

Performance Evaluation of Unidirectional Lightweight Encapsulation using *ns-2* and DVB-S TestBed

Way-Chuang Ang¹, Chee-Hong Teh², Tat-Chee Wan³, Rahmat Budiarto⁴

Network Research Group
School of Computer Science
Universiti Sains Malaysia

11800, Minden,
Penang, Malaysia

{wcang¹, chteh², tcwan³, rahmat⁴}@nrg.cs.usm.my

Abstract—Unidirectional Lightweight Encapsulation (ULE) is a recently published standard to overcome the efficiency problems of Multi-Protocol Encapsulation for satellite data transmission. This paper presents a comparative study of Unidirectional Lightweight Encapsulation performance using *ns-2* [1] simulation and a DVB-S testbed. Performance metrics such link utilization, packet loss and latency were taken. It was found that the results of both approaches were comparable

Index Terms—DVB, IP over MPEG-2 TS, MPE, ULE

I. INTRODUCTION

Satellite communications play a vital role in providing communication links to geographically remote sites. Traditionally we use satellites to receive Standard Definition Television (SDTV), High Definition Television (HDTV) programs, and for voice telephony. However, due to the rapid convergence of data and multimedia applications, the Internet has become an important service for a vast majority of people today. The use of satellite links has changed from not only supporting traditional media applications to packet data transmissions. Now, satellites that were initially deployed for military communications, voice telephony and TV broadcasts are being used to offer Internet access globally. Many Internet Service Providers (ISPs) offer global Internet access using satellite links. Broadcasters can dedicate a portion of a satellite broadcast channel that was traditionally used for audio and video streams to include unicast or multicast IP traffic [2].

Data is sent directly to satellite from a single source, and then received by a large downstream population [3] using packet data services over (Digital Video Broadcasting) DVB, e.g., Internet/World Wide Web (WWW) access with high speed transmission via a DVB satellite hub station to transmit packet data to users' satellite dishes used for TV distribution. This data broadcasting technology enables broadcasters to broadcast large amount of data to personal computers using MPEG-2 Transport Streams (MPEG-2 TS) [4].

MPEG-2 is the standard for generic coding of moving pictures and associated audio information. It has been widely

used to define the standard for high quality video and audio compression which is applicable to high quality real-time conferences. The MPEG-2 standard defines a method for multiplexing different elementary stream types to form a Transport Stream (TS) used in broadcasting systems. This paper studied the performance of IP traffic over DVB Transport Streams. MPEG-2 TS was designed for transporting MPEG-2 over unreliable media where errors usually occur. The TS is a stream that consists of consecutive and relatively short fixed-length packets. The packet length for MPEG-2 TS is 188 bytes. Each TS packet structure starts with a 4-byte fixed length TS header, followed by a 184-byte optional adaptation field and payload as shown in Fig. 1

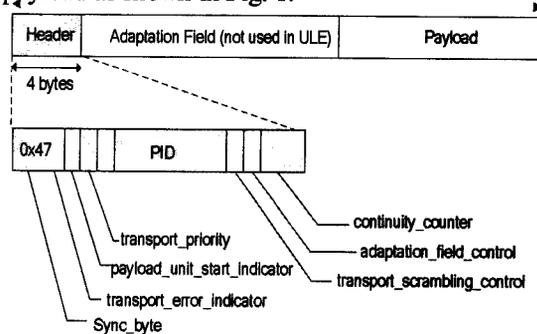


Fig. 1 Transport Stream and Header structure

II. IP OVER MPEG-2 TS

The European Telecommunication Standards Institute (ETSI) specified several methods to transmit network data over MPEG-2 TS. The most widely used method introduced in [5] is called Multi Protocol Encapsulation (MPE). Later, the Internet Engineering Task Force (IETF) proposed an alternative encapsulation method named Unidirectional Lightweight Encapsulation (ULE) [6] that is simpler than MPE.

Unidirectional Lightweight Encapsulation is a recently published standard. A ULE packet is layered directly into the

payload field of MPEG-2 TS frame. Similar to MPE, a ULE packet may span several MPEG-2 TS frames if its size exceeds the size of a MPEG-2 TS payload field. The 2 mechanisms used to encapsulate ULE into MPEG-2 TS frame are packing and padding. These mechanisms will be discussed in detail in later section.

True to its name, ULE only carries mandatory fields and is lightweight by design. The benefits of the simplicity of ULE packet format are two-folds. Firstly, having fewer mandatory fields translates to greater saving in bandwidth. This fact is greatly amplified in situation where bandwidth resource is prohibitively expensive as in satellite networks employed by DVB-S and DVB-S2 standards. Secondly, simple packet header makes ULE easier to be processed and thus demands less processing power on the participating nodes than that of MPE.

In order to meet the criteria of optimized transmission without sacrificing flexibility, ULE introduces extension headers to carry additional header information. Fig. 3 illustrates the packet format of a ULE packet with only mandatory fields. The type field of ULE packet is used to identify the existence of extension header in a packet and the type of payload carried in PDU field. For packets with type field that contain values less than 1536, extension header is present and the values are assigned by IANA. For type field that contains values greater than 1535, the values correspond to the type code specified by the IEEE/DIX type assignment for Ethernet. Thus, both IPv4 and IPv6 are supported directly using the type field without requiring any modification.

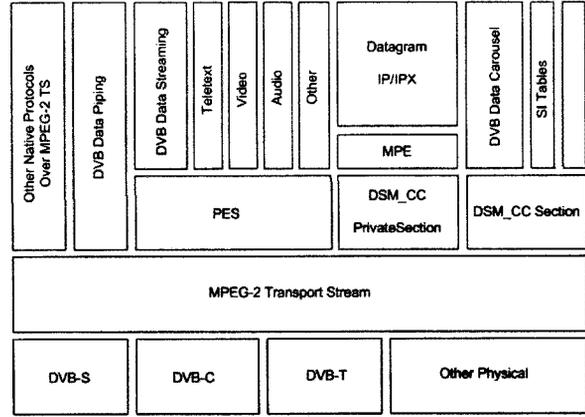


Fig. 3 Architecture of DVB protocol

Before getting into more details about the efficiency offered by MPE and ULE, the two encapsulation mechanisms - Padding and Packing mode will first be briefly discussed.

In MPEG-2 TS Padding mode, a large SNDU need to be fragmented and encapsulated independently into TS packets. Since a TS packet payload is a fixed size cell with 184 bytes, a SNDU, generally, does not fit into an integral number of TS packets. Therefore, the unutilized part of the payload will be padded with stuffing bytes (0xFF also known as NULL bytes). Padding is required to align the end of the data with the final TS packet so that the TS packet maintains 188 bytes.

In MPEG-2 TS packing mechanism, each new SNDU does not have to start in a new TS packet. Like padding mode, a large SNDU will be fragmented and encapsulated independently into TS packets. All filled TS packet will be transmitted immediately but for the last TS packet with some unutilized space, the unutilized part of the payload will not be padded with padding bytes. In the same TS packet, if there are leftover spaces, a new SNDU can start immediately following the end of the previous SNDU payload. Therefore this mechanism has removed the need for padding.

In MPEG-2 TS data transmission, the transmission efficiency can be defined as the ratio of total number of information bytes over the total number of bytes in transmission [8]. By using MPEG-2 TS Padding mode, the transmission efficiency offered by this mode can be calculated based on equation below. Assuming that the sizes of MPE or ULE header is H , L is the size of the SNDU, S is the 1 byte Payload Pointer (S is inserted into first byte of MPEG-2 TS payload field whenever there is a new SNDU) and I is the length of IP datagram. The TS payload utilization, U , which is the total number of TS packets required to transmit the L , can be denoted as:

$$L = H + I$$

$$U = \left\lceil \frac{L + S}{184} \right\rceil$$

where U here is always the smallest integer greater than or equal to the real number U . To calculate the Transmission Efficiency $T_{padding}$ for MPEG-2 TS padding, from (2), $T_{padding}$ can be defined as:

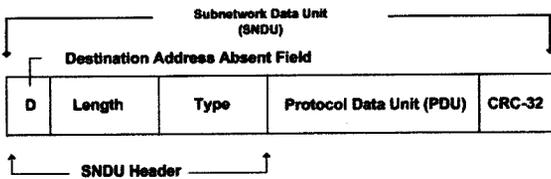


Fig. 2 Unidirectional Lightweight Encapsulation packet format

III. IP DATA ENCAPSULATION PROCESS -EFFICIENCY ANALYSIS AND COMPARISON

In MPEG-2 TS, there are two mechanisms to encapsulate IP packets into MPEG-2 TS using MPE and ULE. These two mechanisms are known as *padding* mode and *packing* mode [7]. To encapsulate an IP packet into MPEG-2 TS, multiple overheads are introduced during the encapsulation process as shown in Fig. 3. During this process, IP packets need to go through multiple protocol stacks, as a result, various overheads will be added, and how much overheads are inserted into the packet during this process depends on the PDU size and also the encapsulation method being used. This amount of overhead will give a direct impact to MPEG-2 TS system transporting IP data.

$$T_{padding} = \frac{I}{U \times 188} \times 100\%$$

Similarly, when the MPEG-2 TS Packing mode is enabled, the TS payload utilization is:
From (2),

$$U = \frac{L + S}{184}$$

where U is an integer. To calculate the Transmission Efficiency $T_{packing}$ for MPEG-2 TS packing, from (4), $T_{packing}$ can be defined as:

$$T_{packing} = \frac{I}{U \times 188} \times 100\%$$

With these equations, the Transmission Efficiency gained using MPE and ULE can be theoretically analyzed, as shown in Fig. 4.

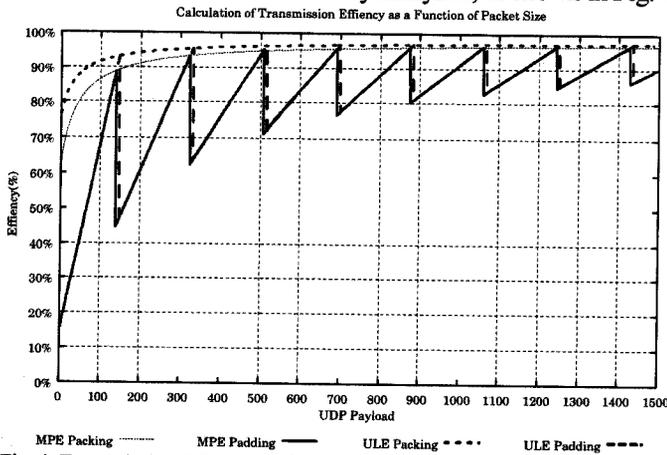


Fig. 4 Transmission Efficiency of MPE vs. ULE [8]

IV. IP DATA ENCAPSULATION– SIMULATION AND TESTBED SYSTEM

Performance of the newly proposed encapsulation method – ULE, has been evaluated using the *ns-2* simulation tool as well as a testbed environment. The aim of this research was to make a comparative study on the performance and the efficiency of the self implemented ULE encapsulator and the simulation model which was provided by European Space Agency (ESA).

A. *ns-2* Simulation

The *ns-2* simulation tool was used for evaluating the performance of ULE encapsulation. The ULE IP packet encapsulation module from ESA was used as the basis of the simulations described in this paper. The simulation model of IP over MPEG-2 TS is shown in Fig. 5.

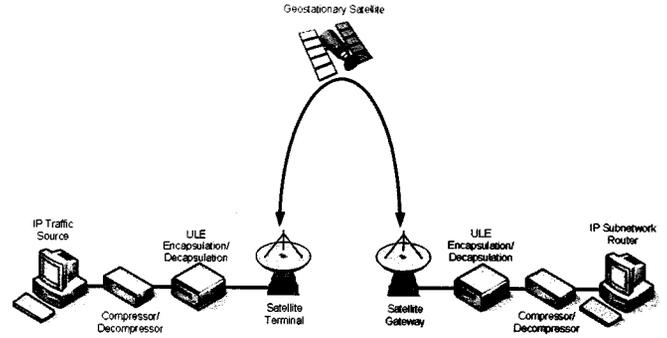


Fig. 5 Simulation model of bidirectional satellite link for IP over MPEG-2 TS

B. TestBed Environment

The layout of the testbed is depicted in Fig. 6 below. The testbed is composed of 2 Unidirectional Link (UDL) gateways with each gateway having a DVB-S modulator. Each UDL gateway has an Asynchronous Serial Interface (ASI) card and at least one DVB-S receiver card. Asynchronous Serial Interface inside UDL gateway sends MPEG-2 TS frames to DVB-S modulator. DVB-S modulator then converts digital data from ASI card into intermediate frequency (IF) in the range of 50-180MHz. Since DVB-S receiver card only accepts input within the range of 950 MHz – 2150MHz, an IF up-converter is needed to convert IF to L-band frequency. In the end, radio frequency is then demodulated into digital data by the DVB-S receiver card of the recipient UDL gateway.

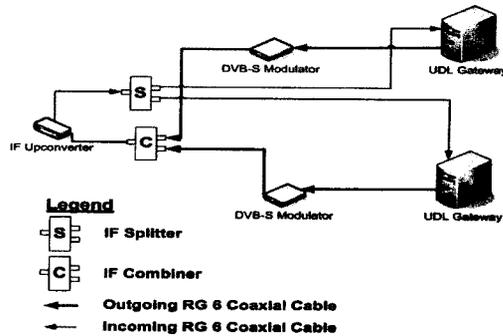


Fig. 6 DVB-S testbed

The decapsulation of ULE packets occurs in the kernel space of the operating system using the *dvbnet* software stack, whereas the encapsulation process occurs in user space using software developed by the author. Here, a simple bidirectional link between 2 sites was assumed; whereas the UDL gateway is capable of operating in a mesh environment with several sites [9].

V. ULE PERFORMANCE EVALUATION

The simulation results from *ns-2* were compared to the testbed results obtained using *iperf* version 2.0.2 and UDP utility programs on the testbed. Delay, packet loss and link utilization metrics were measured. Tests were conducted on 2Mbit/s, 4Mbit/s and 8Mbit/s links for all simulations and

experiments. Both padding and packing modes for ULE were tested. Packing mode tests were carried out using a packing threshold of 40 ms.

In addition, a delay of 250ms was added to each link in the testbed system via the Network Emulator (*netem*) module in Linux. The *netem* module is configured to emulate the 250ms propagation delay found in typical geostationary satellite links.

A. Link utilization tests

Link utilization tests were conducted for both TCP and UDP transport protocols. Due to the fact that TCP doesn't respect record boundary of data being sent, data fragments of smaller sizes may be accumulated into a big TCP packet before the actual transmission occurs. Thus, for testbed environment, the packet size of TCP is controlled by the MTU size of the transmitting network device. The packet sizes indicated in Fig. 7 until Fig. 12 are inclusive of the IP header, UDP header as well as UDP payload. Adjustments were made to the results obtained from *iperf* because *iperf* does not take the size of IP and transport protocol headers into account when calculating results.

It should be noted that the results used to derive the graph for link utilization of TCP traffic were the average values achieved after the TCP stream enters steady state. TCP protocol performance for initial window growth was not considered.

A comparison between simulated and actual link utilization for ULE padding mode is given in Fig. 7 and 8 for both TCP and UDP transport protocols. The saw tooth shaped result obtained from the simulator is due to the packet sizes. Valleys are formed whenever the packet sizes slightly exceed multiple of 188 bytes. For these packets, efficiency is reduced because much of the MPEG-2 TS frame was wasted on padding bytes. However, as the size of UDP packets increases, the saw tooth shape becomes less pronounced. This is because the number of padding bytes is overshadowed by the size of the actual packet. A close resemblance can be found between Fig. 4 and 8.

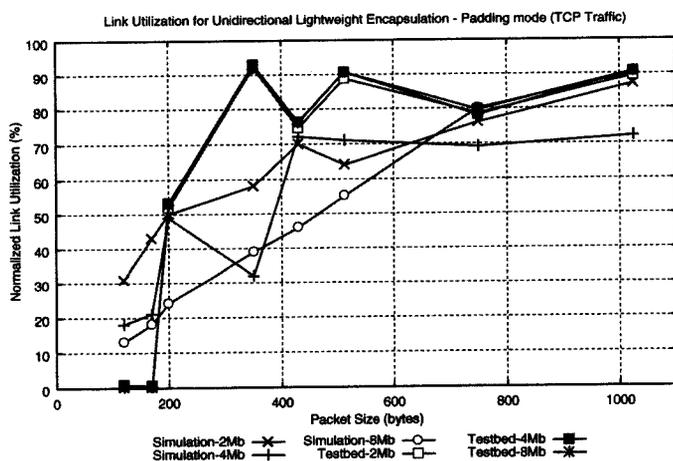


Fig. 7 Comparison of TCP link utilization using ULE padding mode

Surprisingly, results obtained from the testbed were better than that from simulation, especially for TCP traffic. However, results measured for TCP packet sizes less than 200 bytes on the

DVB-S testbed was dismal. This was largely due to the constraint of the TCP protocol because UDP traffic did not exhibit the same symptom for these packet sizes. Through observation, it is noted for these packet sizes, most of the time was spent on waiting for TCP acknowledgments because the advertised TCP windows from the receiver were very small.

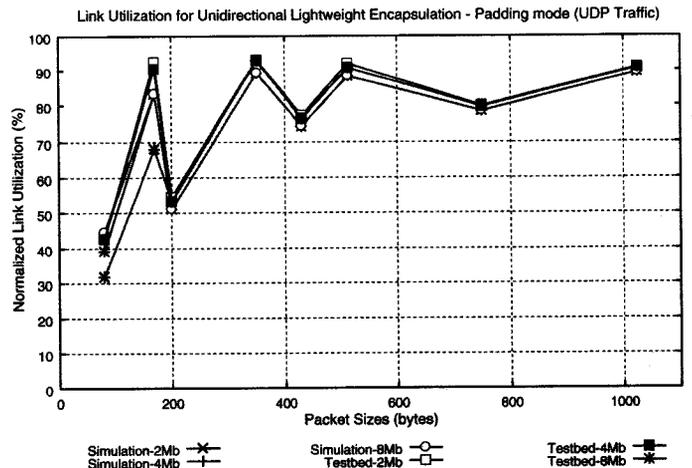


Fig. 8 Comparison of UDP link using ULE padding mode

The UDP packets loss rate shown in Fig. 9 is just the inversion of its UDP link utilization counterparts. The Bit Error Rate (BER) of these links was zero. The links were fed with constant stream of UDP traffic according to the bandwidth of the link. For an 8 Mbit/s link, the stream of UDP data is generated at a rate of 8 Mbit/s regardless of the packet sizes used. However, these UDP streams only take the payload of UDP packets into account. Thus, the actual generated UDP traffic would exceed the capacity of these links. This was more obvious at small packet sizes because the ratio of the combination of IP and UDP headers over UDP payload are higher for smaller packets. As such, some of the UDP packets were dropped due to buffer overrun at the transmitting interface.

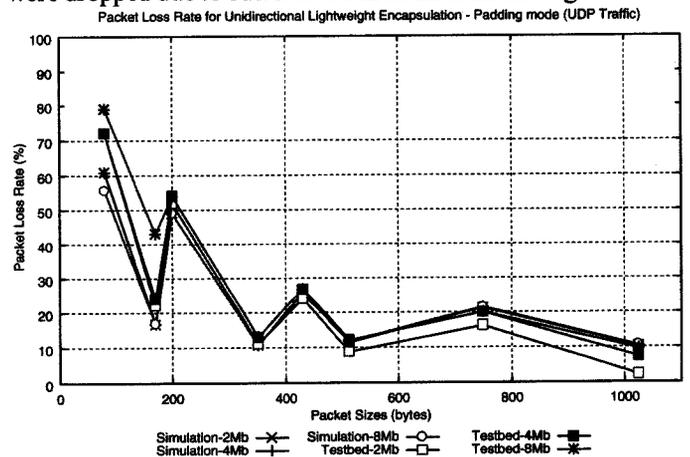


Fig. 9 Comparison of UDP packets drop rate using ULE padding mode

The second set of tests shown in Figs. 10, 11 and 12 were conducted using ULE packing mode. The measured UDP link utilization from the simulator and the DVB-S testbed were

pretty even and is close to theoretical limit [8]. Again, the same problem occurred for TCP packet sizes less than 200 bytes in size on the DVB-S testbed.

Packet loss rates were much better using ULE packing mode. However, the same buffer overrun problem still occurred for very small packet sizes.

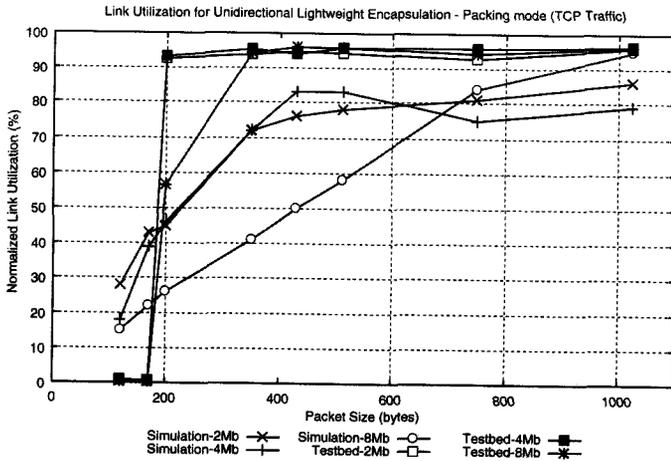


Fig. 10 Comparison of TCP link utilization using ULE packing mode with packing threshold of 40 milliseconds

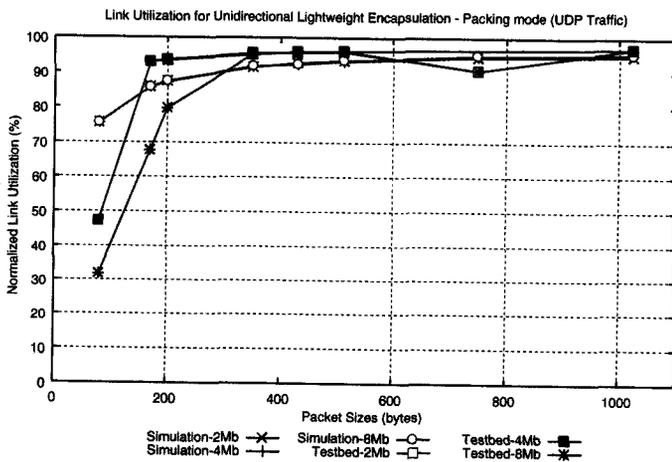


Fig. 11 Comparison of UDP link utilization using ULE packing mode with packing threshold of 40 milliseconds

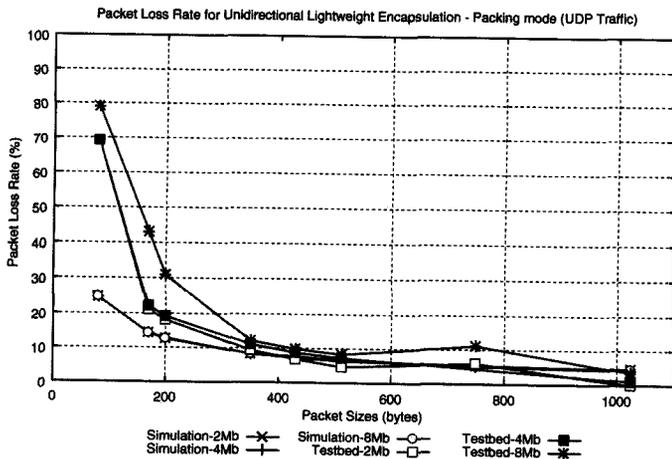


Fig. 12 Comparison of UDP packets drop rate using ULE packing mode

B. Latency tests

For DVB-S testbed, two simple UDP applications were specifically written to measure one-way latency of UDP packet over DVB-S link because ICMP can only measure round trip time normally. The clocks on the 2 UDL gateways were synchronized through the Network Time Protocol (NTP). The time difference between 2 gateways was measured using the same set of UDP applications over an Ethernet link. The results obtained through the DVB-S link were then adjusted using the previously measured time difference.

By comparing Fig 13 and 14, it is obvious that ULE padding mode offers lower average latency over ULE packing mode.

Sampled data from DVB-S testbed used to derive the average were inconsistent and have a surprisingly high latency. This can be attributed to the behaviour of DVB-S receiver cards and ASI card used on the testbed. In general, lower average latency can be obtained by using a higher symbol rate on the DVB-S modulator. In addition, the ASI link has a constant baud rate on its physical link. Thus, to achieve a lower data rate on an ASI link, more stuffing characters (K28.5 character) have to be inserted into the symbol stream. Therefore, real data have to compete with more stuffing characters contributing to the variation in the reported delay because the number of queued stuffing characters varies based on when data arrive at the ASI card.

There were no surprises in the *ns-2* simulator results in Fig. 13 and 14. Unlike the DVB-S testbed, one way latency measured on the *ns-2* simulator increased slowly as packet size increases. Naturally, to accommodate larger UDP packet, more MPEG-2 TS frames are required. Thus, a longer duration is required to send larger UDP packets.

Figs 13 and 14 show that link with higher bandwidth has lower latency. This effect is more obvious on the DVB-S testbed and can be explained by the characteristics of the ASI link mentioned above.

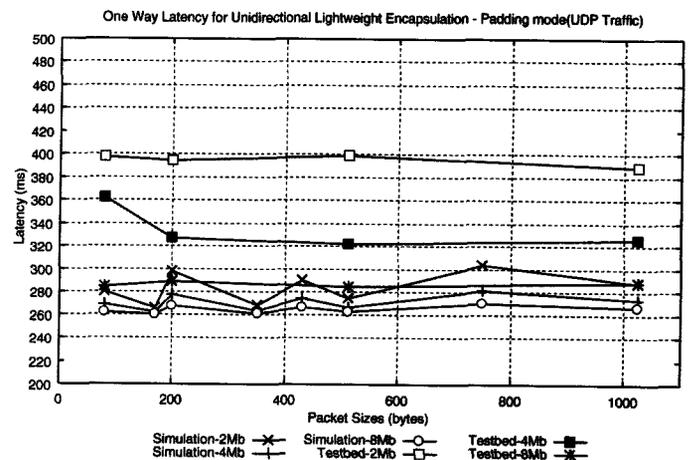


Fig. 13 Comparison of latency of UDP packets using ULE packing mode

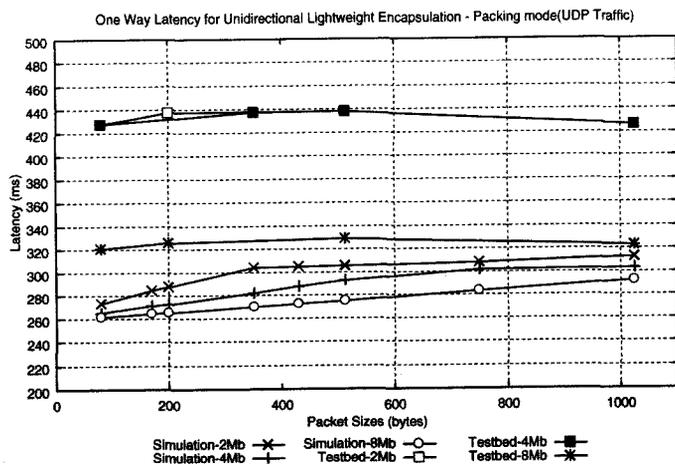


Fig. 14 Comparison of latency of UDP packets using ULE padding mode

VI. CONCLUSION

This paper discussed the characteristics of the new ULE encapsulation method. The results from both simulation and implementation were comparable. Differences arise due to the behaviour of the ASI card and DVB-S receiver card, whereas the simulator simplified the hardware characteristics and therefore could not capture the non-ideal behaviour of the actual equipments. TCP models need refinement to better reflect such issues. Nonetheless, the UDP behaviour of the ns-2 models was quite close to actual measured results.

ACKNOWLEDGMENT

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