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Evaluation of New Dynamic Time Division Multiplexing Protocol for Real-Time Traffic over Converged Networks

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***Abstract*—Packet switched networks have been increasingly used for the transmission of real-time traffic in addition to the traditional best-effort. Although not primarily designed for time critical traffic, industries like professional audio and video, electrical utilities and automation systems, motivated the need for having converged networks that support both time critical as well as best-effort traffic. In this paper a new proposed dynamic Time Division Multiplexing (TDM) protocol,Flexilink,is evaluated for its ability to provide a low latency unified network infrastructure for both real-time traffic as well as traditional best-effort traffic.A simulation model had been developed to evaluate Flexilink performance,however, no research evaluation had been done based on hardware implementation of the protocol.We present a test-bed based study and evaluation for the performance of Flexilink. Test-bed results shows that Flexilink has a better performance compared to other multimedia networking solutions mainly in providing extremely low end-to-end latency and jitter that is well below the audible threshold which makes Flexilink suitable to be used in live multimedia networks.**

1. Introduction

In recent years, using packet switched networks for transmission of multimedia traffic has been popular in industries such as professional audio/video, broadcasting industry as well as other multimedia applications in vehicular and avionics industries. These networks need to be carefully designed and managed in order to ensure maintaining the quality for the multimedia in terms of providing bounds on latency,having the ability to preserve the timing relationship between packets of the same flow (low jitter) as well as high reliability [1]. Although using the same network infrastructure to support both conventional best-effort traffic as well as real-time multimedia traffic has many benefits in terms of cost reduction and ease of monitoring and maintenance, yet the common practice was to have these types of traffic segregated in professional environments either logically or physically by using dedicated network devices [2]. This is due to the fact that in professional environments, multimedia traffic has rigorous constraints on jitter, latency and synchronisation which can be hindered by having best-effort traffic on the same network.

Most protocols designed to support real-time multimedia traffic on packet switched networks rely on Ethernet as the underlying infrastructure due to its availability and widespread adoption. However, Ethernet was not traditionally designed to support real-time traffic, and thus not capable of supporting the level of guarantee on latency and jitter required to ensure the quality of real-time multimedia traffic. These guarantees are usually referred to as Quality of Service (QoS) of the network.Different protocols were then added in order to provide the necessary timing characteristics for the existing Ethernet infrastructure. These protocols, however, have their own limitations which in turn cause various quality issues specifically for multimedia traffic. Glitches and gaps are very common in Voice-over-IP (VoIP) applications. Video-on-Demand (VoD) and Video Conferencing applications also suffer from quality issues like long startup delays, re-buffering and sometimes failure to start videos. These quality issues lead to poor users' Quality of Experience (QoE), a term used as a quality indicator in multimedia applications in which the quality of the received media is expressed as perceived by the user. Thus these solutions and protocols, appear to be only suitable for entertainment purposes and not used in environments where professional guaranteed quality of audio and video is required,such as in the case of broadcasting industries.Another challenge in using packet switched networks for sending real-time traffic is jitter. Real-time systems such as audio playback systems for example require an extremely low jitter, in the order of nanoseconds to avoid being audible [3]. In the current used real-time systems this is mostly achieved through using de-jitter buffers which in turns adds to the overall latency and system complexity. In this paper, the Flexilink protocol is discussed and evaluated in terms of providing end-to-end low latency and low jitter guarantees for real-time traffic even in the presence of variable load best-effort traffic.Flexilink is presented as an alternative professional media infrastructure which had been reported by the European Broadcasting Union (EBU) in the Joint Taskforce for Networked Media (JT-NM) report as a potential candidate for Future Network (FN) infrastructure [4]. The Flexilink architecture [5] provides a mechanism to achieve not only guaranteed quality for professional live multimedia comparable to this achieved by TDM systems, but also a potential true converged solution for different traf-

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fic types including best effort data [2]. The rest of the paper is organised as follows. Section 2 gives a brief overview over existing protocols and solutions used to support multimedia traffic over packet-switched networks.Section 3 explains the rationale behind the design of Flexilink architecture. Section 4 describes the experiments used to evaluate Flexilink performance followed by discussion of the obtained results. Section 5 concludes the paper and discusses future work.

2. Multimedia and Converged networks

2.1. Multimedia over Packet Switched Networks

Sending multimedia (audio and video traffic) involves sending and consuming traffic in real time. This require the network to have an end-to-end low latency (delay) as well as low jitter which is defined as the variation in the experienced delay of audio packets over the network [6].However, since packet switched networks were not primarily designed to provide timing characteristics, different protocols were then added to support real-time traffic. Added protocols included Real-Time Transport Protocol (RTP) [7],Integrated Services Model (IntServ) [8] and Differentiated Services Model(DiffServ) [9]. Adding the RTP protocol to the existing network infrastructure enabled the support of real-time audio and video data that required bounded delay. This made the new structure suitable for applications like VoIP, VoD and video conferencing. However, quality issues such as audio glitches, poor video quality and long startup delays still existed with the new network structure. IntSev model uses Resource Reservation Protocol (RSVP) per flow to guarantee resources availability,e.g. banwidth, which will in turn guarantee flows quality. However,per-flow reservation requires maintaining a lot of flow status information to be stored in network devices along the reserved path, which makes using IntServ model complex and unscalable [10]. On the other hand,DiffServ model classify the traffic into classes and associates different forwarding node behavior to different classes [9]. Real time audio and video streams are given higher priority over traditional best effort traffic that are not time critical. This approach, although does not require information to be stored at network nodes and thus is considered scalable yet does not guarantees real-time traffic performance for example in case of network congestion.This in turn made industries that requires guarantees on audio and video quality like professional audio and video broadcasting use different networks for transmission of multimedia and best-effort traffic to avoid quality degradation in case of network congestion.

2.2. Multimedia over Converged Networks

The demand for having a unified (converged) network infrastructure has been greatly encouraged by the broadcasting industry in their attempt to reduce the cost associated with the upgrading and maintenance of radio stations specialized equipment. Having a converged network can enable radio stations to have their infrastructure equipment moved to a central shared location, where the actual audio files will be stored, streamed, mixed and processed in this remote data center, before being sent to the local radio station along with other best-effort traffic such as social media tools, images, word files among others. This centralisation and convergence of different types of traffic requires the presence of an extremely stable and reliable unified network infrastructure. In 2011, the Audio/Video Bridging (AVB) task group grouped several IEEE standards together in order to achieve the QoS requirements for low latency streaming in Ethernet networks This task group had later changed its name to Time Sensitive Networks (TSN) to include the wide use of deterministic networks and not just the case of professional audio and video [11]. Standards included the IEEE 802.1AS time synchronization protocol, the IEEE 802.1Qav forwarding and queuing protocol and the IEEE 802.1 Qat signaling protocol. This new suite of protocols is known as the IEEE 802.1 AVB protocol. AVB protocol provided convergence between time-synchronized low latency streaming services and best-effort traffic by using diverging media access and prioritization techniques [12] Performance analysis using simulation approach for AVB protocol was carried out using OMNET++ event-based simulation platform. Results of the simulation showed that real time applications constraints, in terms of the maximum allowed latency and synchronization accuracy, over seven hops of AVB streaming data are guaranteed even in high network load situations [13]. However, results also showed that the payload size of the AVB frames has an influence over the performance where using small AVB frame sizes improved the end-to-end latency compared to using bigger frame sizes. Another extension of AVB was introduced by AVB L3 Transport Working group (P1733) where they published the IEEE Standard for Layer 3 Transport Protocol for Time-Sensitive Applications in Local Area Networks []. This standard achieves interoperability between Real Time Transport Protocol (RTP) and the IEEE802.1 AVB protocol thus extending the operation of AVB in transmission of time-sensitive applications to Wide Area Networks (WANs) instead of being limited to bridged LANs. In 2016, the IETF working group Deterministic Networking (DetNet) was formed [1]. The main focus of the group is to introduce a model for forwarding real-time traffic (traffic with timing constraints in general also referred to as deterministic traffic) beyond LAN boundaries.The proposed Detnet architecture classifies traffic according to their requested service level into three categories

- *Synchronous Flow* all links are closely synchronised (less than 1 μsecond) which allows for fixed latency where rescources, such as buffers and links' bandwidth can be shared over the time domain among different flows.
- *Asynchronous Flow* links are not synchronised and thus flows do not have fixed schedule. However, latency is bounded through limiting the bit rate of the admitted flow.

- *Non-DetNet Flow* similar to traditional best-effort traffic where no guarantees on latency or delivery of the flow packets.

The proposed model will enable fully scheduled operation controlled by a central controller to guarantee the timing constraints for deterministic traffic without compromising the ability of the network to carry traditional best-effort traffic.

3. New Flexible TDM - Flexilink

3.1. Background

Sending multimedia traffic over packet switched networks involve reserving and sharing of resources for session duration in order to provide the required QoS.Network resources e.g. bandwidth is shared among multiplexed traffic flows. The amount of reserved bandwidth is determined using traffic models to predict incoming flows characteristics e.g. maximum frame size, peak and average rate. Since different flows have different characteristics, statistical multiplexing is used to reserve bandwidth that is typically less than the sum of peak rates of the incoming flows and close to the sum of the flows average bit rates [14]. Using statistical multiplexing, however can result in packets loss if the actual bit rates of the incoming flows are higher than the reserved bandwidth. Reserving bandwidth based on the actual bit rate of the incoming flows will in turn mean decreasing packet loss probability while achieving full bandwidth utilistaion . This is referred to as deterministic multiplexing. In 2012, Flexilink was introduced as a unified low latency network architecture that supports both multichannel audio and video (AV) streams as well as traditional best-effort traffic. Flexilink uses dynamic TDM for pre-allocation of time slots for incoming AV flows using deterministic multiplexing. This approach guarantees the QoS of the incoming AV streams in terms of low end-to-end delay and packet loss while achieving nearly full bandwidth utilisation [15].

3.2. Flexilink Protocol

Flexilink uses pre-allocated time slots to guarantee deterministic transmission for AV packets.By pre-allocating time slots, queuing delays are eliminated as the number of allocated time slots for each flow can be determined based on each flow sending bit rate.In Flexilink, each packet sent is part of a ’flow’. Flexilink separates packets transmission into two planes, control and data. The control plane is responsible for maintaining flows and routing information and is implemented in software.This in turn allows for simplifying packets by taking away headers that were mainly used for routing,for example, source and destination addresses.This helps reducing packets sizes which further simplifies packets forwarding in switches [16].Used packets headers only contain identification information about the flow that this packet belongs to and the packet length.Packets (header and payload) are carried in the data plane and are being forwarded at different nodes without being inspected or altered.

Flexilink defines two categories of services referred to as AV and IT. AV service offers low latency,low jitter and eliminates packet losses for time critical media such as audio and video. IT service on the other hand, is used for undeterministic, bursty traffic. There is no guarantees for the performance of IT flows and thus are considered as best-effort traffic. The AV packets are sent in predetermined time slots within the point-to-point allocation period. IT packets are then fitted into the space between AV packets. Flexilink adopts dual-buffer design to separate AV and IT traffic. Figure 1 shows Flexilink modified MAC layer.At the receiving side of the switch the incoming traffic is separated into AV and IT flows and the transmitting side merge them again.Thus on the point-to-point links, AV and IT packets are multiplexed together with AV packets having higher priority leaving unoccupied bytes to be used by IT packets.

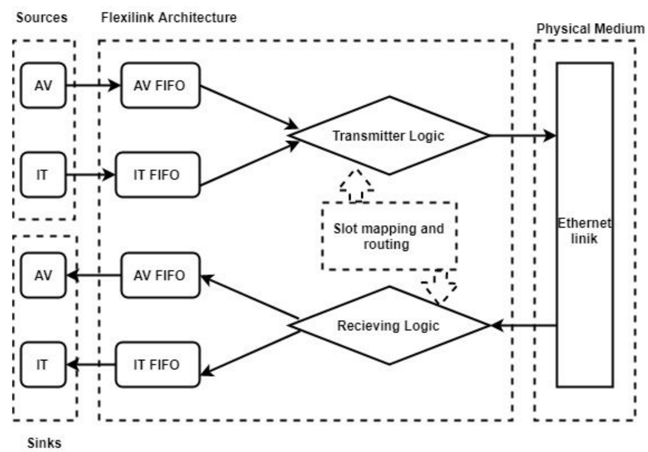


Figure 1: Flexilink Modified MAC layer

3.3. Flexilink Structure

As discussed earlier, Flexilink offers two types of services for incoming traffic

(i) AV-flow: used for critical-time traffic with deterministic intervals and requiring guaranteed QoS.
(ii) IT-flow: used for undeterministic traffics. Packets from this flow are sent at the earliest opportunity without any guarantees on delay or jitter.

AV and IT packets are allocated over an existing point-to-point network infrastructure such as Ethernet and Optical Fibers .In the Ethernet implementation of Flexilink, the frame layer uses reduced jumbo frame (RJF) format to maximise bandwidth efficiency. This format eliminates some unnecessary headers and maximise the payload when compared to the standard Ethernet frame. Headers such as the source and destination addresses are not necessary in RJF where packets are identified by their fixed position in the frame (similar to time slots concept in TDMA frames).

Figure 2 shows Flexilink layered frame architecture. Each two RJF forms up one allocation period (AP). Each RJF has a fixed size payload (P) , in which fixed size time slots can carry AV packet. Originally, the size of the time slots was designed to be of variable size depending on the size f the transmitted AV packet. However, the current implementation of Flexilink, has fixed size time slots of 63 bytes each. This was mainly for two reasons *(i)* to prevent blocking the allocation period for large time in case of big size AV packets which can lead to increasing latency for other flows *(ii)* to avoid the need to implement larger buffers. Unused bytes in each slot can be used to transmit IT packets. The AV packets have a simple single byte header containing only information about the packet size. In case of empty time slot, a one byte header is only sent to indicate to the controller (control plane implementation) that the time slot is available and can be used by IT packet. Both AV and IT packets are allocated in the RJF payload.As mentioned earlier, Flexilink can be implemented on top of various networks infrastructures provided that the bit transfer rate is fixed. As an example,in the Gigabit Ethernet implementation of Flexilink, the AP of the frame has 15570 bytes payload space to transmit AV and IT packets. The theoretical bandwidth utilisation can up to 99.6%.

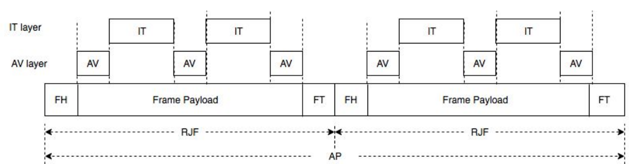


Figure 2: Flexilink Layered Frame Architecture

The RJF frame size including IFG is 7810 bytes long. Two successive RJF combine together to make a 124.96 μsec allocation period (AP) over 1 Gigabit Ethernet link which guarantees 8000 AP/second over Gigabit Ethernet link.

4. Performance Evaluation

Flexilink provides a unified network infrastructure that supports both real-time multimedia traffic as well as traditional best-effort traffic.In their work, [2] had developed a Matlab simulation tool to study and evaluate the performance of Flexilink. Simulation results showed that Flexilink provided better performance for the real-time traffic when compared to both standard Ethernet and priority queuing networks.However, no research evaluation on real kit had been performed. In this section, we will discuss the experiments done to evaluate the performance of the Flexilink over test-bed network. The main evaluation criteria will be whether Flexilink is capable of providing real-time performance guarantees for multimedia traffic over converged networks.A testbed network had been setup to carry out these experiments in which we used prototype Flexilink switches. Two different test cases had been designed to evaluate the performance of Flexilink.

- *Test Case 1 - Proof of Concept* the purpose of this test is to obtain general latency and jitter results for Flexilink experienced by time critical traffic. This test had been carried out on GigabitEthernet as layer 2 network.
- *Test Case 2 - Stress Test* the purpose of this test is to evaluate whether increasing traffic loads (best-effort traffic) will affect the performance of time critical traffic over Flexilink. For comparison purposes, same test had been carried out using state-of-the-art solutions currently used in professional audio networks,namely Dante [1].We tested performance using a testbed network which was built using their RedNet-D16 switches.

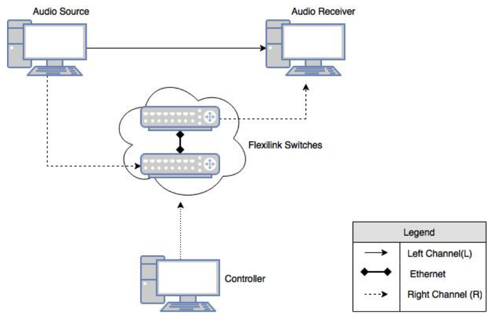
4.1. Testing Method

Flexilink protocol is designed to provide a guaranteed performance for time critical traffic over converged networks. This means that it also supports the best effort traffic without affecting the performance of the real-time traffic.
The real-time traffic is generated using Audacity [2] which is an open source audio software for generating, recording and editing.The audio source used in the test cases is in the form of pulse train generating stereo channels at 44.1 KHz with bit depth 32-bits.This audio stream represent the AV packets in Flexilink. The AV flow consists of stream of packets each is 5 bytes long with 4 bytes payload and 1 byte header. In order to be able to measure the Flexilink network end-to-end latency, stereo channel is split into two mono channels left and right (L & R). The left channel (L) is fed directly to the receiver to be used as reference to measure the network delay. The right channel (R), on the other hand is sent through the network and then recorded at the receiver.By analysing the recorded audio at the receiver, the end-to-end latency of the Flexilink network can be determined. For test case 2, the best effort traffic in the form of UDP flows sent as IT packets in Flexilink, are generated using Ostinato [3], a free software tool used for traffic generation. UDP flows are generated between sender and receiver using maximum size Ethernet packets of 1518 bytes. As mentioned earlier, IT packets are sent in the remaining bytes of each occupied time slot.

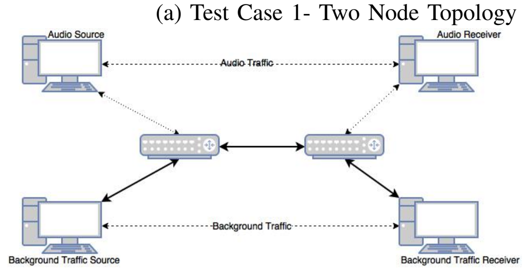
4.2. Testbed Network Topology

Two different network topologies have been used in the test cases:
Test Case 1 - Two node network topology, a sender and a receiver, is used.The sender is used as the source to generate real-time audio. The receiver then records the recieved audio which is analysed for latency and jitter calculations. Figure 3a shows the setup of the test-bed network used.
Test Case 2 - Dumbbell topology is used to measure the performance of the audio real-time traffic in the presence of varying load of background best-effort traffic. The dumbbell

1. http://www.audinate.com
2. http://www.audacityteam.org
3. http://www.ostinato.org



(a) Test Case 1- Two Node Topology



(b) Test Case 2- Dumbbell Topology

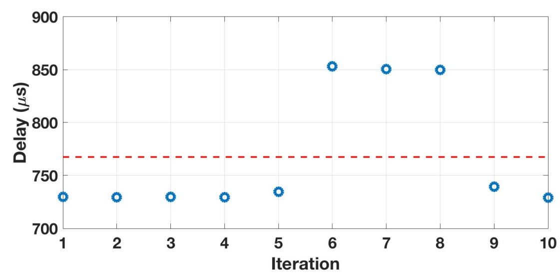
topology creates a single bottleneck which will allow to investigate the performance of the real-time audio traffic while existing with the best-effort traffic on the same link. The dumbbell topology consists of four nodes, two senders and two receivers. The first sender -receiver pair is used for the real-time audio traffic same as in experiment one. The second sender receiver pair is used for the background best-effort traffic. Background traffic is sent at 100 Mbps,500 Mbps and 900 Mbps to increase the overall network load. Figure 3b shows the test-bed network setup for the dumbbell topology.

4.3. Testing Constraints

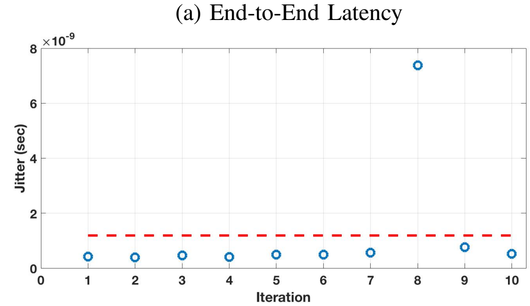
Before discussing the test results, it is necessary to point out that due to using a hardware-based approach to evaluate the performance of Flexilink, a number of constraints existed.The prototype Flexilink switches supports 8x8 audio channels, that is 8 input and 8 output audio channels. However, in test cases only 2 audio channels had been used due limitation of the available media interfaces. The used interfaces were the Digidesign Mbox 2 that provided only 2 analogue inputs and 2 analogue outputs channels. The second constraint is the use of software traffic generator tool to generate IT packets to evaluate Flexilink performance under congestion conditions. The use of software traffic generator tool instead of traffic generator server had limited the ability to over-saturate the link between the Flexilink switches and create congestion in test case 2 as shown in figure 3b. Instead, Flexilink performance was evaluated under various networks loads from 100 Mbps until 900 Mbps as mentioned earlier. In spite of these limitations the presented results are still valid for initial hardware-based performance evaluation of Flexilink. The obtained results are comparable to those obtained from Flexilink simulation presented in [2].

4.4. Test Results

Test Case 1-Proof of Concept - The aim of this experiment is to evaluate the performance of the Flexilink network in terms of providing the required timing constraints for real-time audio traffic. In this experiment, Flexilink is evaluated using Gigabit Etherent as the underlying network infrastucture.Each experiment is repeated 10 times for 2 minutes each. Figure 4a shows the end-to-end delay of audio traffic over Flexilink network using GigabitEtherent. As shown, the audio traffic experienced an average delay of 767.517 μsec over 3-hop Flexilink network.



(a) End-to-End Latency

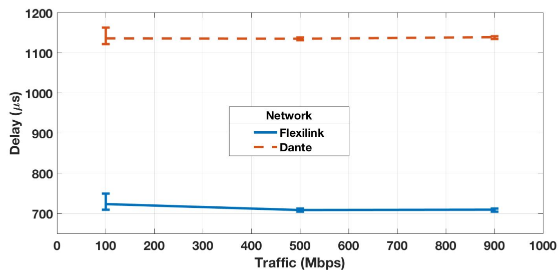


(b) Jitter

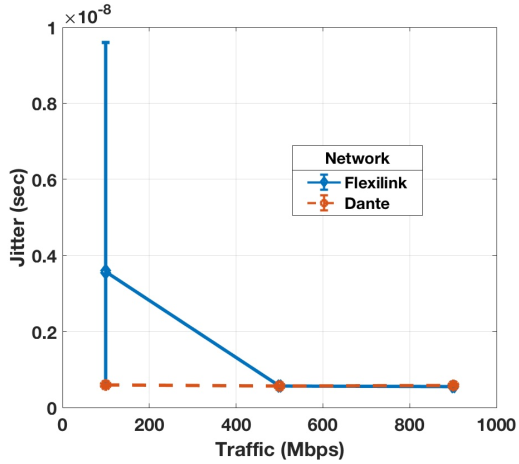
Figure 4: Flexilink over Gigabit Ethernet

Another important parameter for evaluating Flexilink performance is jitter. In packet switched networks, jitter should not exceed 20ns for real-time audio applications. If the value of jitter exceeded this value then it becomes audible and affects the received audio quality [3]. Figure 4b shows the value of jitter for Flexilink to be 1.567ns for Gigabit Ethernet.

Test Case 2- Stress Test The aim of this experiment is to evaluate the performance of real-time audio traffic over Flexilink under stressed network conditions. In order to achieve this, the amount of background traffic (IT packets) is increased to increase the overall network load to see whether this will affect the performance of the real-time audio (AV) traffic. The IT traffic used 10%, 50% and 90% of the available bandwidth i.e. at 100 Mbps, 500 Mbps and 900 Mbps. Figure 5a shows the delay experienced by the real-time audio traffic against the increasing background traffic load.It can be shown that Flexilink provides a stable performance



(a) Audio Traffic Delay over Flexilink and Testbed Networks



(b) Audio Traffic Jitter over Flexilink and Testbed Networks

Figure 5: Flexilink and Test-bed Results

to the real-time audio traffic regardless of the amount of background traffic being sent on the same link. Same tests had been carried out on testbed network. Figure 5a shows the delay achieved by to be much higher than that achieved by Flexilink. The amount of jitter experienced by the audio traffic on both Flexilink and testbed networks is shown in Figure5b. Due to the format of the results being non parametric and not normally distributed, we tested for statistical significance using a non-parametric Mann Whitney-U test with an alpha of 1% comparing the two networks (Flexilink and).A statistically significant difference was found in the delay experienced by the real-time audio traffic between the Flexilink network and network over various network loads(w=0 & $p = 1.083e\text{-}05 < 0.01$). On the other hand, no statistically significant difference was found in jitter between the two networks (w=30 & $p=0.1431>0.01$)

5. Conclusion

In this paper, an evaluation based on hardware implementation for Flexilink had been presented.Results had demonstrated the advantages of the Flexilink architecture in providing a guaranteed performance for real-time multimedia traffic even in case of heavy network load.Results came inline with key quality requirements for professional multimedia namely, end-to-end delay and jitter.This makes Flexilink a promising candidate for the converged network infrastructure in professional media industry. Being a novel technology, Felxilink has still many open areas for research. One of Flexilink main challenges is supporting real-time traffic guaranteed performance over wide area networks This will require the development of multi-hop bandwidth reservation as well as optimisation of routing and scheduling mechanisms for Flexilink.

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