Efficient Real-time Multimedia Streaming System Using Partial Transport Stream for IPTV Services

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Abstract—IPTV Content delivery systems over wired networks confront scalability problems due to their high network bandwidth requirement for real-time services. Especially, VoD service provides Trick Mode features such as pause, fast forward and similar operations. However, Trick Mode services are delivered by the method of unicast only for controlling of the stream. With a point of views, this paper propose a new real-time multimedia streaming architecture over IP Networks, which tries to achieve bandwidth efficiency and supporting for mass clients better than traditional unicast services. The proposed methods divide the Transport Stream into a series of segments. After that, this divided partial Transport Stream makes multicast streaming periodically. Meanwhile Set-top Box of a client makes a rearrangement orderly by using Presentation Time Stamp field from the served Transport Stream packets. While the current Transport Stream segment is playing, it should be guaranteed that the next segment is downloaded on time. Consequently, the original video content can be played out continuously. The detail introduction of a new real-time multimedia streaming system with analysis and simulation follows as below.

Index Terms—Interactive TV, IPTV, MPEG-2 Transport Stream, Multicast Streaming.

1 INTRODUCTION

NOWADAYS multimedia streaming services is the digital television services which deliver data streams over IP Networks. Especially, the VoD(Video On Demand) services are being raised a significant issue in commercial respect. The VoD is an interactive service which is able to provide the video data being stored digital data forms such as movies, education, games and shopping, etc to many users at distant places using communication networks [1]. The video which is a set of a large data such as motion pictures and sounds, etc. must be sent in the form of a lot of data even after the compression. In case of transmitting the video data over network, high network bandwidth and a lot of expenses are required because of getting a sufficient bandwidth. So, it is an important problem to make the best usage of a bandwidth efficiently.

In general, the VoD is mainly classified as the TVoD (True-VoD) and the NVoD (Near-VoD) depending on the method of transmitting the video. The TVoD is a method in which a viewer is able to watch the desired program at a desired time. It is an interactive service of providing multimedia services to viewers by using of only unicast method. While the TVoD has the advantage of interactive services, it requires high network bandwidth to provide video services because the viewer occupies the bandwidth from the broadcasting stream server to the viewer STB(Set-top Box). On the other hand, the NVoD enables many viewers to watch TV at the same time by using of periodically repeating one broadcasting stream in series. The NVoD requires a considerably narrow bandwidth comparing to the TVoD, but the NVoD has a disadvantage in that many viewers are not able to watch the video right away [2], [3]. In this paper, we propose a new real-time

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multimedia streaming delivery architecture to enable viewers to watch the video right away and to use the bandwidth efficiently by dividing the entire TS(Transport Stream) partially. The newly proposed architecture delivers some of initial data by using the unicast method and the afterward data is delivered by using the multicast method periodically in order to improve the existing method of transmitting data by using only unicast. The STB makes packets rearrange orderly by using the PTS (Presentation Time Stamp) field of TS packets. The TS segments are received continuously without discontinuity, while the viewer watches the current TS segments. Consequently, the viewer can watch the video streams continuously.

The related work will be reviewed in Section 2. In Section 3 and 4, we will propose a new system operation, at the server side and the client side orderly. Also, we will show the simulation results in Section 5. Lastly, we will conclude the paper in Section 6.

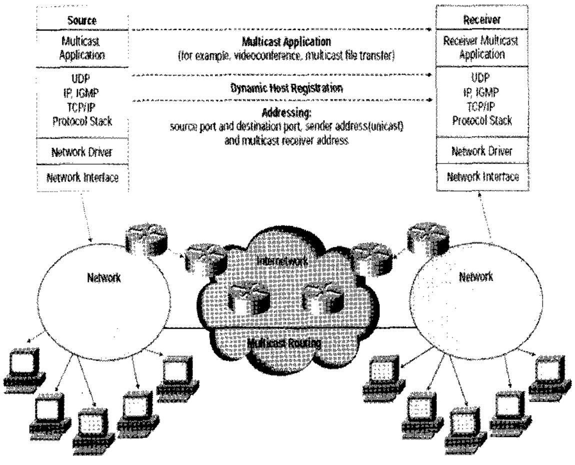


Fig. 1. Multicast-Enabled Network. To support IP multicast, the sending and receiving nodes, intermediate routers and the network infrastructure between them must be multicast-enabled

2 Related Works

2.1 Multicast Routing and IGMP Protocols

The unicast of point-to-point connection method have been used at traditional broadcasting services in order to transmit the broadcasting contents over IP networks. The unicast is a method of transmitting the contents to individual host whenever users request the contents. This creates a requirement of a wide bandwidth and a lot of loads in networks. In order to resolve this problem, a contents delivery method of the multicast has been presented.

The multicast method delivers a broadcasting stream to many hosts that belong to the same group at the same time [4]. In other words, a broadcasting stream data is transmitted from the Head End (H/E) center to the multicast router in a similar the existing broadcasting method.

The multicast router floods multicast packets followed by multicasting forward entry generated by multicast routing protocol once the multicast packet is received. Flooding stands for delivering one packet received from one host to all hosts connected to the router or delivering the routing information corrected at large scale network to all adjacent nodes. Generally, at L2 switch, the IGMP (Internet Group Management Protocol) Snooping technique is used in order to reduce the flooding of multicast packets. The IGMP Snooping Protocol detects IGMP packets being exchanged between router and host while enabling the recognition of port where the router is nearby based on MAC (Media Access Control) address of each port. The hosts must be able to join or leave multicast group in order to support multicasting and must inform the host group membership information to nearby multicast routers. The thing which is defined for this is IGMP. The IGMP is composed of membership query message and membership report message.

In IGMP version 1, multicast routers send the membership query message by having 224.0.0.1 as destination in order to find group subscriber hosts connected to that router. And the hosts send membership report message while checking and maintaining the group membership by having 224.0.0.2 as destination for the membership query message being received from the multicast router. In other words, the multicast router sends the query message of asking hosts whether they are participating in the multicast group and for this, each host maintains group

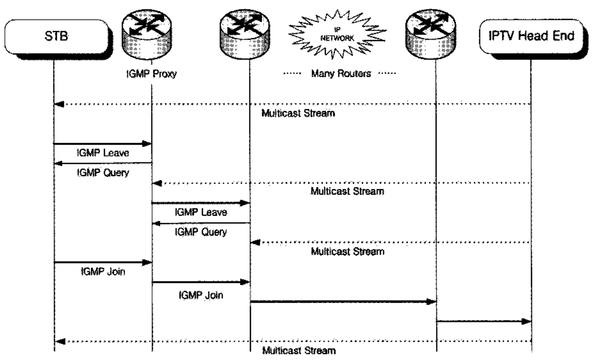


Fig. 2. IGMP join & leave process - in IGMPv2, the host can leave/join at once by transmitting group leave/join message to multicast router

membership by sending report message on that query within a fixed amount of time. If there is no response although the multicast router has asked regularly, the traffic of corresponding group is no longer forwarded. But the problem of wasting bandwidth is created due to unnecessary multicast traffic during periodic query time during group leave. In IGMP version 2, the group leave report to improve such problem as shown in Fig. 2 has been added [5]-[6]. This makes the host be able to leave at once by transmitting group leave message to router aggressively in case it no longer desires the contents they currently belong to [7].

2.2 MPEG-2 Transport Stream

PES (Packetized Elementary Stream) is a specification defined by the MPEG communication protocol (see the MPEG-2 standard) that allows an Elementary stream to be divided into packets [8].

The elementary stream is packetized by encapsulating sequential data bytes from the elementary stream inside PES packet headers. A typical method of transmitting elementary stream data from a video or audio encoder is to first create PES packets from the elementary stream data and then to encapsulate these PES packets inside TS packets or program stream. TS is a format specified in MPEG-2 Part 1, Systems (ISO/IEC standard 13818-1). Its design goal is to allow multiplexing of digital video and audio and to synchronize the output.

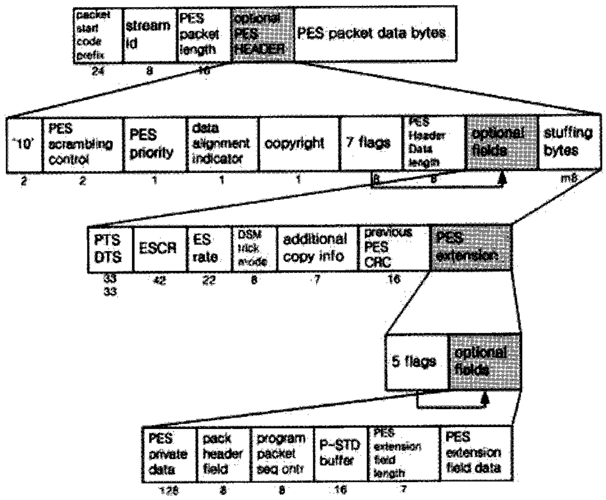


Fig. 3. PES packet syntax diagram. It is a specification defined by the MPEG communication protocol that allows an Elementary stream to be divided into packets

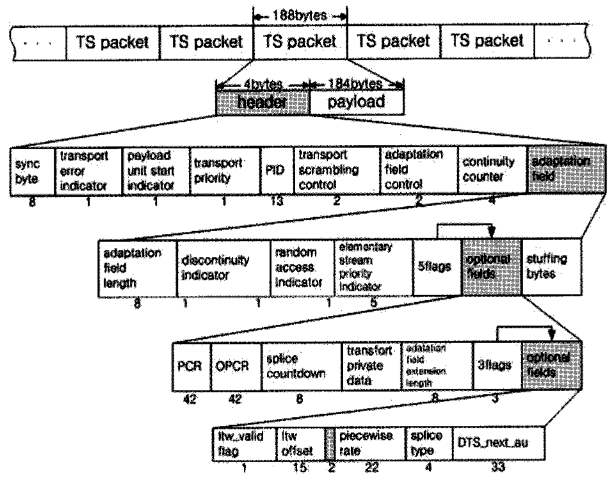


Fig. 4. A packet is the basic unit of data in a transport stream. It consists of a sync byte, whose value is 0x47, followed by three one-bit flags and a 13-bit PID

Transport stream offers features for error correction for transportation over unreliable media, and is used in broadcast applications such as DVB and ATSC. A packet is the basic unit of data in a transport stream. It consists of a sync byte, whose value is 0x47, followed by three one-bit flags and a 13-bit PID. This is followed by a 4-bit continuity counter, which usually increments with each subsequent packet of

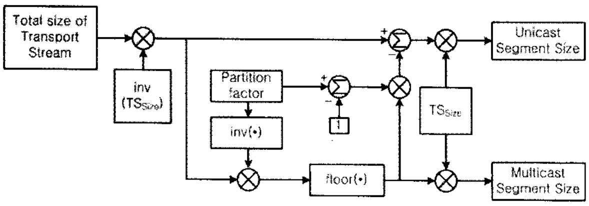


Fig. 5. The process of determining the size of partial unicast/multicast segment followed as a block diagram

a frame, and can be used to detect missing packets. Additional optional transport fields, whose presence may be signaled in the optional adaptation field, may follow. The rest of the packet consists of payload. Packets are most often 188 bytes in length, but some transport streams consist of 204-byte packets which end in 16 bytes of Reed-Solomon error correction data. The 188-byte packet size was originally chosen for compatibility with ATM systems. Each table or elementary stream in a transport stream is identified by a 13-bit PID. A demultiplexer extracts elementary streams from the transport stream in part by looking for packets identified by the same PID. In most applications, Time-division multiplexing will be used to decide how often a particular PID appears in the transport stream [10].

3 SERVER SIDE SYSTEM OPERATION

3.1 Transport Stream Segmentation

Referring to Fig. 5, on the server side, suppose there is a entire transport stream with size, D_{size} [bytes]. TS Segmentation Procedure:

In order to split the TS, the partition factor K is determined. The partition factor means an optional factor to split the entire transport streams as many segments. To split the TS, read the total number of bytes, D_{size}, in the entire size of TS. When the D_{size} is divided by the TS packet size of 188 bytes, TS_{size}, the total number of TS packets, $D_{packets}$, can be calculated.

$$D_{packets} = D_{size} \times \left(\frac{1}{TS_{size}}\right) \tag{1}$$

Fig. 6. The result of sectional partitioning the entire TS stream by selecting the partition factor as 4

The size of multicast segment, MS_{size}, is determined by means of partition factor, integer K.

$$MS_{packets} = \left\lfloor D_{packets} \times \left(\frac{1}{K}\right)\right\rfloor, (K \ is \ even) \tag{2}$$

The MS_{size} can be calculated when the TS packet size value of 188 bytes, TS_{size}, is multiplied to the value after removing the number less than the decimal point.

$$MS_{size} = MS_{packets} \times TS_{size} \tag{3}$$

The size of unicast segment, $US_{packets}$, can be determined when the value of partition factor K minus 1 is multiplied to the $MS_{packets}$ and then subtract this result from the TS_{size}.

$$US_{packets} = TS_{size} - [MS_{packets} \times (K-1)] \tag{4}$$

If the $TS_{packets}$ value of 188 bytes is multiplied to this value once again, the unicast segment size followed by partition factor can be determined.

$$US_{size} = US_{packets} \times TS_{size} \tag{5}$$

When the $TS_{packets}$ value of 188 bytes is multiplied to the $US_{packets}$ once again, the unicast segment size, US_{size}, followed by partition factor K can be determined. Fig. 5 is a block diagram to illustrate the process to determine the size of unicast/multicast segment followed by sectional partition. If the first segment is partitioned with the unicast segment size determined through above process and the stream size is partitioned again by multicast segment size, the entire stream can be partitioned with the number of selected partition factors. For example, the result of sectional partitioning the entire TS stream by selecting the partition factor as 4 is shown as Fig. 6.

Fig. 7. The result of additional sectional partitioning the divided unicast segment by selecting the partition factor as 4

The result by applying the same partition factor in the way of Fig. 6 for the partitioned unicast segment is shown in Fig. 7.

When the additional section partitioning is performed, one segment for transmitting unicast and six segments for transmitting multicast can be obtained. Also, as we can see from Fig. 6, the size of partitioned unicast segment is always bigger than or same as the size of multicast segment. The reason is because the proposed delivery system has the structure of transferring the unicast segment using the unicast method to enable viewers to watch immediately and then transferring repeatedly using the multicast method for the segments afterwards. Therefore, the entire data of the multicast segment 4 can be received before one finishes watching the contents transmitted as unicast segment even if the viewer joins the multicast group at any location of the multicast segment 4.

3.2 Delivery Scheduling

It is necessary for an extra scheduling data in addition to the existing EPG (Electronic Program Guide) based information to transmit the partitioned multicast segments efficiently.

Before starting transmission from the sever side, the scheduling information which is necessary for receiving the segment is transmitted to the client side and accordingly, the device at the client side (e.g., IP-STB) starts the reception. The scheduling information is sent immediately as the text after the contents have been requested by the viewer.

The information which is necessary for scheduling is each of the multicast address and the port number to transfer the multicast segments. Table 1 is an example of scheduling table in case of selecting the partition factor as 4. The scheduling table starts transferring six multicast segments through four links.

TABLE 1
The Scheduling Information Table

Delivery Type	Multicast IP Address	Port Number
Multicast 1	224.24.1.12	1001
Multicast 2	224.24.1.12	1002
Multicast 3	224.24.1.12	1003
Multicast 4	224.24.1.12	1004

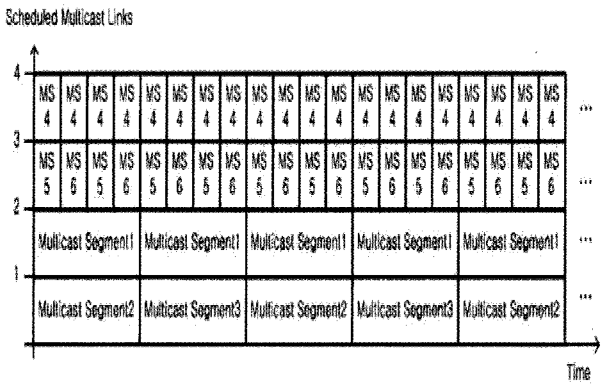


Fig. 8. The multicast segment 4 for the first link, 5 and 6 for second link, 1 for third link and 2 and 3 for fourth link are transmitted using multicast method

The multicast segment 4 for the first link, 5 and 6 for second link, 1 for third link, and 2 and 3 for fourth link are transmitted by the multicast method. At this time, the corresponding stream is transferred repeatedly at each link and the null-packet which the PID is '0x1FFF' is inserted at the back of each segment during the repetition of each segment. The reason is because while receiving the segment at the client side, it is used for the role of classifying each segment if not the continuous reception of Continuity-counter and PTS.

4 Client Side System Operation

4.1 Receiving Procedure

In order to transmit the partitioned segments, the data is encapsulated to the RTP header at the Transport Layer and the RTP header may add a stationary header of 12 bytes and expansion headers having a variable length depending on the type to be transmitted. Like in

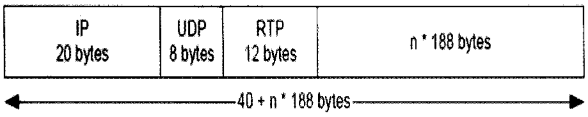


Fig. 9. Data is encapsulation with IP header

Fig. 9, the n number of transport packet charge load is transferred using the 12 byte RTP fixed header format and this is received once again from the server side by being encapsulated as 8 byte UDP header and 20 byte IP header [9].

Once the contents are requested at the EPG level of user's STB, the multicast scheduling table is received first. Next, the reception of unicast segment is started by the unicast listening and the join message is sent to the multicast router in order to receive each multicast segment according to the route of scheduling table at the same time. Especially after joining the multicast segment, the table of TS packets is organized and then the corresponding segment received from each link is stored by the TS packet unit. Also, the PTS confirmation task to reorganize the stream of the packet unit being received gets started.

4.2 Segment Merging

As shown in Fig. 10, the packet is received from the TS packet analyzer for the first time. The TS Sync Byte (0x47) which is the TS header information is checked to confirm the status of a new TS packet. Next, the PID and the Continuity Counter values are saved at the TS packet reception table. Then the Payload Unit Start Indicator value is checked. Subsequently, the Adaptation field control, Stream id and PTS DTS flag values are checked. The adaptation field length is an 8-bit field specifying the number of bytes in the adaptation field immediately following the adaptation field length. The value 0 is for inserting a single stuffing byte in a Transport Stream packet. When the adaptation field control value is '11', the value of the adaptation field length shall be in the range 0 to 182. When the adaptation field control value is '10', the value of the adaptation field length shall be 183. PTS DTS flags is a 2-bit field. When the PTS DTS flags field is set to '10', the

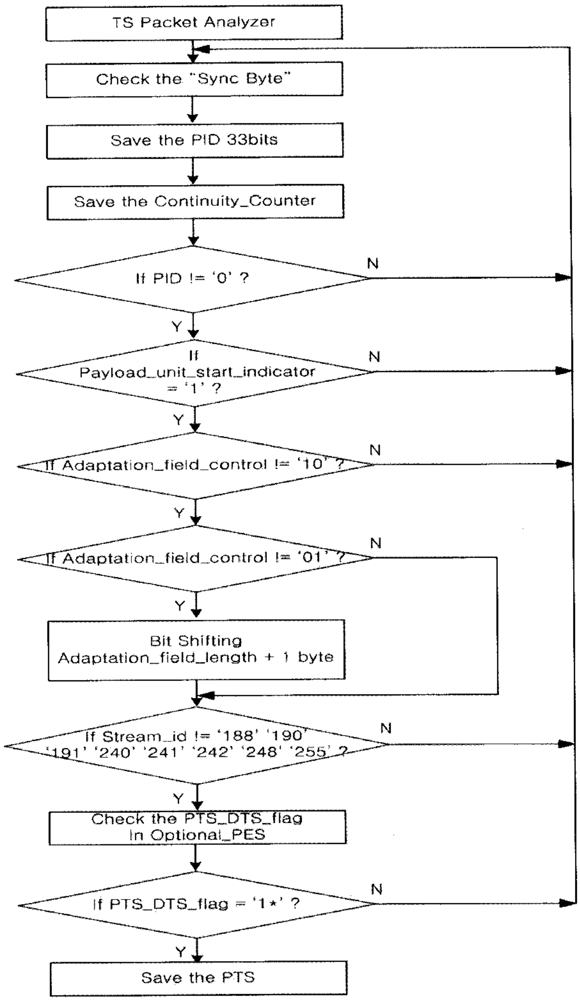


Fig. 10. Flow of obtaining the PTS field information

PTS fields shall be present in the PES packet header. When the PTS DTS flags field is set to '11', both the PTS fields and DTS fields shall be present in the PES packet header. When the PTS DTS flags field is set to '00' no PTS or DTS fields shall be present in the PES packet header. The value '01' is forbidden.

Through the processes above, the PID, the Continuity Counter and the PTS information of received packets are found and the table of being stored the corresponding values can be organized. As shown in Fig. 11(a), the first packet having all of the PID, the Continuity Counter and the PTS information is selected as the reference packet at the TS packet reception table. The purpose of the reference packet is to be able to recognize the overlapping of packets because the stream being received along the corresponding link is repetitive. Therefore, if the packet which is same in comparison with the reference packet is received, the IGMP leave message is sent to stop to receive packets at corresponding link immediately. The Fig. 11(b) shows the rearrangement of the received TS packets. The rearrangement in the order of descending series is attempted according to the PTS information at the TS packet reception table. If the reversion of the PTS has occurred around the reference packet in which the PID is null packet of '0x1FFF', the order of TS packet is rearranged by the position unit of this null packet. For example, in case the reversion of the PTS has occurred around the first null packet after receiving the TS packet, the value from the TS packet saved first to the corresponding null packet is aligned at the bottom of the stored table. Or, in case the reversion of the PTS has occurred around the last null packet of the stored TS packet information table, the value from the corresponding null packet to the last TS packet is rearranged to the top of the corresponding table. The discontinuity of Continuity Counter by each PID is checked at the arranged table.

5 Simulation Results

In unicast, as you increase the number of clients, you linearly increase the network bandwidth used and cost since you generate a separate copy of data to each recipient. The extra bandwidth required may be in excess of some of your communication links.

This means unicast does not easily scale to large numbers of recipients. Broadcast transmissions forward data packets to all portions of the network wasting bandwidth when there are few intended recipients. Multicast transmission sends a single multicast packet addressed to all recipients. It provides efficient communication and transmission, optimizes performance, and enables truly distributed applications [11].

We present the results of simulating the proposed a new real-time multimedia streaming architecture to enable viewers to watch the video right away and to use the bandwidth efficiently by dividing the entire TS partially.

	0x0271	8		
	0x0270	2		
	0x0270	3		
REF	0x0270	4	0x1c48f487d	TS_buffer[0]
	0x0271	9		TS_buffer[1]
	0x0270	5		TS_buffer[2]
	0x0270	6		TS_buffer[3]
	0x1FFF			TS_buffer[4]
	0x0270	14		TS_buffer[5]
	0x0270	15	0x1c46e787d	TS_buffer[6]
	0x0270	0		TS_buffer[7]
	0x0271	4		TS_buffer[8]
	0x0270	1		TS_buffer[9]
	0x0270	2		TS_buffer[10]
	0x0270	3		TS_buffer[11]
	0x0270	4		TS_buffer[12]
	0x0270	5		TS_buffer[13]
	0x0271	5	0x1c46e887d	TS_buffer[14]
	0x0270	6		TS_buffer[15]
	0x0000	8		TS_buffer[16]
	0x0270	7		TS_buffer[17]
	0x0270	8		TS_buffer[18]
	0x0271	6		TS_buffer[19]
	0x0270	9	0x1c47e887d	TS_buffer[20]
	0x0270	10		TS_buffer[21]
	0x0270	11		TS_buffer[22]
	0x0270	12		TS_buffer[23]
	0x0270	13		TS_buffer[24]
	0x1FFF			TS_buffer[25]
	0x0271	7	0x1c47e8f7d	TS_buffer[26]
	0x0270	14		TS_buffer[27]
	0x0270	15		TS_buffer[28]
	0x0000	9		TS_buffer[29]
	0x0270	0	0x1c47f487d	TS_buffer[30]
	0x0270	1		TS_buffer[31]
	0x0271	8	0x1c47f412d	TS_buffer[32]
	0x0270	2		TS_buffer[33]
	0x0270	3		TS_buffer[34]
	0x0270	4	0x1c48f487d	

(a)

	0x0270	14		TS_buffer[5]
	0x0270	15	0x1c46e787d	TS_buffer[6]
	0x0270	0		TS_buffer[7]
	0x0271	4		TS_buffer[8]
	0x0270	1		TS_buffer[9]
	0x0270	2		TS_buffer[10]
	0x0270	3		TS_buffer[11]
	0x0270	4		TS_buffer[12]
	0x0270	5		TS_buffer[13]
	0x0271	5	0x1c46e887d	TS_buffer[14]
	0x0270	6		TS_buffer[15]
	0x0000	8		TS_buffer[16]
	0x0270	7		TS_buffer[17]
	0x0270	8		TS_buffer[18]
	0x0271	6		TS_buffer[19]
	0x0270	9	0x1c47e887d	TS_buffer[20]
	0x0270	10		TS_buffer[21]
	0x0270	11		TS_buffer[22]
	0x0270	12		TS_buffer[23]
	0x0270	13		TS_buffer[24]
	0x0271	7	0x1c47e8f7d	TS_buffer[26]
	0x0270	14		TS_buffer[27]
	0x0270	15		TS_buffer[28]
	0x0000	9		TS_buffer[29]
	0x0270	0	0x1c47f487d	TS_buffer[30]
	0x0270	1		TS_buffer[31]
	0x0271	8	0x1c47f412d	TS_buffer[32]
	0x0270	2		TS_buffer[33]
	0x0270	3		TS_buffer[34]
REF	0x0270	4	0x1c48f487d	TS_buffer[0]
	0x0271	9		TS_buffer[1]
	0x0270	5		TS_buffer[2]
	0x0270	6		TS_buffer[3]

(b)

Fig. 11. TS packet reception sample table

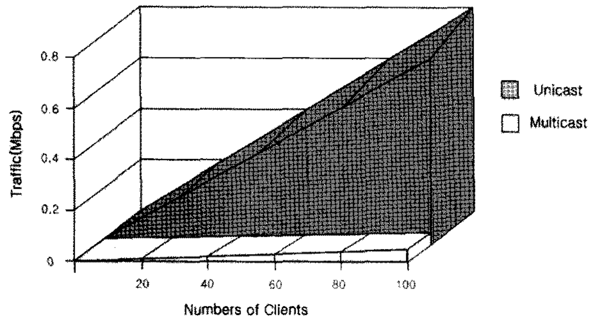


Fig. 12. Multicast traffic compared to unicast traffic. Example: audio streaming, all clients listening to the same 8 Kbps audio

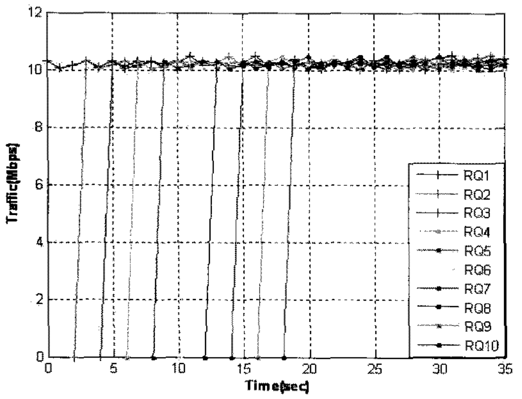
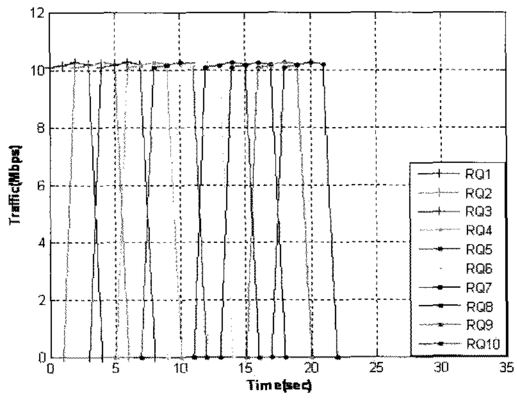
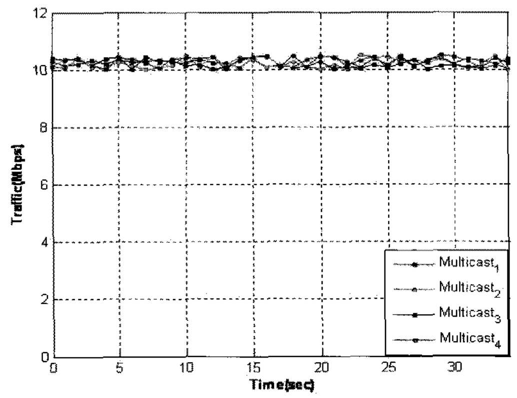


Fig. 13. In case of using only the traditional unicast methods



(a)



(b)

Fig. 14. The new proposed method by using of proposed dynamic Multicast

Assume that 10 people have requested the service with 2 second intervals, the bit rate of sample stream is 10Mbps and the running time is 60 seconds. In case of using only the existing unicast method, we can see that the number of bandwidth being required gets steadily increased in proportion to the number of users requesting the contents whenever the services are requested while the total contents play. We can see a linear increase of traffic in 2 second intervals with the existing method as shown in Fig. 13. On the other hand, the new proposed method has a great advantage of returning to a fixed level quickly after using the additional bandwidth for as much as 1/16 time of contents whenever there is service request based on planned multicast method.

The new proposed method has improved a performance about 350 percent compared with using only the existing unicast method when the performance analysis is assumed that a special case has the service request by 10 people with 2 second intervals. Also, a new proposed real-time multimedia streaming architecture improves the efficiency of bandwidth even more as the amount of service request gets greater.

6 Conclusions

This paper proposes a new method of real-time multimedia streaming architecture which efficiently uses the bandwidth and immediately enables the video service after the viewer's requests. This proposed method for TVoD is possible to convert quickly the unicast methods to the multicast methods through partial

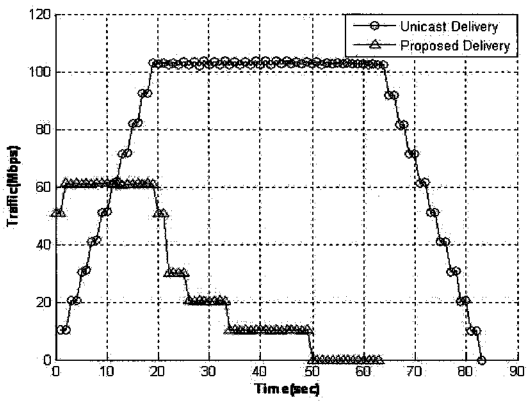


Fig. 15. The performance comparison between traditional unicast delivery and proposed dynamic multicast delivery

segments delivery of MPEG-2 Transport Stream over IP networks. Also, a new real-time multimedia streaming architecture uses defined bandwidth to deliver the contents. When a new viewer requests the contents, bandwidth quickly returns to the defined level after using the unicast segment size of the entire content. Eventually, the proposed a new real-time multimedia streaming architecture provides the real-time services in which the viewers are able to watch TV right away after requesting contents through an efficient bandwidth management.

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