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Internet Protocol TV (IPTV) has recently gained momentum as one of the key applications in the telecommunications market. Most researchers believe that IPTV service represents a key opportunity for operators around the world to benefit from video delivery over IP networks. In this paper, we investigate the quality of Service (QoS) and Quality of Experience (QoE) techniques for IPTV transmissions. Particularly, we address two technical challenges in IPTV system: error concealment and channel zapping. To overcome the above challenges, we extend the function of the set-top box (STB) and the home network gateway. As for channel zapping, we propose to employ peer-to-peer communication as a supplement in the channel switching period. As for error concealment, we develop an approach of AL-FEC based adaptive error recovery. Since our study addresses core problems in the IPTV network, we believe it has practical significance to the industrial community.

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QoS/QoE Techniques for IPTV Transmissions

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Abstract — **Internet Protocol TV (IPTV) has recently gained momentum as one of the key applications in the telecommunications market. Most researchers believe that IPTV service represents a key opportunity for operators around the world to benefit from video delivery over IP networks. In this paper, we investigate the Quality of Service (QoS) and Quality of Experience (QoE) techniques for IPTV transmissions. Particularly, we address two technical challenges in IPTV system: error concealment and channel zapping. To overcome the above challenges, we extend the function of the set-top box (STB) and the home network gateway. As for channel zapping, we propose to employ peer-to-peer communication as a supplement in the channel switching period. As for error concealment, we develop an approach of AL-FEC based adaptive error recovery. Since our study addresses core problems in the IPTV network, we believe it has practical significance to the industrial community.**

I. Introduction

Due to the growing deployment of FTTH (Fiber To The Home) and the start of Next Generation Network (NGN) in Japan, IPTV (Internet Protocol TV) services are becoming popular and are expected to rapidly expand in the near future. IPTV is defined as a service that includes multimedia services such as TV, video, audio, text, graphics, and data over IP based networks by the ITU-T (International Telecommunication Union – Telecommunication standardization Sector). IPTV supports QoS (Quality of Service), QoE (Quality of Experience), security, interactivity and level of reliability required on top of a managed IP network and it is expected to be an important communication and broadcasting convergence service for the future ubiquitous society. Currently, several national and international standards organizations have been actively working on IPTV standardization.

With the release of specifications for IPTV services by the IPTV Forum Japan and subsequently by the ITU-T [1], [2], we are actively involved in the design and implementation of compliant IPTV terminals such as set top boxes (STB) to support IPTV services for NGN communication networks. The implementation is based on a middleware platform simplifying the addition of new IPTV applications and enhancing extensibility. In addition, it can also support future services such as bi-directional (interactive) home services such as content downloading, in-house redistribution services, and home security. Figure 1 shows the structure of an IPTV system. The optical access network terminal equipment in the figure is also called home gateway.

Currently, we are investigating several advanced techniques to further enhance the IPTV-STB design and performance. Two of our studies are discussed in this paper. As it is well known, one of the main problems for deploying high quality IPTV services is noticeable artifacts in the video caused by packet loss. In current IPTV systems, to combat the packet loss, application layer forward error correction (AL-FEC) has been adopted. However, for some homes, especially those using wireless networking, additional loss will occur in the home network. The packet loss at the end devices could be more severe than expected sometimes. In this case, packets reaching the home gateway from public IP network are sufficient to recover all the source information, but IP-STB can not do so because of considerable packet loss with home network. To solve this problem, we propose a scheme, which adds QoS enhancement components to both home gateway and IP-STB and allows IP-STB to send home gateway feedback signals indicating the condition of the home network. If the home network has severe packet loss, the QoS enhancement components will be activated to utilize extra error correcting scheme.

Another problem for IPTV services is a relatively long channel switching time or channel zapping time. In the traditional broadcasting services, the terminal can immediately display the program when a user changes the channel being watched because the STB receives all of the channels regardless of whether they are being used or not. However, the IPTV service cannot transmit all the channels at the same time due to the lack of network bandwidth. Therefore only a part of channels are immediately available at the STB, and some delay is inevitable to display a new channel when it is not available at the STB. In the literature, this kind of delay is called channel zapping time. Reducing channel zapping time is crucial to the successful deployment of IPTV. In this paper, motivated by the fact that unicast has much less establishment overhead than multicast, we make use of peer-to-peer communication as a supplement in the channel switching period to reduce channel zapping time.

The rest of the paper is organized as follows. We first introduce the current development of IPTV network and STB in Section II. Then we investigate QoS enhancement for home network in Section III. Next channel zapping time is studied and the approach of supplemental peer-to-peer communication is proposed in Section IV. Finally, we conclude the paper in Section V.

II. Development Background of IPTV

A. IPTV Services in Japan

In Japan, NGN-based IPTV service was launched commercially in March 2008. This service provides TV content at HD resolution including IP broadcasting (IP multicasting of linear TV programs), VOD (Video on Demand) and from May 2008 IP retransmission of digital terrestrial broadcasting. IPTV service enables a true communication and broadcasting convergence service for the future ubiquitous network society. On the other hand, video distribution services using non-NGN regular Internet are becoming popular and video content downloading services are also planned by several companies.

B. IPTV Standardization

For the international standardization of IPTV, FG-IPTV (Focus Group on IPTV) of ITU-T started the study of the specifications

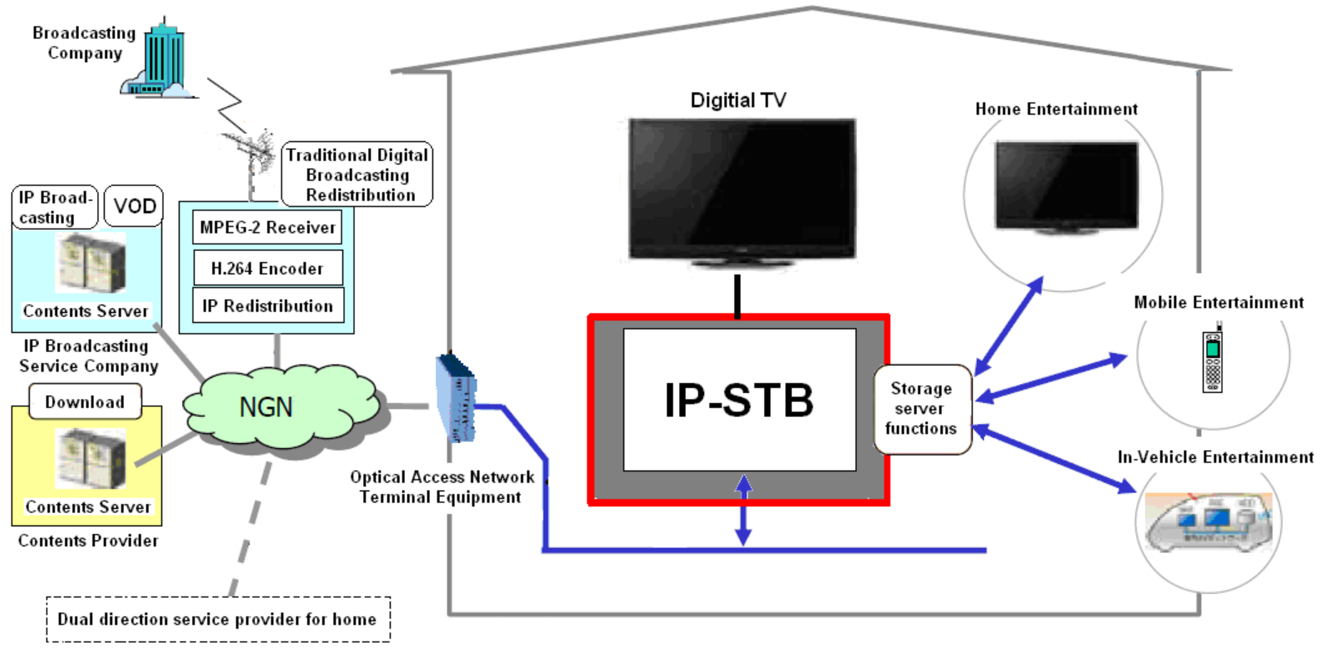

Figure 1. Structure of IPTV system

from July 2006 and the base technical specification documents were released in December 2007. IPTV-GSI (Global Standard Initiative) was established in January 2008 and it has been working on draft recommendations based on the documents prepared by FG-IPTV. The ITU-T Study Groups coordinated under the IPTV-GSI have completed several key recommendations.

In Japan, the Telecommunication Technology Committee (TTC) established an IPTV expert committee and it has worked closely with the Association of Radio Industries and Broadcast (ARIB) and the IPTV Forum Japan to collaborate with the ITU-T IPTV-GSI.

IPTV Forum Japan was established as a limited liability organization by communication, broadcasting and consumer electronics manufacturers in May 2008. Technical specifications integrate various existing standards into a common specification available to the public since September 2008.

Japanese broadcasters including NHK and commercial TV operators support terrestrial digital broadcasting retransmission and as result technical guidelines for IP retransmission technology and management requirements were developed.

C. Features of Current IP-STB

In compliance with IPTV Forum specifications, our IP-STB supports the following services: IP broadcasting using H.264/AVC coding technologies, VOD, and retransmission of terrestrial digital broadcasting.. The IP-STB has the following features:

1) Compliant with IPTV Forum technical specifications for the domestic terminal
2) H.264/AVC video compression technology to support high-definition video
3) Based on a SoC (System On Chip) to comply with multiple encoding requirements
4) Middleware platform supporting application extensibility
5) Improved ease of use by sharing the same UI (User Interface) with current digital broadcast receiver.

III. AL-FEC Based QoS Enhancement for Home Network

In this study, we consider an error-prone home network, where IP-STB is connected to home gateway via wireless links. Especially, we address the scenario that packet loss rate suffered by IP-STB cannot be totally recovered by the error control codes embedded in the original source packets. To solve this problem, we propose an approach of AL-FEC based adaptive error recovery.

A. Basics of AL-FEC

The AL-FEC protocol is a layered protocol based on a combination of the following two forward error correction (FEC) codes: (1) a simple packet-based interleaved parity code (SMPTE 2022-1 code); (2) the Raptor code.

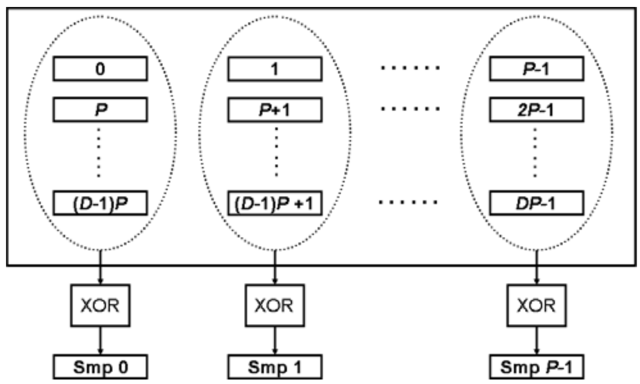

Figure 2. Packet-based interleaved parity code

Packet-based interleaved parity code is described in Figure. 2. Data is organized into columns and rows with the block size of $D \times P$. The data blocks in the figure are populated from left to right and top to bottom. Once a matrix is filled, an exclusive or (XOR) function is executed on all of the data in one column to generate an SMPTE repair packet.

Raptor codes are a powerful rateless code proposed in [3]. Given a message consisting of k source symbols, a Raptor code encodes it into a potentially limitless sequence of encoding symbols. The knowledge of any k or more encoding symbols allows the message to be recovered with some non-zero probability. When the number of received symbols is only slightly larger than k, the probability that the message can be recovered becomes very close to 1. Let binary matrix $\mathbf{A}'$ be the decoding matrix of the Raptor code. If $\mathbf{A}'$ has full rank, all source symbols can be recovered.

Raptor codes may be systematic or non-systematic. Systematic Raptor codes are used in IPTV standards, where the symbols of the original message are included within the set of encoding symbols. IPTV standards adopt hybrid decoding, which involves 3 steps:

Step 1: SMPTE 2022-1 decoding

In this step, the repair packets generated according to SMPTE 2022-1, together with the received source packets, are processed as usual to recover source packets.

Step 2: Raptor decoding

In this step, if source packets are still missing, then the repair packets generated according to Raptor, together with the received source packets and any source packets which were recovered in Step 1, are processed using the standard Raptor decoding procedures.

Step 3: Hybrid decoding

In this step, if source packets are still missing, then remaining (unprocessed) SMPTE 2022-1 packets are converted to a form in which they can be added to the Raptor decoding process, and Raptor decoding is then continued.

B. Proposed Adaptive Error Recovery

As shown in Figure 3, our approach introduces two simple modules in home gateway, i.e., QoS Controller and QoS Enhancement Encoder. IPTV standards state that audio/video streams are transported using RTP protocol. In our design, we employ a sister protocol of RTP, namely RTCP, to monitor the packet lose rate of a given STB. Once the STB finds its packet loss rate goes out of a certain threshold, it will exchange RTCP messages with the QoS Controller in home gateway. Then, the Controller will activate QoS Enhancement Encoder, which is described in Figure 4. It has a layered structure. The first layer enhancement is to add more protection to the SMPTE repair packets. And the second layer enhancement is to add more protection to the Raptor packets. When QoS Enhancement Encoder is activated, one or more appropriate layers are selected based on the feedback signal from the STB.

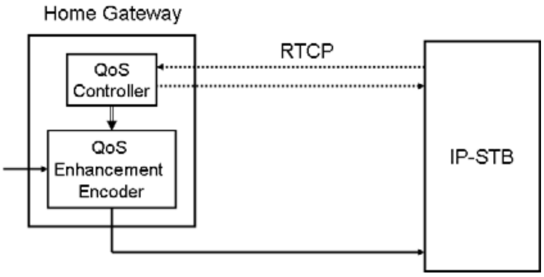

Figure 3. Structure of QoS enhancement components

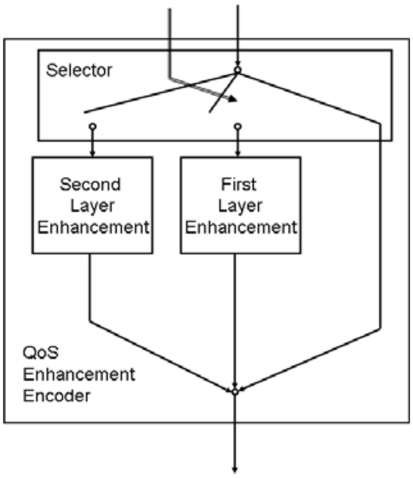

Figure 4. Structure of QoS Enhancement Encoder

1) First layer enhancement

In the first layer enhancement, we make use of SMPTE 2022-1 repair packets to counteract the packet loss in home network. As shown in Figure 2, in SMPTE 2022-1 encoding process, a source block with $D \times P$ packets is arranged into a *D* by *P* array and we will have *P* repair packets. Let $P = f \times r$. In the first layer enhancement block, a buffer collects SMPTE 2022-1 repair packets and checks their packet indices in their header. Repair packets can be divided into *f* groups with *r* packets each. The index of the *j*th packet in group *i* can be represented by $i+(j-1)f$, where $i=1, 2, \ldots, f$ and $j=1, 2, \ldots, r$. If all the *r* packets in the same group have been stored in the buffer, they are exclusive-ORed to generate a new packet. Figure 5 illustrates the first layer enhancement block for a source block with 18 source packets, where D = 3 and P = 6, i.e., there are 6 repair packets transmitted. Suppose all the repair packets are received and stored in the buffer. Set f = 3 and r = 2. Then based on our design, Smp 0 and Smp 3 go through an XOR block to generate a new packet Smp 6. Smp 1 and Smp 4 are combined to generate Smp 7 and Smp 2 and Smp 5 are combined to generate Smp 8.

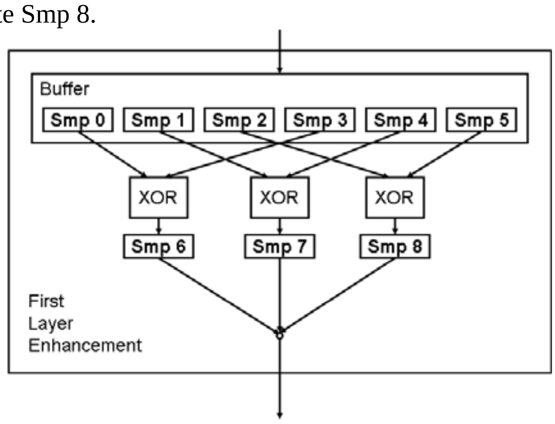

Figure 5. An example of the first layer enhancement

2) Subsequent layer enhancement

If the condition of home network is severe, besides the first layer enhancement block, a second layer enhancement block is activated as well. The second layer enhancement block also uses a buffer to store the received Raptor encoded packets (including source and repair packets). Suppose the buffer arranges the received Raptor packets in a *Q* by *Y* array. Then the *Q* Raptor packets in the same column are XORed to generate a new packet. We call it enhancement packet. This arrangement can effectively combat the burst errors introduced by home network. Since the Raptor packets do not necessarily have the same size, the smallest size of the packets that participate in the XOR operation is chosen as the size of the corresponding enhancement packet. For the Raptor encoding packets with larger size, their extra symbols are not involved in the XOR operation.

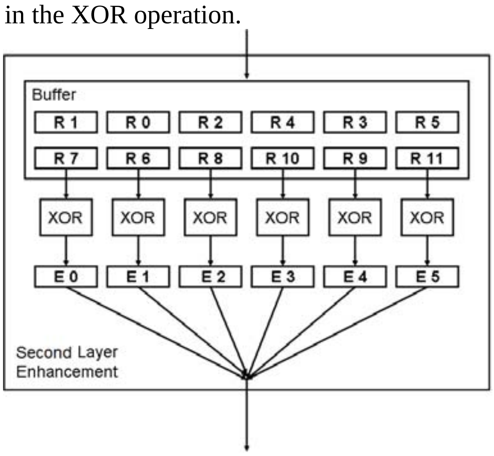

Figure 6 An example of the subsequent layer enhancement

Figure 6 describes the structure of a subsequent layer enhancement block. The received Raptor encoding packets are stored in a buffer, which has a size of 12 in this example and the received Raptor packets are arranged into a 2 by 6 array. The Raptor packets in the same column go through an XOR block to generate an enhancement packet. Then 6 enhancement packets are sent to the set-top box sequentially.

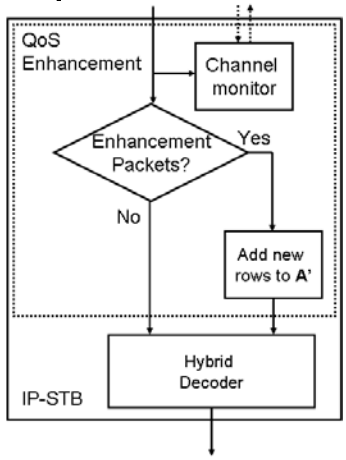

Figure 7. Structure of IP-STB with QoS enhancement.

Figure 7 shows the structure of IP-STB with QoS enhancement. It includes a channel monitor, which monitors the channel condition of the home network and communicates with home gateway using RTCP messages. For decoding, since the packets generated by the first layer enhancement block are compatible with the original SMPTE repair packets, they can directly be decoded by the original SMPTE decoder. For Raptor codes, if the received packet is an enhancement packet, new rows should be added to the decoding matrix $\mathbf{A}'$; otherwise the decoding matrix remains the same. Hybrid decoder works in the same way as before.

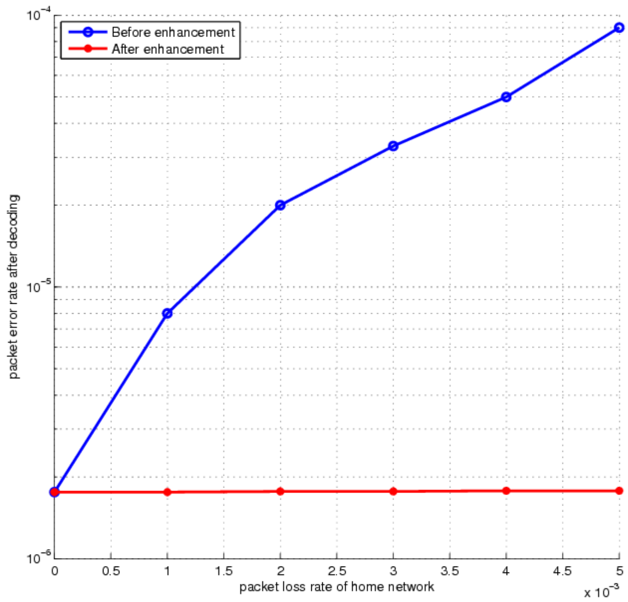

Figure 8 Performance comparison of IPTV system with and without QoS enhancement.

Figure 8 depicts the packet error rate of the original system and the system with our proposed QoS enhancement scheme. The number of symbols in each source block is 800 and every 4 symbols form a packet. Set D = 20, P =10 and the packet loss rate of public IP network to 0.001. The AL-FEC codes are constructed as in [4]. As shown in Figure 8, with the increase of packet loss rate of home network, the performance degrades quickly. For QoS enhancement of home network, we apply both first layer and subsequent layer enhancement. We can see the performance after enhancement is kept almost the same as the case of perfect home network.

VI. Rapid Channel Zapping for Multicast systems

The channel zapping time generally consists of four parts: 1) command processing time, 2) network delay, 3) STB jitter buffer delay, and 4) video decoding delay [5]. Amongst them, network delay is the passing time that a requested stream is arrived after the transmission of Internet Group Management Protocol (IGMP) Join message. According to the test results from Agilent Technologies, the multicast operation time for 1000 users varies from 0.9 seconds to 70 seconds when changing channels [6]. Therefore, IP multicast operations of IGMP are considered as one of the major reasons contributing to the channel zapping time.

Existing work mainly addressed the enhancement on current multicast protocols to reduce the delay caused by IGMP. For example, in [7] Cho et al. proposed a new approach of making adjacent multicast groups of current channel come to home gateway in advance. As a result, when IP-STB requests join to an adjacent channel, home gateway can forward the corresponding multicast stream immediately. Also, in [8] Kim et al. proposed an efficient way to diminish the General Query Interval (GQI) of IGMP parameters.

Since IGMP processing delay is identified as a major contributor to IPTV channel zapping time, to reduce it we propose an approach to establish a peer to peer fast channel from neighboring nodes to channel switching STB.

A. Introduction to IP Multicast

IP Multicast is a technique for one to many communication over an IP infrastructure. It scales to a larger receiver population by not requiring prior knowledge of who or how many receivers there are. Multicast uses network infrastructure efficiently by requiring the source to send a packet only once, even if it needs to be delivered to a large number of receivers. The nodes in the network take care of replicating the packet to reach multiple receivers only where necessary. The most common low-level protocol to use multicast addressing is UDP.

An IP Multicast group address is used by sources and the receivers to send and receive content. Sources use the group address as the IP destination address in their data packets. Receivers use this group address to inform the network that they are interested in receiving packets sent to that group. For example, if some content is associated with group 239.1.1.1, the source will send data packets destined to 239.1.1.1. Receivers for that content will inform the network that they are interested in receiving data packets sent to the group 239.1.1.1. The receiver "joins" 239.1.1.1. The protocol used by receivers to join a group is called the Internet Group Management Protocol (IGMP).

Once the receivers join a particular IP Multicast group, a multicast distribution tree is constructed for that group. The protocol most widely used for this is Protocol Independent Multicast (PIM). It sets up multicast distribution trees such that data packets from senders to a multicast group reach all receivers which have "joined" the group. E.g. all data packets sent to the group 239.1.1.1 are received by receivers who joined 239.1.1.1. There are many different flavors of PIM: Sparse Mode (SM), Dense Mode (DM), Source Specific Mode (SSM) and Bidirectional Mode (Bidir) [Also commonly known as Sparse-Dense Mode (SDM)]. Of these PIM-SM is the most widely deployed as of 2006 [update]; SSM and Bidir are simpler and more scalable variations developed more recently and gaining in popularity.

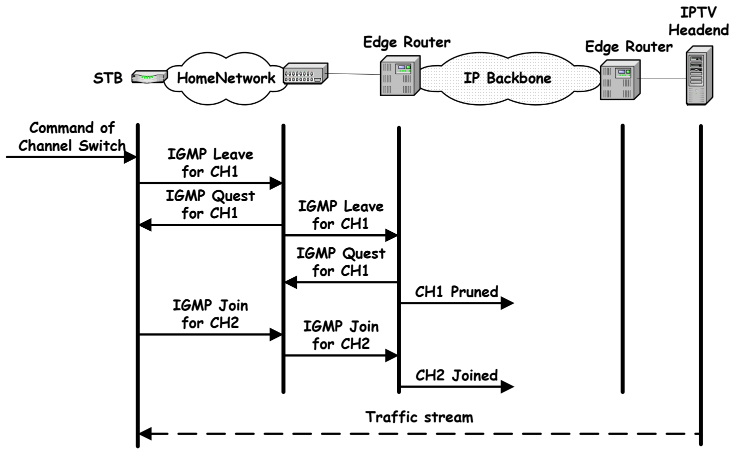

Figure 9. Signaling process of IPTV channel switch (CH1→CH2)

IPTV uses IP multicast techniques to reduce overall bandwidth requirements for delivering contents and services. Channel zapping time of IPTV services is considerably affected by IP multicast operations of IGMP, such as joining and leaving multicast groups that serve the IPTV channels, which were not needed for the traditional all-channel broadcasting TV systems. Current zapping time does not satisfy the quality of experience (QoE) of users, due to the complex signaling process (as shown in Figure 9). Above fact reveals that IP multicast has the con of long IGMP initialization delay, despite its advantage of high bandwidth efficiency.

B. Rapid Channel Zapping based on Peer to Peer Fast Channel

To overcome the problem of IGMP initialization delay, we introduce peer-to-peer communication as a supplement in the channel switching period (see Figure 10). We assume that a TV viewer currently tuned at channel I (IP multicast address: 239.1.1.1) intends to switch to channel II (IP multicast address: 239.1.1.6). To this end, its STB sends both IGMP and peer-to-peer communication signaling messages. Here, IGMP messages head for IP multicast address: 239.1.1.1 to quit the current channel and IP multicast address: 239.1.1.6 to join the new channel.

In addition to IGMP messages, the channel switching STB also sends a probe message to the new multicast group addressed 239.1.1.6 to construct the peer to peer fast channel. All the group members at 239.1.1.6 will receive the probe message and calculate how many hops they are from the channel switching STB. Then, the replies are sent back to the channel switching STB only from the group member near to it (for example, 3 or 4 hops away) using private unicast IP address. Once channel switching STB receives the feedback information, it employs certain policy to choose one group member at 239.1.1.6 as the peer to peer partner. Subsequently, peer to peer fast channel will soon be established with a few more signaling messages between channel switching STB and the peer to peer partner. Using the established IP unicast fast channel, expected IPTV stream keeps being transmitted from peer to peer partner to channel switching STB until the data from native IP multicast arrive and are buffered well to play.

C. Performance Comparison

Figure 11 illustrates that IP multicast involves both IGMP and PIM signaling to switch IPTV channel. IGMP is used both by the client computer and the adjacent network switches to connect the client to a local multicast router. IGMP snooping is the process of listening to IGMP network traffic. IGMP snooping, as implied by the name, is a feature that allows a layer 2 switch to "listen in" on the IGMP conversation between hosts and routers by processing the layer 3 IGMP packets sent in a multicast network.

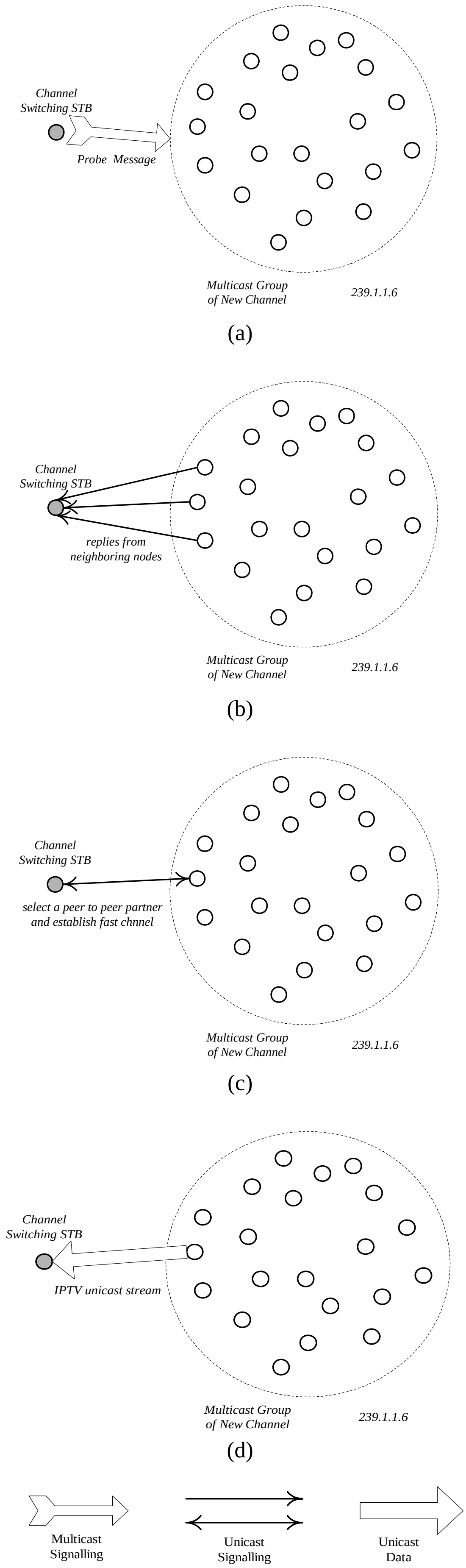

Figure 10. Establishment of peer to peer fast channel

After IGMP, PIM is then used between the local and remote multicast routers, to direct multicast traffic from the video server to many multicast clients. PIM is a family of multicast routing protocols that can provide one-to-many and many-to-many distribution of data over the Internet. The "protocol-independent" part refers to the fact that PIM does not include its own topology discovery mechanism, but instead uses routing information supplied by other traditional routing protocols such as Border Gateway Protocol (BGP).

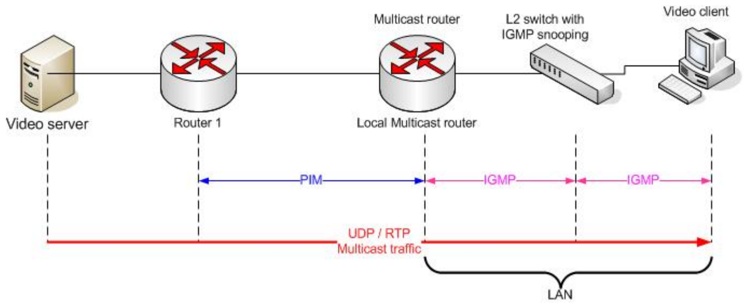

Figure 11. IP multicast network infrastructure and protocols

To change IPTV channel, the STB has to leave the current multicast group and join the new one. This process incurs unbearable delay because of the complexity of IGMP and PIM. As a proof, Agilent Technologies claimed that the multicast operation time for 1000 users varies from 0.9 seconds to 70 seconds when changing channels [6].

With our proposed approach of peer to peer fast channel, channel switching delay is significantly reduced, since we replace multicast signaling with unicast signaling. Unicast message exchange introduces only negligible delays. For instance, below lists the result of an ICMP test case achieved from a laptop experiment.

Ping statistics for 72.14.205.104 (www.google.com):
Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
Approximate round trip times in milliseconds:
Minimum = 25ms, Maximum = 60ms, Average = 35ms

With above results, we can conclude that the unicast signaling process would take less than half second to establish the peer to peer fast channel.

V. Conclusions

In this paper, we discuss the Quality of Service (QoS) and Quality of Experience (QoE) techniques for IPTV transmissions. In particular, we address two problems, error concealment and channel zapping for further enhancement of IPTV-STB. For error concealment, we use the AL-FEC based adaptive error recovery approach; for channel zapping reduction, we propose to employ peer-to-peer communication as a supplement in the channel switching period. Preliminary simulation and experiment results show that both methods can provide significant performance improvement with simple protocol and design change.

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