A Transmitting Isochronous Multimedia Streaming Method in a Residential Ethernet Network

1, *Chi-Chun Chen and 2Chang-Hsien Chen

Abstract

With development of digital home appliances, the necessity of technology that guarantees low delay among digital devices in the LAN is increased. The study presents a transmitting isochronous packet method for time-sensitive traffic applications. The proposed method includes an information module, a scheduling module, and a forwarding module. The information module receives and records the media information of the plurality of multimedia streams. The scheduling module calculates the guaranteed bit rate of each multimedia stream according to the media information provided by the information module. The scheduling module also rearranges isochronous packets of the multimedia streams in the first time slots of a plurality of clock cycles according to the guaranteed bit rates so that the transmission of the isochronous packets satisfies the guaranteed bit rates. The lengths of the first time slot and a clock cycle are in a predetermined ratio. The forwarding module transmits all the packets of a clock cycle to a network at a time interval of the predetermined length. The simulation result of the proposed method can reduce 67% of the time-sensitive transmission delay for a 100% load.

Keywords: home appliances, transmitting network packets, information module, scheduling module, forwarding module

1. Introduction

With development of digital home appliances, applications of enjoying high quality real-time video services through a network are greatly increased. For example, a residential Ethernet is used to connect every room in the home, and multimedia data in a digital versatile disc (DVD) player located in a bedroom can be viewed via a television located in a living room in real-time through a network streaming, as shown in Figure 1. The conventional Ethernet applies a carrier sense multiple access communication protocol with collision detection (CSMA/CD) to compete with each other for the same bandwidth to transmit packets [1]. Such mechanism is not suitable for transmitting time-sensitive packets such as video and audio files since a transmitting time is liable to be delayed to decrease a quality of service (QoS) [2][3][4]. Therefore, under an environment of limited bandwidth and in case that file transmission trends to be more complicated, to effectively adjust the bandwidth to increase a system performance is an important issue [5][6].

An institute of electrical and electronics engineers (IEEE) 802.1 p/q is a standard established for resolving the transmission QoS issue, in which a tag of 4 bytes is added in a header of layer 2 [7]. The front two bytes are fixed to be 8100H, which serve as an identification of an Ethernet packet type. The first 3 bits of the next 2 bytes are used for storing priorities. A transmission packet can be classified into 8 priorities, and the higher priority denotes a higher sending priority. If the priorities of isochronous packets of a video/audio file are increased, the transmission delay of the conventional Ethernet due to all of the packets being the same priority can be ameliorated. Therefore, 802.1 p/q has a problem which is mentioned below.

- The packet is simultaneously in switch, and some of the packet that arrived may be lost by the restricted queue size in the switch. If the lost packet is best-effort traffic, it matters little for communication; but if it is an isochronous packet, it matters for real-time processing because the transmission will have many jitters [8].
- Ethernet that follows IEEE 802.1 p/q is able to transmit earlier in a high-priority queue than in a low priority queue. Nevertheless, an isochronous packet that is a high-priority packet may be delayed by the low-priority packet, due to a nondeterministic transmission [9].
- There is a greater possibility of delay in the isochronous traffic when the transmission hop count is increased more. The reason for the increase of delay in the effect of the low-priority packet by nondeterministic characteristic of communication and cumulated jitter exists due to same priority of the transmissions in both packets [10].

*Corresponding Author: Chi-Chun Chen
(E-mail: chichun@ncut.edu.tw)
1Department of Electronic Engineering, National Chin-Yi University of Technology, Taiwan
2 Cloud Service Center, Industrial Technology Research Institute, Taiwan

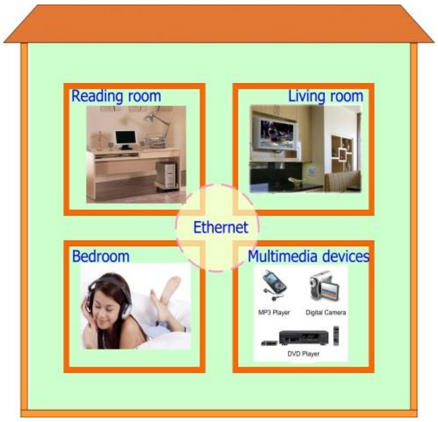

Figure 1: Communication between time-sensitive consumer electronics applications in a home network

Another technology used for time-sensitive traffic applications is the time slot transmission. Cho et al. provided a concept of time slot to monitor an applicable bandwidth of the network, and a counter is used to count a time interval for transmitting the isochronous packets [11]. When the applicable bandwidth is relatively great, a relatively fast bit rate is used to transmit the isochronous packets; and when the applicable bandwidth is relatively small, a relatively slow bit rate is used to transmit the isochronous packets. However, the packet delay of some nodes would occur when multiple nodes simultaneously transmit isochronous packets in the same time slot. In this study, each node would synchronize media information with each other. Therefore, the transmission time of isochronous packets do not affect each other, in which a plenty of multimedia streams can be accommodated in a limited bandwidth, so as to achieve a better usage rate of the network bandwidth to guarantee a quality of service (QoS).

2. Related work

The proposed method is based on technology of IEEE 802.1 audio video bridging (AVB) standard. IEEE 802.1 AVB is to provide the specifications that will allow time synchronized low latency streaming services through 802 networks [12]. IEEE 802.1 AVB contains three main components: a timing and synchronization method (IEEE 802.1 AS), reservation method (IEEE 802.1 Qat), and frame forwarding rules (IEEE 802.1 Qav), as illustrated in Figure 2.

IEEE 802.1AS is based on IEEE Std 1588TM–2008 [13] (referred to as a precision time protocol (PTP) profile). This standard specifies the protocol and procedures used to ensure that the synchronization requirements are met for time sensitive applications, such as audio and video, across Bridged and Virtual Bridged Local Area Networks consisting of LAN media where the transmission delays are fixed and symmetrical. This includes the maintenance of synchronized time during normal operation and following addition, removal, or failure of network components and network reconfiguration. It specifies the use of IEEE 1588 specifications where applicable in the context of IEEE Stds 802.1D and 802.1Q.

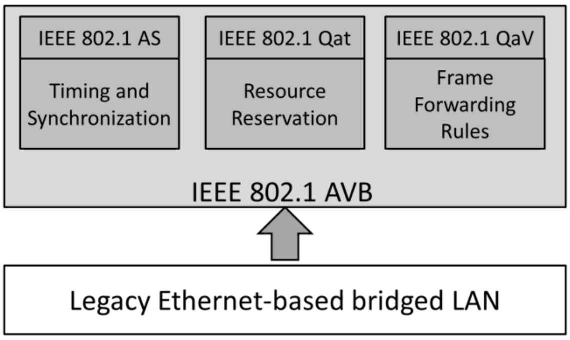

Figure 2: Overview of IEEE 802.1 AVB

Synchronization to an externally provided timing signal (e.g., a recognized timing standard such as UTC or TAI) is not part of this standard, but it is not precluded [14].

IEEE 802.1 Qat proposes a standardization of a reservation method. This standard specifies protocols, procedures and managed objects, usable by existing higher layer mechanisms, which allow network resources to be reserved for specific traffic streams traversing a bridged local area network. It identifies traffic streams to a level sufficient for bridges to determine the required resources and provides a mechanism for dynamic maintenance of those resources [15].

IEEE 802.1 Qav is a standard for frame forwarding and queuing enhancements for time-sensitive streams. This standard allows bridges to provide guarantees for time-sensitive, loss-sensitive real-time audio video (AV) data transmission (AV traffic). It specifies per priority ingress metering, priority regeneration, and timing-aware queue draining algorithms. This standard uses the timing derived from IEEE 802.1AS. Virtual Local Area Network (VLAN) tag encoded priority values are allocated, in aggregate, to segregate frames among controlled and non-controlled queues, allowing simultaneous support of both AV traffic and other bridged traffic over and between wired and wireless LANs [16].

3. Methodology

The proposed method is performed by a media server and can be described as follows. First, media information of a plurality of multimedia streams provided by the media server is received and recorded. A guaranteed bit rate of each of the multimedia streams is calculated according to the media information. Isochronous packets of the multimedia streams are rearranged in first time slots of a plurality of clock cycles according to the guaranteed bit rates, so that the transmission of the isochronous packets satisfies the guaranteed bit rates. Then all the packets of one of the clock cycles are transmitted to a network at every a time interval of a predetermined length.

3.1 Clock Cycles of Isochronous Packets Transmitting

A clock cycle of transmitting multimedia streaming packets according to the present study is shown in Figure 3. The method was performed by a media server connected to a network, such as a residential Ethernet. The present embodiment is based on the IEEE 802.1 audio video bridging (AVB) standard, wherein a clock cycle of 125 microseconds (μs) is adopted as the basic unit for packet transmission. As shown in Figure 3, the clock cycle of 125 μs is divided into two time slots, wherein the front ¾ is the first time slot, and the rear ¼ is the second time slot. The first time slot is used for transmitting isochronous packets, which are indicated as “ISO” in Figure 3, and the second time slot is used for transmitting asynchronous packets, which are indicated as “ASY” in Figure 3.

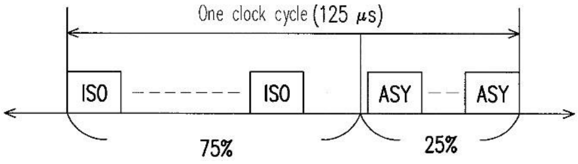

Figure 3: One clock cycle of transmitting multimedia streaming

3.2 Proposed System

Figure 4 is the system architecture for transmitting network packets according to the present paper. The system for transmitting network packets may be part of the aforementioned media server, which can be used in a residential Ethernet for providing multimedia streams to various video audio players. The proposed system for transmitting network packets can be implemented by hardware or software.

The system for transmitting network packets includes a classification module, an information module, a scheduling module, a fragmentation module, a synchronization module, and a forwarding module, as shown in Figure 4. A network interface module is an interface between the proposed system for transmitting network packets and a network. The classification module stores an asynchronous queue and a plurality of isochronous queues, wherein each of the isochronous queues corresponds to one of four multimedia streams (S1-S4) provided by the media server. Although four isochronous queues and four multimedia streams are illustrated in Figure 4, the present study is not limited to the number of four. The isochronous queues of Figure 4 are all linked lists, though the present disclosure is not limited to the queue of the linked list type.

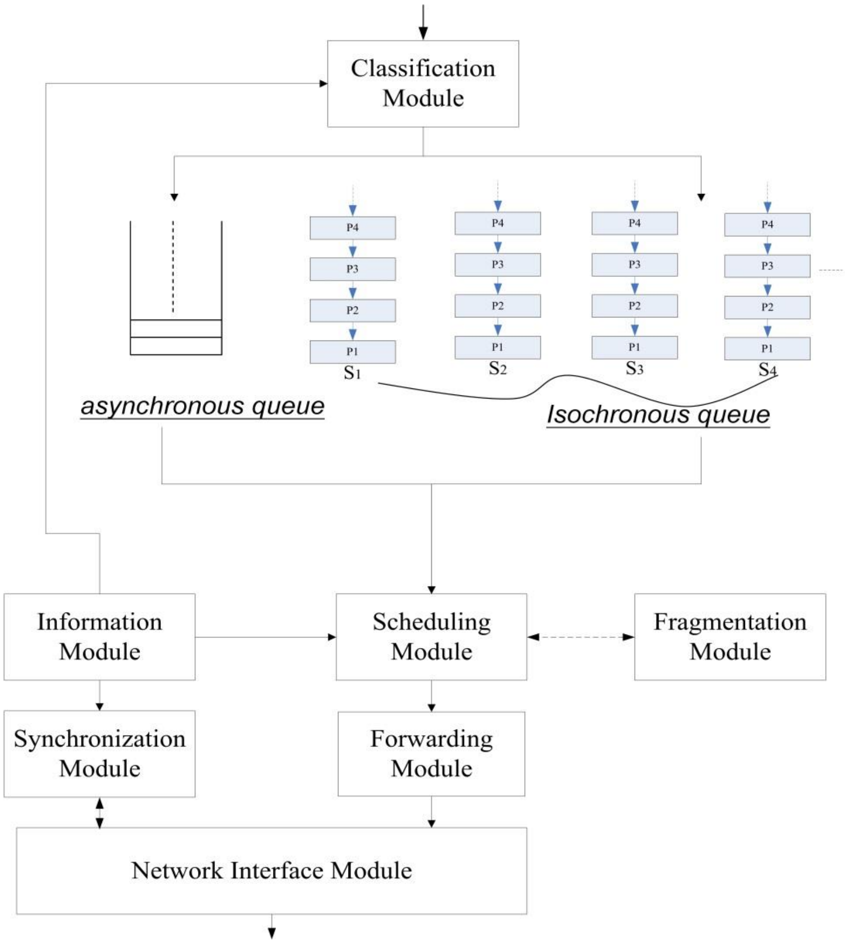

Figure 4: Block diagram of the proposed system

Figure 5 is a flowchart illustrating a method for transmitting network packets according to the present paper. The method is performed by the aforementioned media server. First, the information module receives and records media information of the multimedia streams (S1-S4) from an application layer of the media server. The media information may include information such as the session ID, the frame rate and the frame size, etc., of each of the multimedia streams (S1-S4).

Then the classification module receives packets from the application layer, and classifies a type of the received packet; stores each isochronous packet of the received multimedia streams (S1-S4) into a corresponding one of the isochronous queues according to the media information provided by the information module; further stores received asynchronous packets into the asynchronous queue. In the present embodiment, each of the isochronous packets includes the session ID of the corresponding multimedia stream. The classification module receives the media information provided by the information module, and allocates four corresponding isochronous queues according to the session IDs in the media information of the multimedia streams (S1-S4). The classification module further compares the session IDs of the isochronous packets with the session IDs of the multimedia streams (S1-S4), so as to store each isochronous packet into the corresponding isochronous queue of the corresponding multimedia stream.

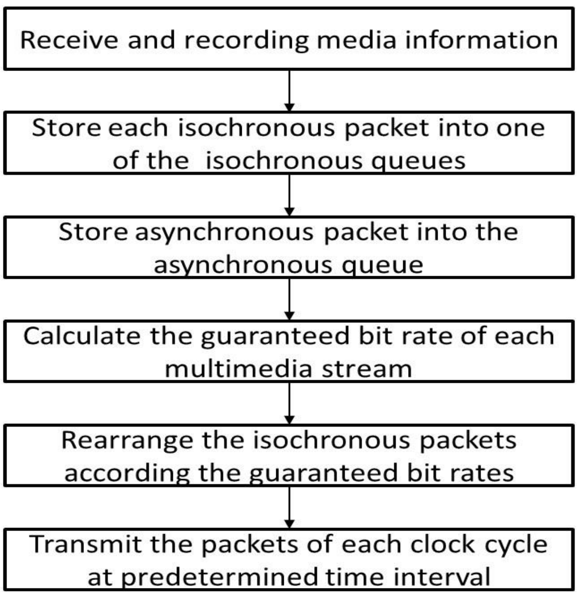

Figure 5: Flowchart of the proposed transmitting network packets

3.2.1 Scheduling Module

Figure 6 is a block diagram illustrating the scheduling module. The scheduling module includes a bit rate generating module and a rearranging module. The bit rate generating module calculates a guaranteed bit rate of each of the multimedia streams (S1-S4) according to the media information provided by the information module. The rearranging module rearranges the isochronous packets of the multimedia streams (S1-S4) into the first time slots of a plurality of clock cycles according to the guaranteed bit rates, so that the transmission of the isochronous packets satisfies the guaranteed bit rates. The bit rate generating module can calculate the guaranteed bit rates of the multimedia streams (S1-S4) according to the frame rates and the frame sizes in the media information of the multimedia streams (S1-S4), and then the rearranging module determines a transmitting sequence of the isochronous packets in each clock cycle according to the guaranteed bit rates.

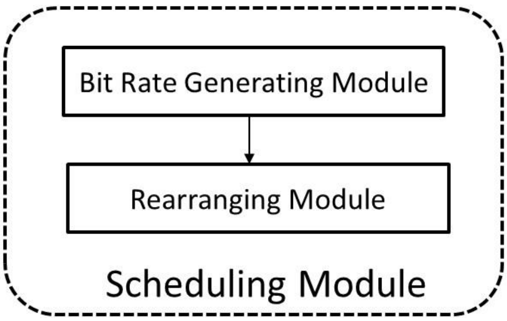

Figure 6: Block diagram of the scheduling module

For example, the rearranging module may arrange enough isochronous packets to satisfy the guaranteed bit rates in the first time slot of each clock cycle according to the guaranteed bit rates of the multimedia streams (S1-S4). As shown in Figure 7, assuming a size of each of the isochronous packets (P1-P4) is X bytes, the guaranteed bit rate required by each of the multimedia streams S1, S3 and S4 is X bytes per clock cycle, and the guaranteed bit rate required by the multimedia stream S2 is X/2 bytes per clock cycle. In the example of Figure 7, the rearranging module uses a linked list to link the isochronous packets arranged in each of the clock cycles according to a transmitting sequence. Since the guaranteed bit rate required by each of the multimedia streams: S1, S3 and S4, is X bytes per clock cycle, the rearranging module arranges one isochronous packet of the multimedia stream S1, one isochronous packet of the multimedia stream S3, and one isochronous packet of the multimedia stream S4 in each clock cycle. Since the guaranteed bit rate required by the multimedia stream S2 is X/2 bytes per clock cycle, the rearranging module arranges one isochronous packet of the multimedia stream S2 in every two clock cycles. As shown in Figure 7, in the first clock cycle, the first isochronous packets P1 of the multimedia streams: S1, S3 and S4, are transmitted. In the second clock cycle, the second isochronous packets P2 of the multimedia streams: S1, S3 and S4, and the first isochronous packet P1 of the multimedia streams S2 are transmitted. In the third clock cycle, the third isochronous packets P3 of the multimedia streams: S1, S3 and S4, are transmitted. If a guaranteed bit rate required by a certain multimedia stream is 2X bytes per clock cycle, the rearranging module arranges two isochronous packets of such multimedia stream in each clock cycle, so as to guarantee the guaranteed bit rate.

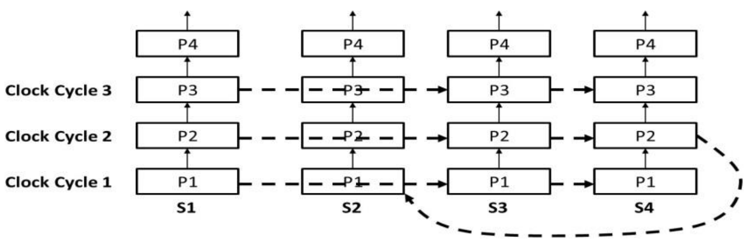

Figure 7: Packet rearrangement of the present method

2.2.2 Forwarding Module

The forwarding module transmits all of the packets of a current clock cycle to the network at every time interval of the clock cycle according to the transmitting sequence arranged by the rearranging module, as shown in Figure 8. The forwarding module uses the first time slot to transmit the isochronous packets of each clock cycle, and uses the second time slot to transmit the asynchronous packets in the asynchronous queue. The asynchronous packets are not rearranged, and are directly transmitted according to a sequence of entering the asynchronous queue. When there is no isochronous packet to be transmitted at a clock cycle, or when transmission of the isochronous packets of such clock cycle has been completed before the first time slot ends, the forwarding module is switched immediately to transmit the asynchronous packets in the asynchronous queue, so as to improve a usage rate of the bandwidth.

If there is a plurality of the same systems for transmitting network packets in a local network, the synchronization module can ensure that these systems for transmitting network packets do not affect the transmission quality of service (QoS) to each other. The synchronization modules of the systems for transmitting network packets can mutually exchange the respective media information, i.e. synchronize the media information of all of the multimedia streams in the whole network. In this case, the scheduling module does not rearrange the isochronous packets only according to the media information of the multimedia streams provided by the media server to which the scheduling module belongs, but it rearranges the isochronous packets in the first time slots of each clock cycle according to the media information of all of the multimedia streams in the whole network.

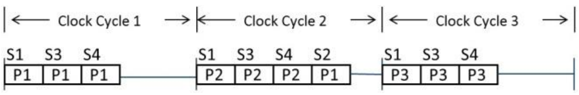

Figure 8: Packet transmitting timing diagram corresponding to Figure 7

2.2.3 Synchronization Mechanism

Figure 9 illustrates a synchronization example of the present embodiment. Wherein, node 1 and node 2 are two media servers in the local network, which respectively include a proposed system for transmitting network packets shown in Figure 4. The node 1 provides four multimedia streams: S1-S4, and the node 2 provides two multimedia streams: S5 and S6. The synchronization modules of the node 1 and the node 2 can mutually exchange the media information, so that the scheduling modules of the node 1 and the node 2 can rearrange the respective isochronous packets according to the media information of all of the multimedia streams: S1-S6. Shown as a result of Figure 9, the scheduling modules of the node 1 and the node 2 can cooperate with each other. During the odd clock cycle, the node 1 only transmits three isochronous packets, and the node 2 transmits two isochronous packets. During the even clock cycle, the node 1 transmits four isochronous packets, and the node 2 transmits only one isochronous packet. In this way, a mutual influence of the transmission QoS in the limited bandwidth can be avoided. Otherwise, if the scheduling modules of the node 1 and the node 2 do not cooperate with each other, the node 2 probably transmits one isochronous packet during the odd clock cycle, and transmits two isochronous packets during the even clock cycle, which may lead to a waste of the network bandwidth during the odd clock cycle, and insufficient network bandwidth during the even clock cycle.

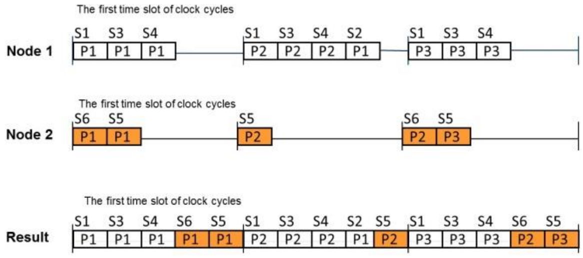

Figure 9: In case of synchronous media information of the present method

2.2.4 Fragmentation Mechanism

According to a fragmentation mechanism of the fragmentation module, a remaining bandwidth of each clock cycle can be used to transmit more packets. The fragmentation mechanism of the isochronous packets is shown as Figure 10 (a). The scheduling module already arranges a packet transmitting timing shown as node1 of Figure 10 (a), and meanwhile an isochronous packet P1 of another multimedia stream is still waiting to be transmitted (shown as a transmitting timing node2 of Figure 10). Now, the remaining bandwidth of the first time slot is insufficient to transmit the isochronous packet P1, especially during the third and the fourth clock cycles. In this case, the scheduling module activates the fragmentation module, and the fragmentation module divides the isochronous packet P1 into a plurality of small packets (P_F) according to the remaining bandwidth of the first time slots of the current clock cycle and follow-up clock cycles. Then the scheduling module arranges the small packets (P_F) into the remaining bandwidth of the first time slots of the current clock cycle and the follow-up clock cycles, as shown in a transmitting timing result of Figure 10 (a). In this way, transmission of the isochronous packet P1 can satisfy the guaranteed bit rate required by its multimedia stream, and meanwhile the usage rate of the network bandwidth is increased.

A fragmentation mechanism of the asynchronous packets is shown in Figure 10 (b). The scheduling module uses the second time slot of the clock cycle to transmit the asynchronous packets, as shown in a transmitting timing node1 of Figure 10 (b). However, asynchronous packets P1 and P2 in the asynchronous queue are still waiting to be transmitted. Now, the remaining bandwidth of the second time slot is insufficient to transmit the entire asynchronous packets P1 and P2. In this case, the scheduling module activates the fragmentation module, and the fragmentation module divides the asynchronous packets P1 and P2 into a plurality of small packets (P_F) according to the remaining bandwidth of the second time slots of the current clock cycle and follow-up clock cycles. Then the scheduling module arranges the small packets (P_F) into the remaining bandwidth of the second time slots of the current clock cycle and the follow-up clock cycles, as shown in a transmitting timing result in Figure 10 (b). In this way, the usage rate of the network bandwidth is increased.

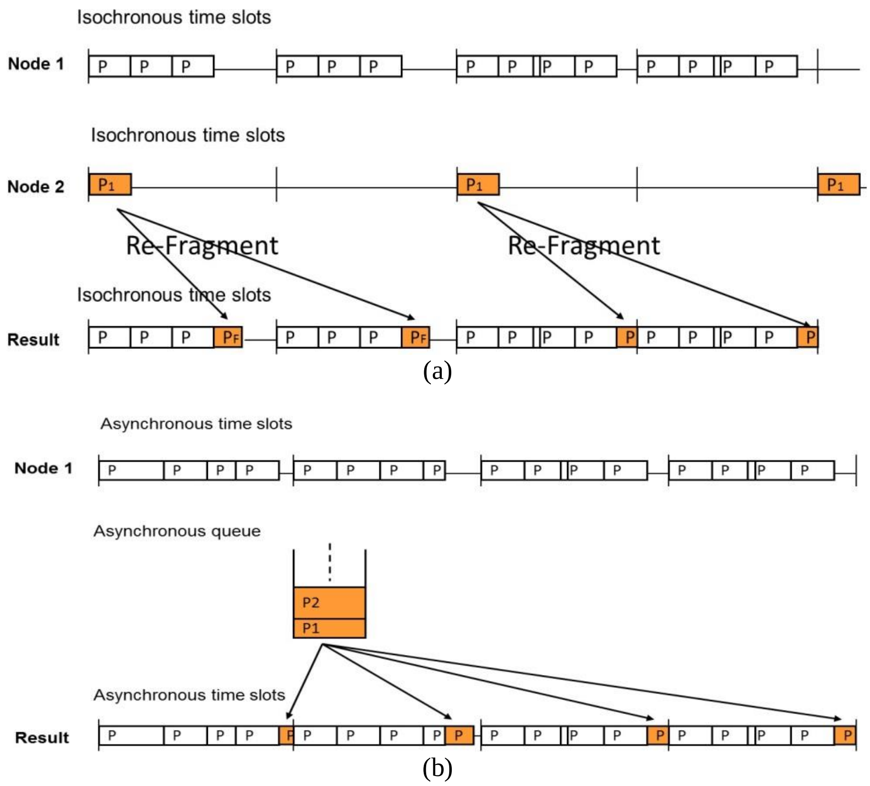

Figure 10: (a) Isochronous packets fragmentation. (b)Asynchronous packets fragmentation.

4. Discussion and Conclusions

In order to verify the main idea proposed in this paper, the network model of the proposed method is implemented using OPNET. The end device generates best-effort traffic and time-sensitive traffic. The end-to-end delays comparison of time-sensitive traffic based on network loads are shown in Figure 11. The transmission delay of time-sensitive traffic in the proposed system took 0.9ms for a 10% load, and 3.1ms for a 100% load when the generation of time-sensitive traffic to best-effort traffic was 1:1. The transmission delay of best-effort method took 1.1ms for a 10% load, and 5.2ms for a 100% load. Therefore, the proposed method can reduce 67% of the time-sensitive transmission delay for a 100% load.

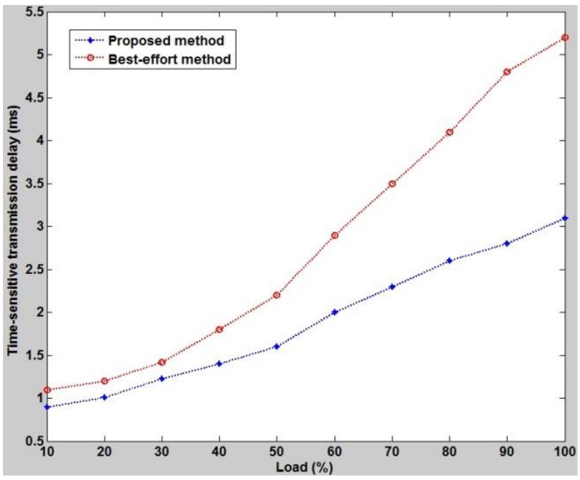

Figure 11: Time-sensitive transmission delay of proposed method and best-effort method

According to the aforementioned system and method for transmitting network packets, a plenty of multimedia streams can be accommodated in the limited bandwidth. The media servers of the same network can mutually exchange media information, so that they do not influence each other's respective transmission QoS. According to the packet fragmentation mechanism, the packets can be arranged into the remaining bandwidth, so that the usage rate of the network bandwidth is increased, and the QoS is guaranteed.

References

[1]. G. Antonio, "On the applicability of the number of collisions in p-persistent CSMA/CD protocols," Computers & Operations Research, vol. 37, no. 7, pp. 1199-1211, Jul. 2010.

[2]. O. P. Igumbor, and R. Foss, "Control protocol command translation for device interoperability on ethernet AVB networks," Journal of the Audio Engineering Society, vol. 61, no. 4, pp. 224-234, Apr. 2013.

[3]. G. Alderisi, G. Iannizzotto, and L. Lo Bello, "Towards IEEE 802.1 Ethernet AVB for advanced driver assistance systems: a preliminary assessment," IEEE Conference on Emerging Technologies & Factory Automation (ETFA), pp. 17-21, Sep. 2012.

[4]. O. Kleineberg, P. Froehlich, and D. Heffernan, "Fault-tolerant Audio and Video Bridging (AVB) Ethernet: A novel method for redundant stream registration configuration," IEEE Conference on Emerging Technologies & Factory Automation (ETFA), pp. 17-21, Sep. 2012.

[5]. J. Diemer, J. Rox, and R. Ernst, "Exploring the worst-case timing of Ethernet AVB for industrial applications," IEEE Conference on Industrial Electronics Society (IECON), pp. 25-28, Oct. 2012.

[6]. Y. D. Choi, J. W. Lee, and S. R. Ryu, "A forwarding method on the IEEE 802.1 AVB," 7th International Conference on Advanced Language Processing and Web Information Technology, pp. 497-502, Jul. 2008.

[7]. Y. D. Choi, J. W. Lee, and S. R. Ryu, "An UPnP based media distribution system supporting qos in a converged home network," IEEE Conference on Network Operations and Management, pp. 3-7, Apr. 2006.

[8]. F. Li, J. Yang, and C. An, "Towards centralized and semi-automatic VLAN management," International Journal of Network Management, vol. 25, no. 1, pp. 52-73, Jan. 2015.

[9]. A. Gopalan, and S. Ramasubramanian, "Fast recovery from link failures in Ethernet networks," IEEE Transactions on Reliability, vol. 63, no. 2, pp. 412-426, Jun. 2014.

[10]. T. Ando, O. Shimokuni, and K. Asano, "Network virtualization for large-scale data centers," Fujitsu Scientific & Technical Journal, vol. 49, no. 3, pp. 292-299, Jul. 2013.

[11]. J. H. Cho, S. H. Kim, Y. J. Oh, Y. H. Joo, S. Wang, and J. H. Yoon, "Residential Ethernet node device for transmitting synchronous data using counter and synchronous data transmitting method thereof," United States Patent Application Publication, Pub. No: 20070025385, Feb. 1, 2007.

[12]. IEEE 802.1 Audio/Video Bridging Task Group, http://www.ieee802.org/1/pages/avbridges.html .

[13]. A. Vallat, and D. Schneuwly, "Clock synchronization in telecommunications via PTP (IEEE 1588)," IEEE International Frequency Control Symposium, pp. 334-341, May 2007.

[14]. G. M. Garner, and H. Ryu, "Synchronization of Audio/Video Bridging networks using IEEE 802.1AS," IEEE Communications Magazine, vol. 49, no. 2, pp. 140-147, Feb. 2011.

[15]. G. M. Garner, F. Feng, and K. den Hollander, "IEEE 802.1 AVB and its application in carrier-grade Ethernet," IEEE Communications Magazine, vol. 45, no. 12, pp. 126-134, Dec. 2007.

[16]. IEEE Standard for Local and metropolitan area networks, "Virtual Bridged Local Area Networks Amendment 12: forwarding and queuing enhancements for time-sensitive streams," IEEE Computer Society, Dec. 2009.

Chi-Chun Chen was born in Hsinchu, Taiwan, in 1977. He received the Ph.D. degree in Electrical Engineering from National Cheng Kung University, Tainan, Taiwan, R.O.C., in 2014. He is currently an Assistant Professor with the Electronic Engineering Department, National Chi-Yi University, Taiwan. His research interests include network protocol, cloud service and biomedical instrumentation.

Chang-Hsien Chen was born in Yilan, Taiwan, in 1977. He received the M.S. degree in the Department of Electronic Engineering, Institute of Computer and Communication, National Taipei University of Technology, Taiwan, in 2002. He is currently a Computer Engineer with the Cloud Server Center, Industrial Technology Research Institute (ITRI), Taiwan. His research interests include network protocol, cloud service and embedded system.