Peer-reviewed version available at Appl. Syst. Innov. 2018, 1, 51; doi:10.3390/asi1040051

Type of the Paper (Article)

Interconnecting Haptic Interfaces Through The Internet

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Abstract: Supermedia streams transfer video, audio, haptic and other sensory data. Real -time transfering of supermedia streams over the Internet is quite challenging. This paper outlines the proposed protocols for transferring supermedia streams over the Internet. Moreover, it describes the Quality of Service (QoS) requirements for supermedia applications that a network has to fulfill. Extensive simulations and experiments for the performance evaluation of transport protocols for real time transferring HEVC streams with supermedia data are carried out. Complements, differences and relevancies between simulation and real world experiments are discussed. The metrics that are measured for the performance evaluation are delay, jitter, throughput, efficiency, packet loss and one proposed by the authors, packet arrival deviation. The simulation tests reveal which protocols could be used for the transfer of real-time supermedia data with a HEVC video stream.

Keywords: Supermedia, Haptics, HEVC, Tactile feedback, Transport protocols, Teleoperation, Interactive applications, Real-time Protocol, Internet Status.

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1. Introduction

This paper outlines the existing transport protocols of multimedia streams. Simulation and emulation tests for transferring supermedia streams over the Internet are undertaken. A thorough analysis of these results is presented.

Real-time supermedia streams transfer audio, video, graphics, haptics, smells and other sensory data. Supermedia data obtain massive variety and volume. This increase of data deteriorates the network status of the Internet. A promising solution to this is the new video encoding standard HEVC. It offers 50 % improvement in video compression over the existing H.264 Advanced Video Coding standard, keeping comparable image quality, at the expense of increased computational complexity [1].

A lot of research [2, 3, 4, 5] has been made on the QoS that a network should support, in order to have the maximum Quality of Experience (QoE) [6] in a supermedia application through the Internet. The network conditions in the Internet are not yet stable. They are changing from one area to another and from one hour to another. Internet network conditions mainly depend on the QoS that the Internet Service Provider enforces and the general state of the network. The recent network conditions of the Internet might permit supermedia applications to flourish.

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In order to transfer supermedia data through the Internet, specific transport protocols should be enforced. Several protocols have been developed for this reason. The most important are the ALPHAN Protocol [7], the SMOOTHED-SCTP [8], the ETP [9, 10], the IRTP [11], the RTP/I [12] and the RTNP [13]. Other protocols that are being widely used to transfer real-time multimedia data such as UDP, RTP [14], Datagram Congestion Control Protocol (DCCP) [15] and Stream Control Transmission Protocol (SCTP) [16] should be tested for supermedia transferring as well.

The rest of this paper is organized as follows. Section II outlines the most recent supermedia applications. Section III depicts the network conditions that are met in the Internet today. Section IV presents the simulation scenario that is used for the evaluation of transport protocols. Section V analyses the results of the simulation testing. Section VI discusses the complements, differences and relevancies between simulation and real world experiments. Finally, section VII concludes the paper.

2. Internet-Based Supermedia Applications

The expansion of Internet has led to the emergence of supermedia applications. Several interesting studies [17] have shown that the transfer of real-time supermedia applications through the Internet is possible. Several obstacles, such as delay and jitter, may still impede the flourishing of supermedia applications [18]. Apart from delay, the scaling factor in macro-micro teleoperations [19, 20] and the difference in inertia between the master and the slave system [21] can also deteriorate systems transparency [22]. Time-Delay compensation techniques [23, 24, 25] can overcome these barriers while Fuzzy Controller techniques [26, 27] can protect the haptic systems from failure when data transmission is insufficient. The high-computational-cost in Tele-Haptic applications can be moderated with a high-performance computing environment such as a computational grid [28].

Apart from teleoperations, where the kinesthetic part of haptics plays the major role, supermedia applications through the Internet can be applied to many other fields. Recent studies have shown that supermedia applications could be applied to military operations [29], education [30], telesurgery [31], video games [32], and video enhancement [33]. Furthermore, a motion-copying system (MCS) [34] can be useful for the digital preservation of motions by skilled experts as a haptic database. Supermedia can also enhance communication between people [35] and upgrade the virtual reality to a promising augmented reality [36].

Since supermedia refer to the many human senses, it follows that supermedia can benefit people with impairments. Haptic devices can help visually impaired people with route navigation and neighboring information [37, 38]. With the help of tactile sensors [39], impaired people can now visit a Haptic-museum from their home and explore all its exhibitions with the sense of touch. Haptic devices can help people with kinesthetic disabilities [40], but they can also improve the movement of humanoid robots like the haptic sensing foot system [41].

A new and promising opportunity for supermedia applications to flourish in the Web is HTML5. The HTML5 formal supports the java script based Web Graphic Library WebGL. The display of 3D graphics using a Web browser is now easier. A HTML5 Haptics (H5H) Plugin [42] runs on most popular web browsers, uses HAPI as a haptic rendering machine and supports most of the commercially available kinesthetic haptic interfaces. The window for easier creation of web haptic applications is now opening.

3. The Network Conditions of the Internet

A lot of research [43, 44, 45] has been conducted for the network conditions of the Internet. Network conditions refer to the amount of traffic that is being transferred through the Internet, the End to End delay as well as the jitter between source and destination, and the available bandwidth for data transport.

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The results from research [2, 3, 4, 5] have concluded that in order to maximize the Quality of Experience (QoE) of the user for supermedia streams, the network conditions should satisfy the Quality of Service (QoS) requirements of Table I.

TABLE I. QOS REQUIREMENTS FOR SUPERMEDIA STREAMS [2,3,4,5,6]

QOS	HAPTICS	VIDEO	AUDIO	GRAPHICS
JITTER (ms)	≤ 2	≤ 30	≤ 30	≤ 30
DELAY (ms)	≤ 50	≤ 400	≤ 150	≤ 100-300
PACKET LOSS (%)	≤ 10	≤ 1	≤ 1	≤ 10
UPDATE RATE (Hz)	≥ 1000	≥ 30	≥ 50	≥ 30
PACKET SIZE (bytes)	64-128	≤ MTU	160-320	192-5000
THROUGHPUT (kbps)	512-1024	25000 - 40000	64-128	45-1200

All the above metrics vary in time and space. They depend on the number of the online users, the amount of data that is being transferred at the specific moment of the measurement, and the available equipment of lines and routers. It has been recorded that the amount of data transferred through the web is constantly increasing [46]. Apart from that, the number of online users is increasing as well. The growth of data transfer is compensated by continuing infrastructure upgrades of computer networks.

There are two types of approaches for monitoring the network status. The two disciplines of network monitoring are the active and the passive measurements [43]. In the active measurement, specific generated probe packets, ICMP messages, are sent to specific destinations; measurements for delay, round trip time, jitter and packet loss are made. Some common diagnostic tools for active measurements are the ping, traceroute, capprobe, pathchar, netem and dummynet [44]. On the other hand, passive approach is based on the observation of the traffic that flows on the links. Some passive monitoring tools, commonly called sniffers, are the Tcpdump, Wireshark, Ethereal, Netflow and JFlow [43].

In order to monitor the network status, the authors actively measured the average and the standard deviation of the delay, the packet loss rate, and the number of hops of networks between countries and continents. Measurements for the above metrics were made by the authors between Japan and Korea, between Japan and Greece [47]. A recent measurement has also been made between two cities of Greece, Grevena and Thessaloniki. The distance between those two cities is 175 Km. Two different networks were used for this measurement, the private optical network, GRNET [48], part of the pan-European GEANT network with speeds up to 4x10Gbps, on the one hand and one simple 8 Mbps Adsl connection on the other. The results of these measurements are shown in Table II and III. For the above experiments, 3,000 ICMP packets for each of 0, 6, 11, 15 and 19 o' clock standard time were sent from one destination to the other.

It is understood that the Internet connection between Japan and Korea satisfy all the restrictions of Table I for transferring supermedia data through the Internet. For the Internet connection between Japan and Greece, the values of Table II are relatively high because they refer to intercontinental pings. The average delay exceeds the limit of Table I. This is due to the fact that the physical distance between Japan and Greece is much larger than Japan and Korea. That's why the number of hops is much bigger in this intercontinental connection.

TABLE II. INTERNET STATUS FOR INTERCONTINENTAL COMMMUNICATION[47]

COUNTRIES CONNECTED	AVG. DELAY (ms)	Standard DELAY Deviation(ms)	PACKET LOSS (%)	No. HOPS
JAPAN – KOREA	27.01	0.19	0.02	11
JAPAN – GREECE	331.10	6.30	1.53	26

TABLE III. INTERNET STATUS FOR COMMMUNICATION BETWEEN CITIES

CONNECTED CITIES	AVG. DELAY (ms)	Standard DELAY Deviation(ms)	PACKET LOSS (%)	No. HOPS
GREVENA – THESSALONIKH THROUGH GRNET [48]	*19.12*	*1.70*	*0*	*5*
GREVENA – THESSALONIKH THROUGH ADSL LINE	*53.19*	*5.31*	*0.11*	*8*

For the Internet connection between the two cities of Greece the results are shown in table III. In the case of the simple 8 Mbps Adsl connection the results are slightly worse than the limits in Table I for the average delay and jitter. The packet loss is within the limits of table I. On the other hand in the case of the private optical network, GRNET, the results are encouraging. The average delay is only 19.12 ms , the jitter is only 1.70 ms and packet loss is 0.00 %. All the above results are much lower than the limits in Table I, which means that the transport of supermedia data through the Internet is feasible under some circumstances

Another important factor that describes the network conditions of the Internet is the connection speed of the end user. The recent spread of ADSL and VDSL connections provide consumers with connections up to 50 Mbps bandwidth. This bandwidth is by very sufficient for the requirement of 1 Mbps throughput that is being produced from haptic applications, based on Table I.

We can conclude that the Internet network conditions are now suitable for supermedia applications, especially when these applications take place in near regions.

4. Simulation Scenario of Existing Protocols

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In order to monitor the metrics of section IV and evaluate the transport protocols of section III, simulation tests were carried out. The network simulator that has been chosen for these tests is the Network Simulator 2 (NS2) [49]. It is an open source, discrete event simulator with substantial support for protocol evaluation over wired and wireless networks. A lot of common protocols have already been implemented and tested in the NS2. Unfortunately, not many supermedia protocols have been applied in NS2 so far.

One supermedia protocol that has been implemented in NS2 is the ETP. Apart from that, a lot of real-time protocols that could be used for supermedia applications have been applied in the NS2. Some of them are the RTP, the SCTP, the DCCP and the UDP.

All the above protocols were attached to different nodes in the NS2 as shown in Figure 1. The protocol TCP was mainly applied to the simulation scenario as a traffic generator. All the other protocols try to send a stream of packets with a packet rate of 1000 packets per sec. Most of them have a congestion control algorithm and minimize their sending rate in case of congestion. The haptic packet size that every protocol sends is 64 bytes of data payload [11] plus the overhead of the protocol.

The sample video for the HEVC encoding was the mobile_cif YUV series [50] with 352×288 resolution at 24 Hz. The data rate of this video sample after the HEVC encoding with Quantization Parameter QP= 27 and Low-Delay inter-prediction is 642 kbps [51]. This data stream sent over the RTP protocol with packet size of 1500 bytes.

The audio stream was sent over the RTP protocol with 128 kbps bit rate, packet size 320 bytes and sending rate 50 packets per sec.

A. Static Network Bandwidth, Delay and Internet Traffic

The square nodes in Figure 1 are routers that are connected with each other through the Internet.

The connection speed between those routers is set to 1, 5, 10, 15 and 20 Mbps for each simulation and is stable for the whole simulation period. The Internet bandwidth of 1 Mbps has been chosen so that a fully congested network can be represented. The Internet bandwidth of 20 Mbps corresponds to a network with no congestion. The Internet bandwidth of 5 Mbps corresponds to a network with low congestion. The connection between the server nodes 6 and 7 is regarded to be the Internet bottleneck of the simulation.

The end to end delay in connection between nodes 6 and 7 was set to 5, 10, 20, 30, 40, 50 and 60 ms for each simulation and was stable for the whole simulation period. The 5 ms delay is a very small delay that rarely occurs in Internet connections. On the other hand, 50 ms is regarded to be the upper tolerable limit of delay, based on Table I, that's why the 60 ms is set as the maximum delay of the simulations. Of course, the end to end delay is changing dynamically in the real world connections. The authors deliberately kept the end to end delay constant throughout each simulation so as to study the behavior of each protocol at the specific delays. This would help the researchers decide which protocols are preferable when the characteristics of the network delay are known. Most of the diagrams depicted in this paper are for an Internet delay of 40 ms which is an acceptable delay, based on Table I, and commonly encountered in the network. The simulations tests undertaken were 5 (scales for the Internet speed) X 7 (scales for the delay) = for a total of 35 tests.

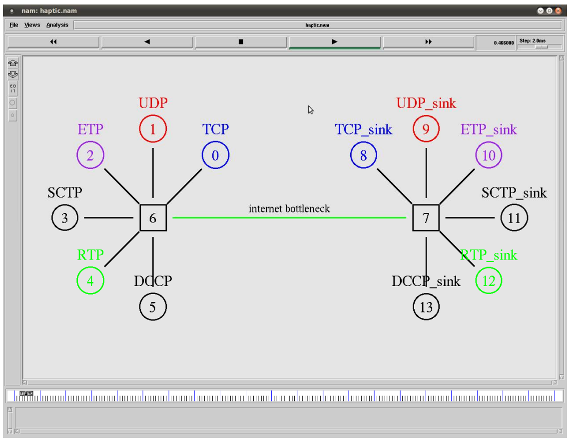

Figure 1. Simulation Environment of NS2.

The connection speed between the nodes 0-5 and server 6 as well as nodes 8-13 and server 7 is 100 Mbps as they are considered to be in the same local area network. The delay in those connections was 1 ms.

The simulation time for each simulation was 20 sec. At time 0.5 sec the FTP application which was attached at node 0 started to send data. At time 2 sec all the other CBR applications which were attached to nodes 1-5 will started to send packets with a rate of 1000 packets per sec. The packet size varied from node to node depending on the header of the transport protocol.

B. Dynamic Network Bandwidth, Delay and Internet Traffic

An interesting case of study is to examine the behavior of all the above protocols in a dynamic environment such as the Internet. In such an environment, the delay of the network, the available bandwidth and the packet loss are constantly changing. In order to simulate such an environment, the DelayBox [52] and the TMIX [53] modules have been added to the ns2 simulator. With the help of these modules, realistic Internet traffic is being fed to the network through the inbound (node 14) and outbound (node 17) initiators of figure 2. The delaybox nodes on the other hand, enforce a variable bottleneck of 1-20 Mbps, a variable packet delay of 1-20 ms and a packet loss of 0-1% on all the TCP packets that pass through them. As the TCP packets dynamically change their behavior, all the UDP packets are affected as well.

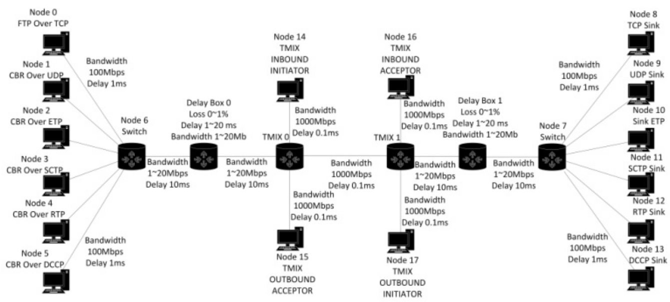

Figure 2. Topology of NS2 with DelayBoxes and TMIX traffic.

5. Results and Analysis

A. Protocol Efficiency

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Protocol Efficiency was one of the metrics to be analyzed. Protocol efficiency is uninfluenced by the network status and is determined by the payload of each application and the header of each protocol Eq. (1). As supermedia applications demand very high update rate, the efficiency of the transport protocol is crucial. The small payload should not be overshadowed by big overheads of the transport protocols.

Table IV shows the efficiency of the five protocols under test. The protocol with the highest efficiency was, as expected, the UDP protocol. This result derives from the fact that the UDP is a "best effort", unreliable protocol with no congestion control and no packet sorting. Therefore, UDP has the smallest header of only 8 bytes.

TABLE IV. EFFICIENCY OF TRANSPORT PROTOCOLS

	ETP	UDP	RTP	SCTP	DCCP
HEADER (bytes)	12+8(UDP)	8	12+8(UDP)	12+4 (CHUNK INF.)	12
HAPTIC PAYLOAD (bytes)	64	64	64	64	64
EFFICIENCY	76.19%	88.88%	76.19%	80%	84.21%

B. Packet Loss

Figures 3 and 4 depict the percentage of packet loss with relation to the delay of the network. This diagram is important as it reveals the correlation between the network delay and the packet loss for each protocol. If the characteristics of the End-to-End delay of the network are known, it can be decided which protocol should be used for the transport of supermedia data so as to avoid high values of packet loss.

In figure 3 the Internet bottleneck is 20 Mbps for all simulations. The delay was set to 5, 10, 20, 30, 40, 50 and 60 ms for each simulation and was stable for the whole simulation period. All the protocols present quite a low packet loss, lower from the limit of 10 % of Table I. The worst performance is presented by protocol the SCTP and the DCCP but still they have a packet loss lower than 0.76 %.

On the other hand, Figure 4 presents much higher values of packet loss. In this scenario, the Internet bottleneck has only 1 Mbps bandwidth for all simulations. It is obvious that the network is congested. Six protocols are trying to send a throughput of at least 3 Mbps over the network with bandwidth of 1 Mbps. The protocol with the smallest percentage of packet loss is the UDP protocol. The protocol with the higher packet loss is again the SCTP and the DCCP protocol.

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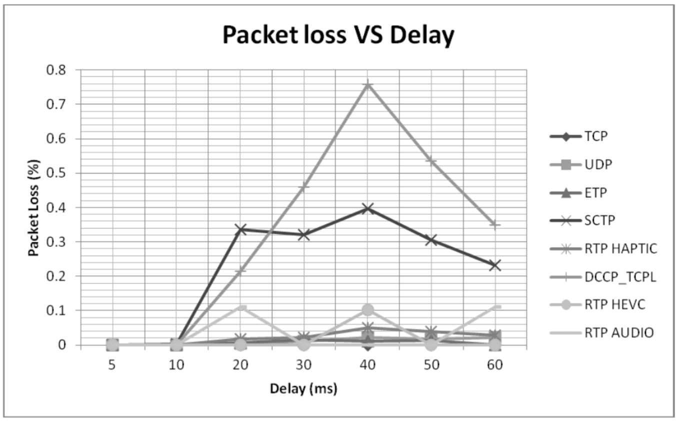

Figure 3. Packet Loss for Internet Bandwidth 20 Mbps.

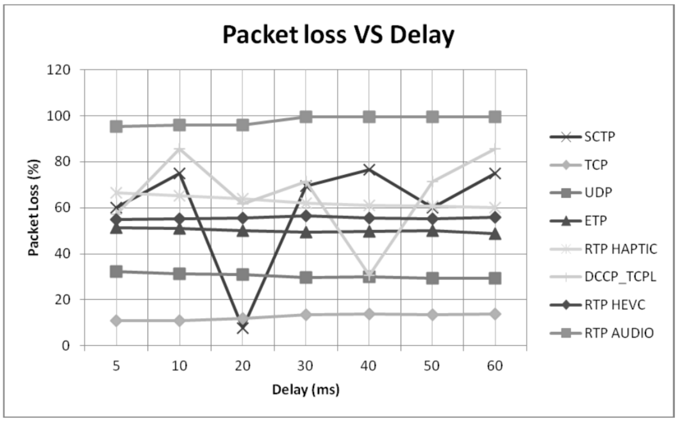

Figure 4. Packet Loss for Internet Bandwidth 1 Mbps.

Figures 5 and 6 depict the correlation between the packet loss and the available bandwidth of the Internet. Figure 5 depicts a network with static network conditions while Figure 6 depicts a network with dynamic delay and Internet bandwidth for the TCP packets. Both charts have similar behavior for Internet bandwidth higher than 5 Mbps, where no significant congestion is occurred. From these charts it can be derived which protocols behave better in a congested network. It is quite obvious that as the Internet bandwidth increases, the packet loss decreases. For Internet bandwidth higher than 5 Mbps the packet loss is lower than 1 %. This means that there is no congestion on the network for available bandwidth higher than 5 Mbps. When the available Internet bandwidth is only 1 Mbps, the network is heavily congested and the packet losses are unacceptable high. The worst performance regarding packet loss, based on figures 3, 4, 5 is being presented by the DCCP and the SCTP protocol. This performance is being deteriorated especially in heavy congested networks. The congestion control algorithm of the SCTP and the DCCP protocol is a TCP-like Congestion Control, which is similar to that of TCP. The sender maintains a congestion window and sends packets until that window is full. The response to congestion is to halve the congestion window. This means that the DCCP and the SCTP protocol sends its packet in bursts. This cause the buffers of intermediate routers to overflow and some packets to be dropped.

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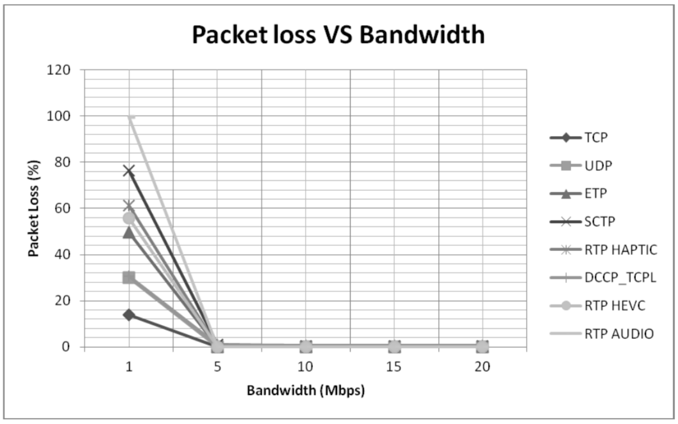

Figure 5. Packet Loss for Internet Delay 40 ms.

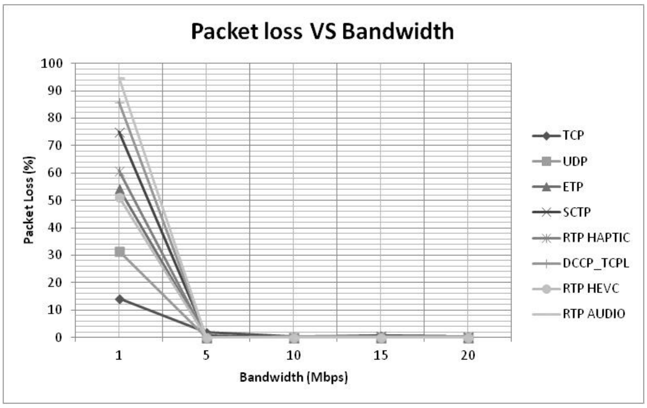

Figure 6. Packet Loss Vs Bandwidth for Dynamic Network Conditions.

C. Throughput

One more metric that has to be monitored is the throughput that every protocol loads to the network. The higher the throughput is, the higher the possibilities are for network congestion. In connections where the available bandwidth is low, protocols with small throughput should be preferred.

Figure 7 depicts the throughput of the protocols for an Internet bandwidth of 20 Mbps, so that no congestion should occur in the network. The delay of the Internet is set to 40 ms, as 50 ms is the maximum accepted delay, based on Table I. TCP throughput varies between 1660 Kbps and 3300 Kbps because of the congestion window of the TCP's congestion control. UDP and RTP protocols pose a steady throughput of 576 and 672 Kbps for haptic data respectively. This means that their sending rate is constant. The higher throughput of RTP is due to the higher header of the protocol. Protocol ETP tries to reach its highest sending rate, but its growth is very slow due to its congestion control. After 20 sec of simulation time it had not yet reached the sending rate of 1000 packets per sec. The SCTP protocol presents the highest throughput, after TCP, among the other protocols. Apart from that, it also presents the highest deviation of the throughput. DCCP for the first 6 sec presents quite a big deviation of the throughput and it is not stabilized before the 8th second.

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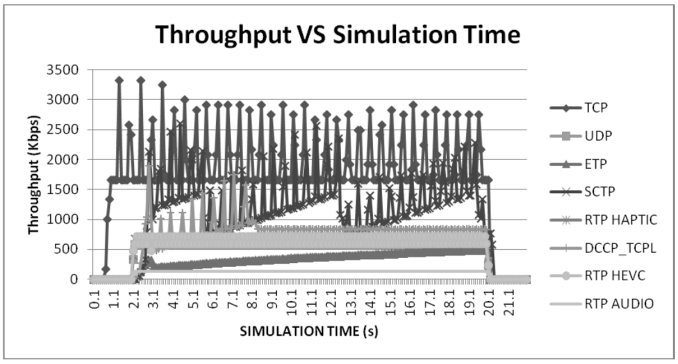

Figure 7. Throughput for Internet Bandwidth 20 Mbps and Delay 40 ms.

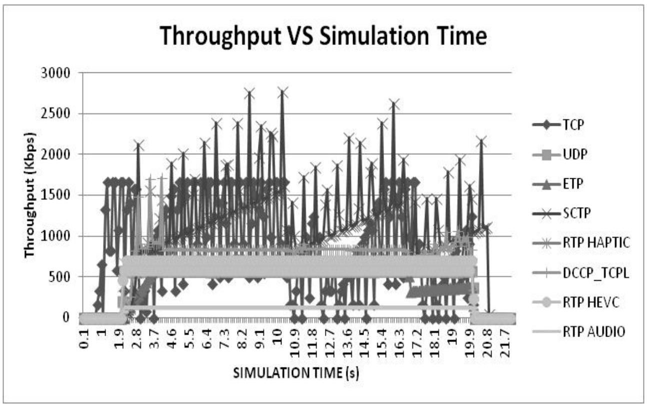

Figure 8. Throughput Vs Bandwidth for Dynamic Network Conditions.

Figure 8 depicts the behavior of the protocols for a dynamic network. The big different between figure 7 and 8 is the behavior of the TCP, the SCTP, the ETP and the DCCP protocol. All the above protocols have a tcp friendly congestion control protocol. As the available bandwidth dynamical changes, the above protocol change their transmission rate in order congestion to be avoided. On the other hand the UDP and the RTP protocol exhibit the same performance because they don't have a tcp friendly congestion control.

Figure 9 displays the throughput of the protocols for Internet bandwidth 5 Mbps and delay 40 ms. The bandwidth of 5 Mbps is chosen so that the network is under low congestion. TCP has lowered its throughput that now varies from 1660 Kbps to 2160 Kbps. UDP and RTP protocols display almost the same steady throughput with very small deviations as they do not have a TCP-Friendly rate control. ETP protocol adapts the smallest throughput. It cannot increase its sending rate because there is some congestion on the network. It can be seen that the congestion control of TCP binds more bandwidth than that of ETP. DCCP shows almost the same performance with the previous simulation. It adapts almost the same steady throughput, with very small deviations, 2 seconds later than in the previous simulation. SCTP still presents the biggest deviation and does not manage to obtain a steady throughput.

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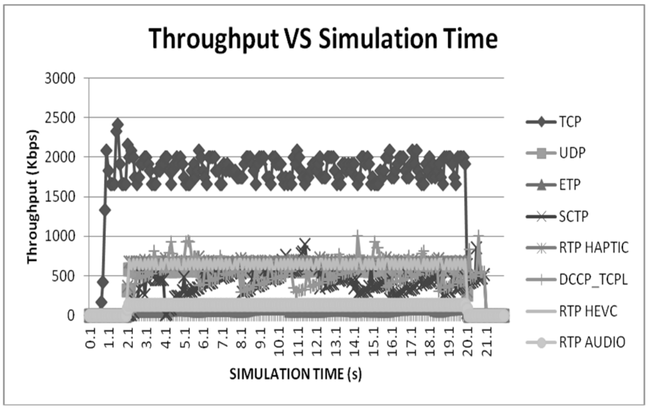

Figure 9. Throughput for Internet Bandwidth 5 Mbps and Delay 40ms.

D. Jitter

One crucial metric that has to be monitored is the packet delay deviation. A high value of jitter is a crucial factor that often leads to system instability and failure. Based on Table I haptic applications have the lower tolerant limit of jitter of all multimedia applications. The jitter effect occurs when there is congestion on the network.

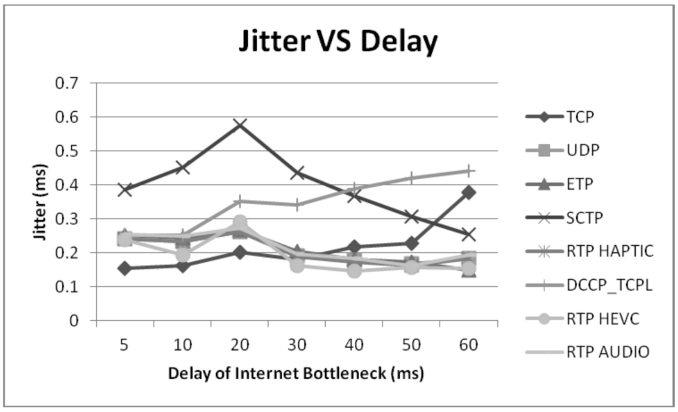

Figure 10. Jitter for Internet Bandwidth 20 Mbps.

Figure 10 illustrates the jitter of the protocols for an Internet bandwidth of 20 Mbps and for delays from 5 ms to 60 ms. At this high bandwidth no congestion occurs. Almost all of the protocols show very small jitter, lower than 0.6 ms.

Figure 11 depicts the jitter of the protocols for an Internet bandwidth of 1 Mbps. It is a fully congested network with a lot of packets lost. The jitter is now much higher than in figure 10. The protocols with the highest jitter are the DCCP, the TCP and the SCTP. RTP, UDP and ETP protocol have smaller jitter than 6.7 ms, with the ETP exhibiting the best performance with a jitter smaller than 2.6 ms. According to Table I, the jitter should be lower than 2 ms, a goal that most of the protocols could not achieve due to the congestion of the network. An interesting observation is that the protocols TCP, SCTP, and DCCP lower their jitter as the delay of the Internet bandwidth increases. All these protocols have the almost the same tcp-like congestion control algorithm.

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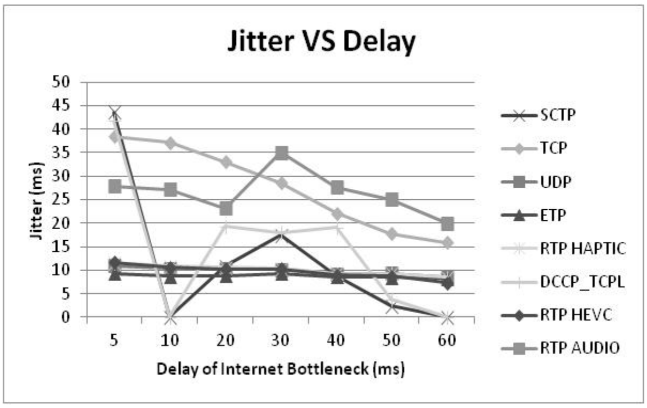

Figure 11. Jitter for Internet Bandwidth 1 Mbps.

Figures 12 and 13 depict the jitter of the protocols when the Internet bottleneck bandwidth varies from 1 Mbps to 20 Mbps. The average Internet delay is near 40 ms. All the protocols show a different behavior when the Internet bandwidth is 1 Mbps and the Network is heavily congested. For higher bandwidth values all protocols present almost the same behavior. As the Internet bandwidth increases, the congestion and as a consequence the jitter effect, are decreasing.

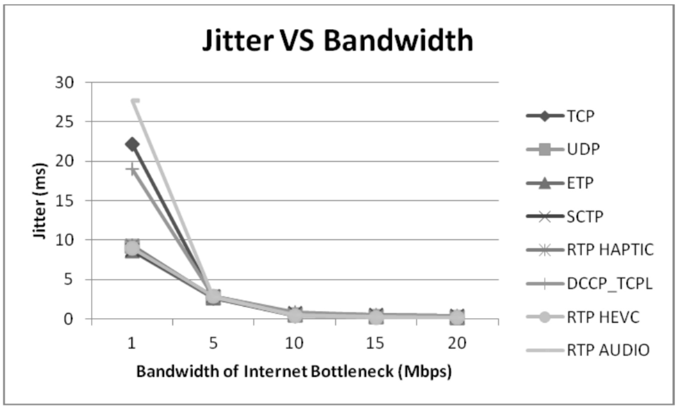

Figure 12. Jitter for Internet Delay 40 ms.

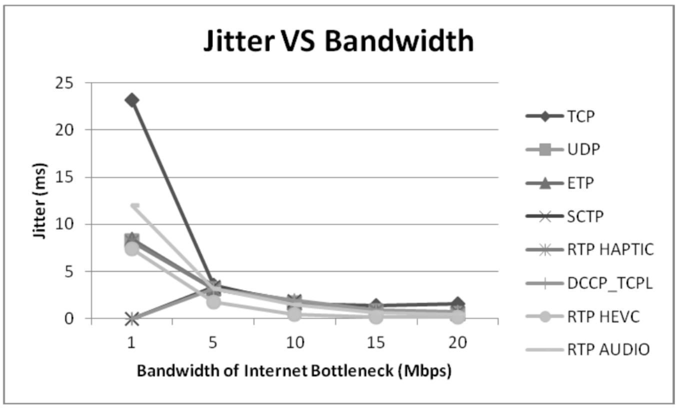

Figure 13. Jitter Vs Bandwidth for Dynamic Network Conditions.

E. Packet Arrival Deviation

Packet Arrival Deviation (PAD) is the proposed metric for the performance evaluation of supermedia protocols. It shows similar behavior with jitter but it can offer more precise picture of the real-time network delay conditions than jitter, since it can take into account both the receiver end for the time variation of received packets (packet reception jitter at the receiver end), as well as the

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ACK packets reception time variation at the sender (packet reception jitter at the sender end). Moreover, it takes also into account the changes of the sending rate of the source focusing on the fluctuations of the Internet bandwidth. Table I does not include the PAD, as it is only presently proposed by the authors. The upper tolerant limit for PAD should be equal with jitter's, meaning 2 ms, as they should show almost identical behavior when there is no packet loss and no changes in the sending rate and the Internet bandwidth.

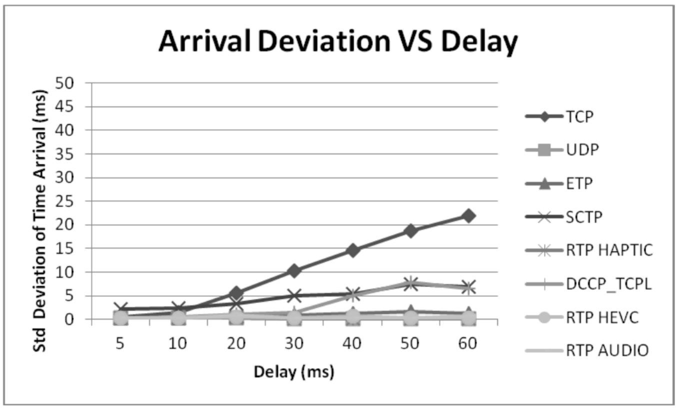

Figure 14. Packet Arrival Deviation for Internet Bandwidth 20 Mbps.

Figure 14 displays the standard deviation of the packet arrival for different delays. The Internet bandwidth is 20 Mbps. The protocols with the higher standard deviation are the TCP, SCTP and the DCCP protocols. UDP and RTP and ETP protocols present very small standard deviation of packet arrival, lower than 2,5 ms. This difference between TCP,SCTP, DCCP and the UDP, ETP,RTP is the way they send their packets. The first group sends its packets in burst inside a congestion window (CWND), while the second group sends its packets with an almost steady inter packet gap.

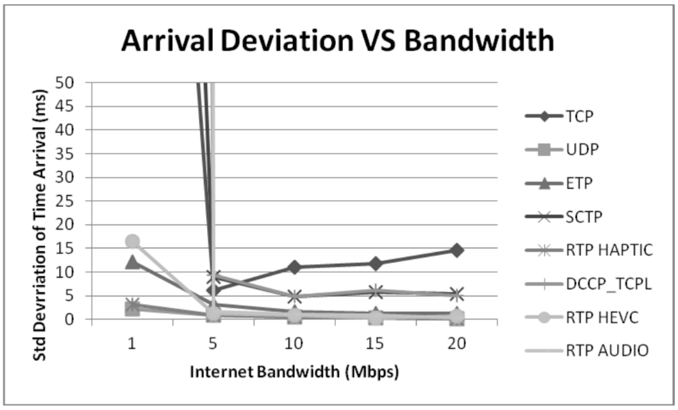

Figure 15. Arrival Deviation for Internet Delay 40 ms.

Figure 15 and 16 shows the Packet Arrival Deviation as the bandwidth of the Internet varies from 1 Mbps to 20 Mbps. Protocols SCTP and DCCP and TCP are not included in the graph for 1 Mbps because they could not perform at such a congested network. The conclusions of this graph are similar to those of Figure 14. The protocol with the highest standard deviation is the TCP, because it sends its packets in bursts. UDP and RTP have almost none arrival deviation as they send their packets with constant bit rate with no congestion control. When the Internet bandwidth gets values higher than 10 Mbps, the performance of all the protocols, except TCP, resembles as no congestion occurs.

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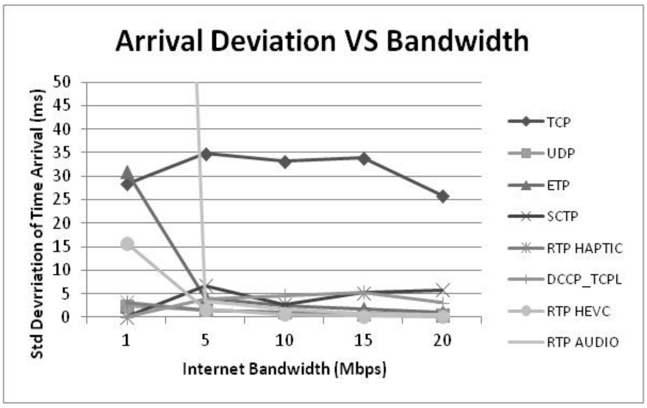

Figure 16. Arrival Deviation Vs Bandwidth for Dynamic Network Conditions.

Analyzing the above results, it is understood that not all multimedia protocols are suitable for transferring supermedia data. Protocols such as TCP, SCTP and DCCP do not perform very well in heavily congested networks, as they are not designed for timely delivery of information. The most stable protocols for real-time data are UDP, ETP and RTP. Both ETP and RTP protocols are based on UDP. The UDP protocol is the lightest, fastest and most efficient protocol from all. Table IV shows that UDP has 88.88% efficiency for a payload of 64 bytes. As far as packet loss is concerned, all protocols show similar behavior, as shown in figure 5. The only case that they behave differently is when the Internet bandwidth is only 1 Mbps (Figure 4), which means that the network is under unacceptable heavy congestion. As far as throughput is concerned, Figures 7 and 9 which correspond to 20 and 5 Mbps Internet bandwidth with 40 ms delay, UDP and RTP show a steady and similar behavior with UDP consuming little less bandwidth due to its better efficiency. ETP protocol performs rather well in good network conditions but it can't reach the optimum sending rate of 1000 packets/sec when a little congestion occurs. Regarding jitter and Arrival Deviation UDP and RTP show almost the same behavior in all network conditions. Their behavior is better than all the other protocols. The only case where ETP is showing a little better behavior than UDP and RTP is when the Internet bandwidth is only 1 Mbps (Figures 11, 12), which means that the network is unacceptable heavily congested. To avoid congestion, ETP is lowering its sending rate, and that's why it presents lower jitter and arrival deviation. Summarizing, it could be said that RTP and UDP present the best performance with similar behavior in most network conditions. The ETP protocol could be used in cases with network congestion, due to its congestion control.

6. Complements, Differences and Relevancies Between Simulation and Real World Experiments

In section VI, our previous real world experiment is described, while in section VII and VIII a simulation experiment has taken place. Anyone would have thought that a real world experiment would give more accurate results than a simulation test and a simulation test is unnecessary. The truth is somewhere in the middle. The simulation tests do not contradict real word experiments but they complement them.

The real world scenario helped us understand the status of the Internet and define the values of the variables for the simulation experiment. The only variable that is difficult to define is the available bandwidth of the Internet for the whole path. It is a metric that is being changing dynamically and rapidly as it is based on the number of the online users and the data that are exchanged.

The real world scenario revealed that a real teleoperation task through the Internet is feasible, while the simulation experiment helped us conclude which protocols suits better for these teleoperation tasks under specific network conditions.

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In the real world experiment, it is understood that the experiment results are depending on the physically distance between the source and the destination and the ambiguous network conditions of the Internet. In the simulation tests network conditions are fully controlled, so more accurate results can be produced.

The real world experiment helped the authors define the mean end to end delay, the standard delay deviation (jitter) the packet loss and the number of hops between source and destination. For this experiment the UDP transport protocol was chosen. It is the simplest transport protocol and it is being used for most cases of real-time multimedia applications. Two completely different pairs of source and destination were chosen, in order to examine the dependency between the results on the distance between source and destination.

Despite the ambiguous network condition of the Internet the simulation results of the UDP protocol matches the results of the real world scenario for the connection between Korea and Japan, Table V. At the specific simulation, the Internet end to end delay was set 32 ms. The Internet bandwidth was set to 20 Mbps. These settings depict a network with no congestion. Both simulation and experiment transport data over the UDP protocol. For the connection between Japan and Greece there were no corresponding simulation tests as the results from the real world experiment were outside the acceptable limits of Table I.

TABLE V. Similarities Between Simulation And Real World Experiments

	DELAY (ms)	JITTER (ms)	PACKET LOSS (%)
CONNECTION BETWEEN JAPAN – KOREA	27.01	0.19	0.02
SIMULATION OF UDP PROTOCOL WITH 20 MBPS INTERNET BANDWIDTH	32.10	0.19	0.02

7. Conclusions

It is obvious that the transfer of real-time supermedia data through the Internet is now possible. The network conditions of the Internet are continuously improving. The transport protocols that are being used for the transfer of the multimedia real-time data are not specialized in transferring supermedia streams with haptic data. Specific protocols for supermedia transferring should be standardized. The Internet Service Providers should integrate the network conditions that are required for the transfer of haptic data in their QoS.

The new HEVC video encoding is offering great improvements in the supermedia transferring through Internet. The reduced by half bit rate of the video stream provides congestion avoidance and reduced jitter, delay and packet loss.

The experiments that have been carried out in this paper revealed which protocols could be used for the transfer of real-time supermedia data.

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