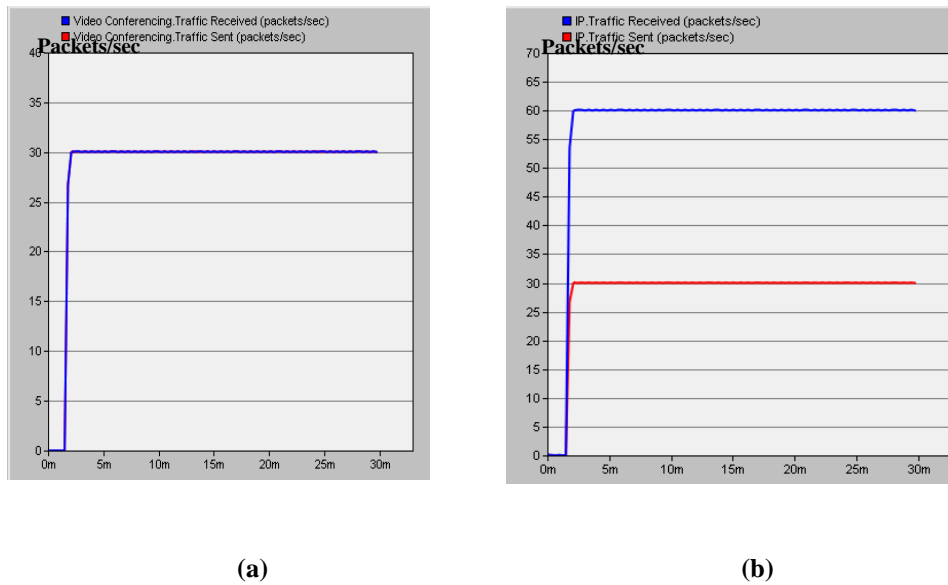
Fig.8: Traffic sent and received for one session (a) at receiver, (b) at sender

The scenarios four to seven are modeled by adding two users to show the IPTV channel with 30 and 50 fps. Then, after 10 and 2 seconds another two users added again to see the same channel, and so on. We can calculate the maximum number of users by equation

$$2 + \left[\frac{\min \times 60 + \sec - 110}{2 \, or \, 10} \right] \times 2$$

For each case, the supported number of users, will be obtained by replacement in this equation [10], as shown by the following paragraphs.
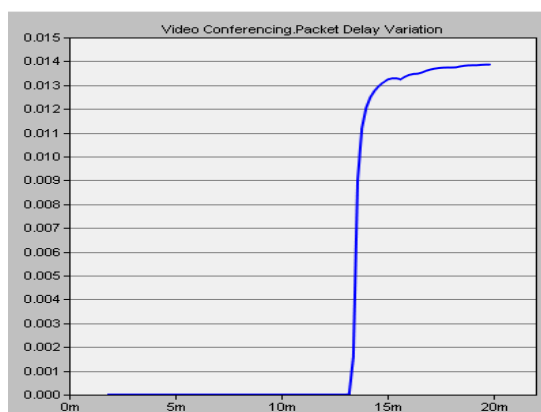
From Fig 9 (a, b, c) scenario 4 shows that the last successful user joint group was approximately at 13 minutes and 30 seconds for inter-repetition time 10 seconds and the number of frames is 30 frames/sec. So, the maximum number of users that can be supported at this point is $2 + \left[\frac{13 \times 60 + 30 - 110}{10} \right] \times 2 = 142 \, \text{users}.$
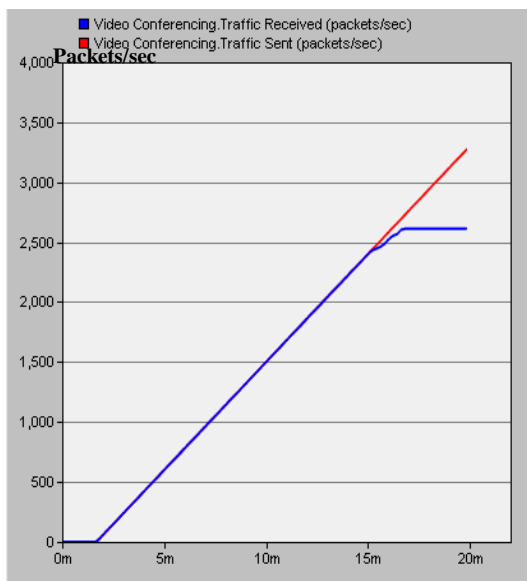
(a): The traffic received and the traffic sent of the receiver (packets/sec-vs-Time)
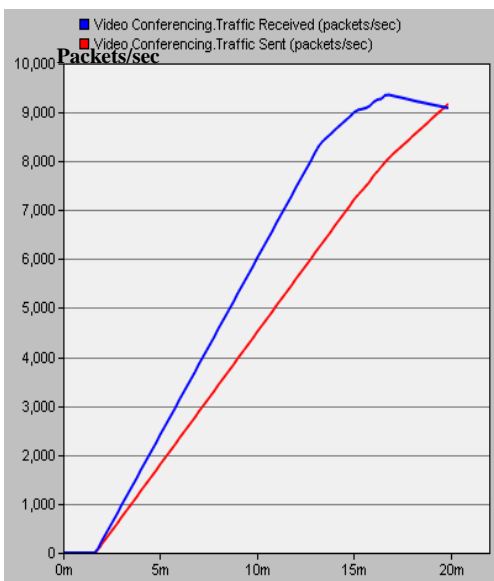
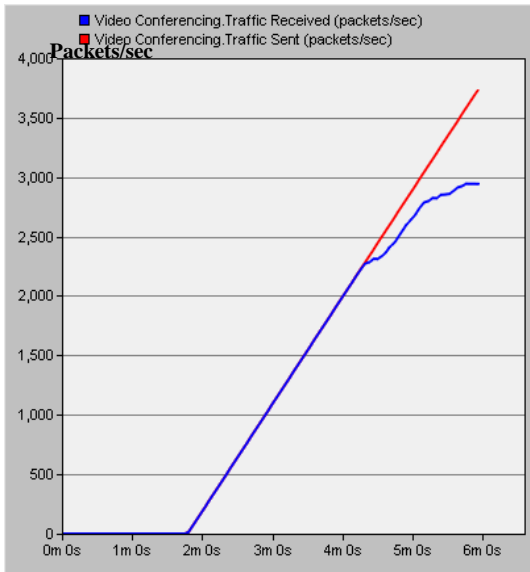**(b): The traffic received and the traffic sent as global statistics (packets/sec-vs-Time)**

(c): Packet Delay Variation as global statistics (packet delay variation-vs-time)
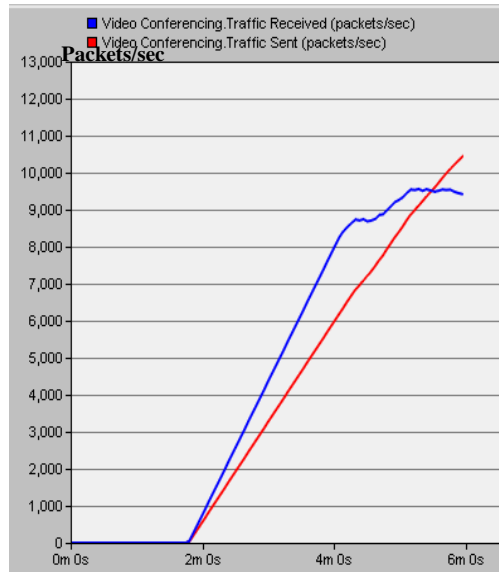
Fig 9: The last successful user join group for scenario 4

From Fig 10 (a, b, c) scenario 5 shows that the last successful user joint group was approximately at 4 minutes and 14 seconds for inter repetition time 2 seconds and the number of frames is 30 frames/sec . So, the maximum number of users that can be supported at this point is $2 + \left\lceil \dfrac{4 \times 60 + 14 - 110}{2} \right\rceil \times 2 = 144$ users.
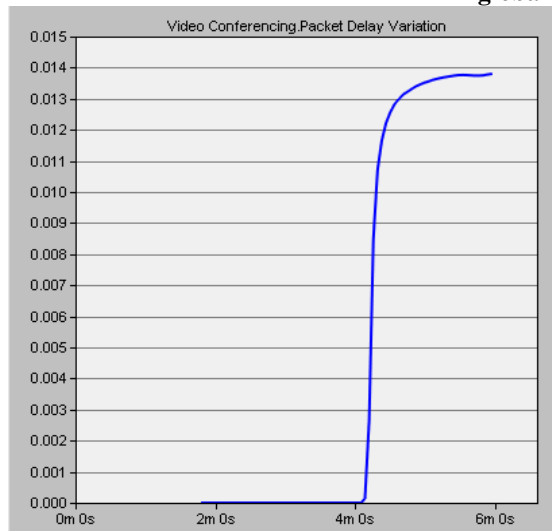
(a): The traffic received and the traffic sent of the receiver

(b): The traffic received and the traffic sent as global statistics
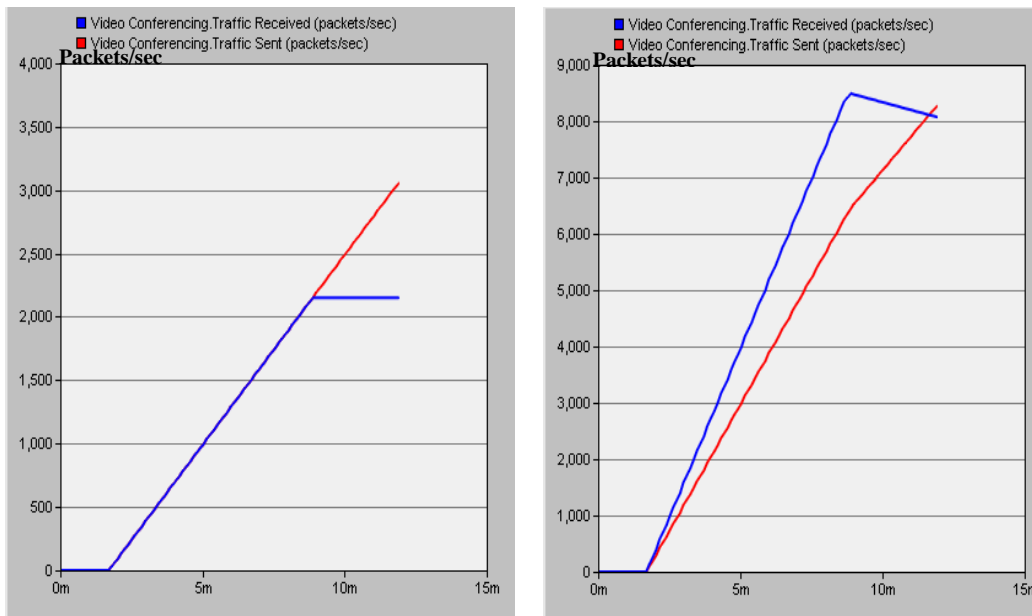
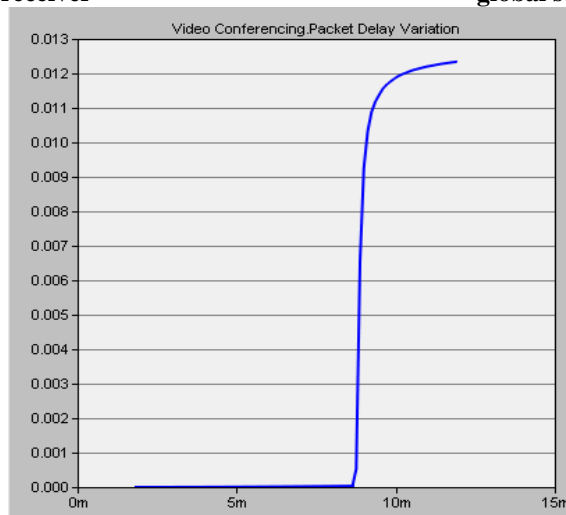

(**c): Packet delay variation as global statistics**
Fig 10: The last successful user join group for scenario 5

From Fig 11, scenario 6 shows that the last successful user joint group was approximately at 8 minutes and 45 seconds for inter-repetition time 10 seconds and the number of frames is 50 frames/sec. So, the maximum number of users that can be supported at this point is $2+\left[\dfrac{8\times 60+45-110}{10}\right]\times 2 = 85$ users.

(a): The traffic received and the traffic sent of the receiver

(b): The traffic received and the traffic sent as global statistics
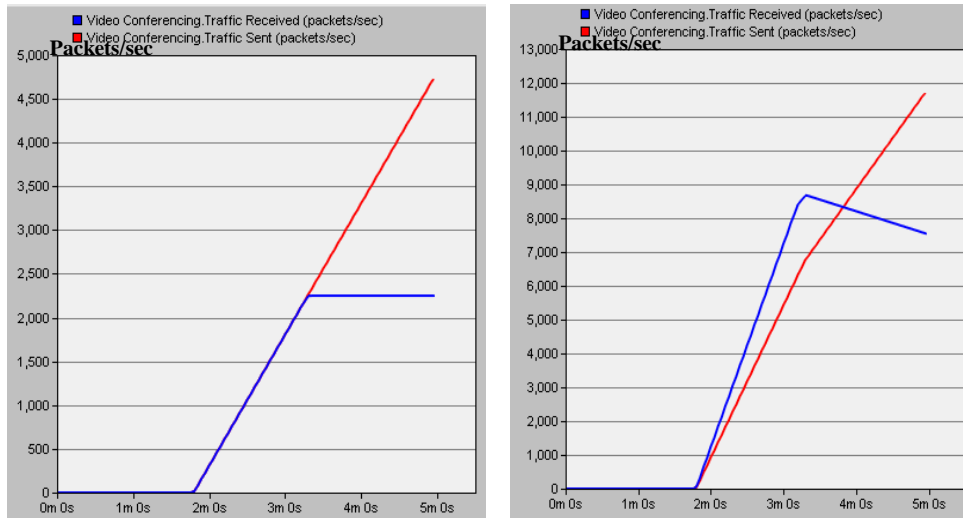
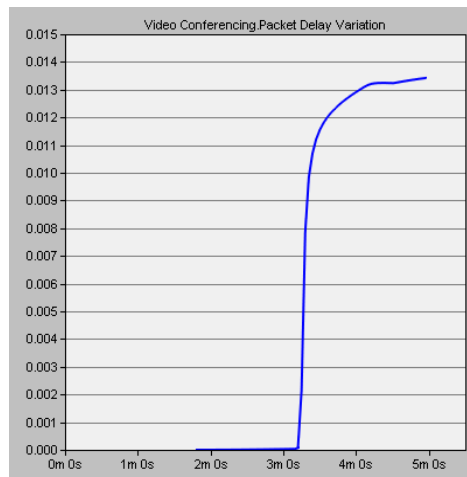

(c): Packet delay variation as global statistics
Fig. 11: The last successful user join group for scenario 6.

From Fig 12, scenario 7 shows that the last successful user joint group was approximately at 3 minutes and 18 seconds for inter-repetition time 2 seconds and the number of frames is 50 frames/sec . So, the maximum number of users that can be supported at this point is $2 + \left[\dfrac{3 \times 60 + 18 - 110}{2} \right] \times 2 = 90$ users.

(a): The traffic received and the traffic sent of the receiver

(b): The traffic received and the traffic sent as global statistics

(**c) Packet delay variation as global statistics**

Fig. 12: The last successful user join group for scenario 7.

8. Conclusion

This paper presented a simulation approach on how interactive multimedia can be deployed successfully based on MPLS and IP-Multicast. From the results we conclude that the throughput and the traffic received are increased in the network that support MPLS technology. Also it is to be concluded that the network cannot be sustained if the received traffic became smaller than the sent traffic or if the packet delay variation increases sharply at the time that determines the maximum number of users or sessions that can be sustained by the network. We see that the network can be sustained by smaller number of users if there are large number of frames in a second.

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