A Novel Scheme for a Fast Channel Change in Multicast IPTV System

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Abstract — Nowadays, television is the most popular media for informing and entertaining in the world. The generalization of IPTV brings advantages in terms of IP service convergence, but has also some drawbacks. One of the most them is the channel zapping delay. In this paper we show how it is possible, when IPTV is multicast, to improve the zapping delay and therefore the quality experienced by viewers, by using an additional stream with specific packet ordering rules.

Keywords-component: IPTV zapping, multicast, TSP, MPEG-TS.

I. INTRODUCTION

Since the fifties, Television (TV) has completely changed communication methods and concepts between people. Becoming the first media for informing and entertaining, TV contents can be transmitted by different ways. The one that we know today was at first broadcasted with airwaves. Currently, many possibilities exist depending on the carrier’s and the receiver’s type. Therefore, we can receive TV over cable connection, satellite, with DSL, over FTTH networks, on a TV, a computer or even a mobile phone screen.

With the massive usage of the Internet Technologies, many Internet Service Providers (ISP) propose, beside their Internet Access and the Voice over IP (VoIP) offer, a TV channels offer based on IP protocols. We speak therefore about IPTV channel and Triple Play service offer [1].

Viewers who get used airwaves reception for years to even watch High Definition TV (HDTV) do not only attach an importance to the video/audio quality but also to the delay required to watch the requested channel. Indeed, with the broadcast television technology, simultaneous reception of all programs at the viewer side leads to almost a very small zapping time involving mainly the time to decode the audio/video content by the TV receivers such as a Set Top Box (STB). However, in an IPTV Network System, all channels are no longer present at the same time at the viewer side; this will consume too much bandwidth. Therefore, the required time for channel change involves a Network Delay for making the channel available at viewer side and right after receiving the first packets of the requested channel, a Buffering Delay and a Decoding Delay are also needed to finally play out the requested channel.

Since viewers are able to change channels several times especially during commercial breaks or at program termination, we consider that a too long channel change delay could badly affect the Quality of Experience (QoE) of viewers [2].

In order to reduce the zapping time, we present a novel rapid channel zapping scheme for Multicast IPTV Systems. As in some previous works, to reduce the Buffering Delay, we propose to add an additional multicast stream to the main one. Additionally, we apply specific ordering rules to the packets of this secondary stream to reduce the First I-Frame Delay, which even leads to avoid some steps of the channel change process and improve the quality experienced by the viewers.

The remainder of the paper is organized as follows. In section II, we provide a detailed study of the IPTV delivering system in a multicast network. Section III is devoted to the presentation of our proposition and in section IV an evaluation of our proposition is presented. Finally, we conclude in section V and give some tracks about the future works to run.

II. BACKGROUND AND RELATED WORK

A. The IPTV Delivery System

In IPTV systems, Moving Pictures Experts Group (MPEG) algorithms are applied to compress the media content and minimize the bandwidth consumption of the TV channel. This kind of coding is based on the spatial and temporal redundancy of the pictures. So, an MPEG compressed video is divided into segments, called group of pictures (GOP) that contain a short part of the video stream, from a half second up to few seconds. Every GOP starts with an I-Frame, which contains an entire coded picture and is followed by a sequence of P/B-Frames. Those frames provide only incremental relative changes to the previous one and the coming picture. Since an I-Frame contains the entire picture information, it can be decoded independently from any other frame. That is why a STB can only start playing out a received media stream from the beginning of a GOP.

As presented in Fig. 1, the MPEG coded streams are transmitted in the MPEG-Transport Stream format (MPEG-
TS) [3], which is a communication protocol for audio, video and data. It allows multiplexing of digital video and audio flows and synchronizes the output. MPEG-TS streams are composed by TS Packets of 188 bytes long which are typically carried by groups of 7 in a single UDP/TCP packet. Each TS Packet is referring to a segment of a Program Elementary Stream (PES) such as video, audio or data content.

Unlike the unicast transmission, the multicast technology allows to send data in one time from a source to a selection of destinations. This technique saves bandwidth consumption compared to the unicast transmission but needs specifics routing and hosts management protocols.

As presented in Fig.2, in an IPTV multicast environment, channels change process is divided into several steps. If a viewer is receiving an audio/video data flow compressed with MPEG2 [4] or H.264 [5] format on a given multicast address and he/she wants to watch another channel, the STB sends IGMP messages [6] to firstly leave the current channel and then join the new requested one. After a Network Delay of several hundreds of milliseconds, which depends on the routing mechanisms and the network load, the STB starts receiving the first packets of the requested channel. To avoid under-run and compensate network jitter, a Jitter Buffer is first filled up. Then, when a GOP is starting, the Mpeg Buffer is filled up. This Buffering Delay is usually in a range of a half a second up to 2 seconds. The waiting time for the beginning of a GOP is called First I-Frame Delay (FID) and may take a whole GOP time size. Finally, when the STB buffer reaches the threshold limit, the STB starts the decoding of the received data and plays out the video/audio contents.

B. Schemes to reduce the Buffering/Decoding Delay

In order to reduce the FID, Y.Bejenaro and P.V.Koppel present in [7] the Multicast Assisted Zap Acceleration (MAZA) method. They suggest using additional sub-channels, time back shifted from the main requested IPTV channel. Depending on the instant of zapping, the IPTV receivers choose which sub-channel to join and have the minimal FID. When the buffer of the STB is full, a specific mechanism is proposed to switch back from the joined sub-channel to the main channel. Authors show that with this method, in an ADSL network and with 300ms time back shifted sub-channels, the FID was measured equal to 300ms for a 2 seconds GOP time size.

The Multicast Instant Channel Change (Multicast ICC) presented in [8] is a specific version of the Unicast Instant Channel Change (Unicast ICC) [9]. The basis of this mechanism is to join, additionally to the requested channel, a specific I-Frame flow. While the first I-Frame from this stream is displayed, the requested multicast stream is buffered up until the play out point. Then the IPTV receiver leaves the I-Frame stream and plays out the requested stream stored in its buffer. Compared to the Unicast ICC, the Multicast ICC has both lower display latency and bandwidth consumption. However, since this technique does not require the pre-buffering of a secondary I-Frame stream, it needs a dedicated and specific architecture. Indeed, the network latency and reception time variations lead to a lack in robustness of this solution.

Another scheme for rapid channel zapping which accelerates the data buffering of the IPTV receiver is proposed in [10]. In this solution, using an additional multicast stream instead of the conventional unicast burst reduces the Buffering Delay. A ratio packet-ordering rule for the secondary multicast stream was also proposed to adapt the timing variation of the channel change on each IPTV receiver. In a prototype system, authors confirm an average of 0.51 seconds reduction of the zapping delay during commercial breaks.

III. PROPOSED MECHANISM

A. The problem statement

As presented in Fig.2, after receiving the first packet of the requested channel, three steps are required until the start of decoding and playing out the requested channel. These steps are; - the jitter buffer filling, - the GOP start detection and finally - the mpeg buffer filling. Each delay of these steps is a part of the channel change latency. Previous works do not propose a global solution to reduce the delay of all these steps at once. Indeed, in [7] authors focus only on
reducing the FID, while in [8] and [9] the Instant Channel Change mechanisms use additional I-Frame only flow without considering network jitter impact. Finally in [10], a secondary accelerated multicast stream is joined together with the main stream leading to reduce the delay to fulfill the STB buffer, but without offering a direct solution to reduce the FID.

B. Proposed approach

To improve the channel zapping delay we propose a novel approach to reduce the buffering delays (jitter and mpeg) at the same time as the First I-Frame Delay.

As in [10], we add to the main stream a secondary stream, which is time back shifted from the main and is streamed with specific ordering rules that take into account the MPEG-TS packets contents types.

Fig. 3.a illustrates the scheme proposed in [10]. At each channel change instant, the IPTV receiver joins both the main and the secondary multicast streams of the requested channel. The secondary stream has a packet transmission ratio equal to r and a packet-ordering rule based only on the packet number in the main stream and the buffer size b of the IPTV receiver. If G’j is one of the packets of secondary stream G’, for any j=1,2,…,r and for d=⌈b/(r+1)⌉, the ordering rule proposed in [10] is expressed by:

\[ G'_{ij} = G_i - jd \] (1)

Once the IPTV receiver has stored the b packets upon buffering, the secondary stream is left and the buffered packets are sorted based on their ordering number to be decoded and then displayed.

In this scheme, authors did not pay attention to the type of the content of each packet. Indeed, as showed in Fig. 3.a, the buffer can be filled up without any I-Frame Header packet, which is mandatory to start the decoding of a GOP. In those cases, as what the evaluation section of [10] did show, this packet ordering rule does not reduce the zapping time compared to the standard case.

To eliminate those situations, we choose to take into account the content’s type of each TSP Packet in the packets ordering rules of the secondary stream as presented in Fig. 3.b. Therefore we define new policies to stream out the secondary flow:

1. With each packet of the main stream G, r packets of the secondary stream G’ are sent. These r+1 packets can therefore be seen as the elements of a Process Unit PU.

2. As presented in Fig 3.c, each packet Gj of the main stream carries n (n=7) TSP units and can be expressed by:

\[ G_i = \{ X_{i,1}^{n_i}, X_{i,2}^{n_i}, \ldots, X_{i,n_{i-1}}^{n_i}, X_{i,n_i} \} \] (2)

Where gn−i+1 ≤ gn−i+2 ≤ … ≤ gn−i+n ≤ gn are the GOP numbers that TSP X belongs to. Since the contents types are multiplexed, X may be a part of an I-Frame (I), B or P-Frame (B or P), an audio-stream (A) or data (D).

\[ X \in \{ I, P, B, A, D \} \] (3)

1. As the secondary stream is back time shifted from the main stream, a boot phase is mandatory. During this phase the Process Units can be expressed by:

\[ PU_i = \left\{ G_i, G_{i+1}, \ldots, G_{i+r} \right\} \] (4)

2. After this boot phase, the TSP elements of packets G’i of the secondary stream are picked up from the TSP elements of the packets Gi of the main stream with the new ordering rules presented in Fig.4. PUi can therefore be expressed by:
To emphasize the I-Frame contents, we prioritize them from all the other types. Therefore, when the buffer of an IPTV receiver is full and the packets are ordered, it should always contain I-Frame elements that should always be within the first packets. Additionally, the content of a given GOP is prioritized from the content of the previous GOP. This avoids under-run or video sequence jump when the buffer is filled up and the secondary stream is left. Fig. 3.c illustrates a simple case in which these rules are applied. So we can see that the packet #7 which contains B-Frame, Audio and Data elements, will be dropped while packets 1, 13 and 19 will be streamed out with packet 25 of the main stream (see Fig. 3.b).

To run our evolutions, we applied the rapid channel change with our new packets ordering rules in the IPTV system depicted in Fig.5. We setup two accelerator servers. On the first one, we applied the ordering rule used in [10], which is based only on the packet number of the main stream, where the secondary stream is streamed on G1. On the second one, we applied our new packet ordering rules based on the TSP types where the secondary stream is streamed on G2. Each of these secondary streams has a retransmission ratio r set to 3.

IV. EVALUATION

To run our evolutions, we applied the rapid channel change with our new packets ordering rules in the IPTV system depicted in Fig.5. We setup two accelerator servers. On the first one, we applied the ordering rule used in [10], which is based only on the packet number of the main stream, where the secondary stream is streamed on G1. On the second one, we applied our new packet ordering rules based on the TSP types where the secondary stream is streamed on G2. Each of these secondary streams has a retransmission ratio r set to 3.
Main stream ($G$) is an MPEG-2 compressed stream with a 4Mbit/s channel stream rate (around 500 RTP packets per second). Each GOP sequence has an IBBPPBBPBB structure and a mean size of 250±30 RTP packets. An I-Frame is meanly streamed in 60±10 RTP packets (around 24% of the GOP size).

Since in our approach the new packets ordering rules depend on the content’s type of the packets, the retransmission ratio $r$, the buffer and GOP sizes, as a first step, we ran some trials to define the optimal buffer size of the IPTV receiver for which the number of the missing TSP packets is the lowest and First I-Frame position in the buffer is the best (after reordering). After that, with this optimal buffer size, we compared our solution to the standard approach and the one proposed in [10].

Results presented in Fig.6 and Fig.7 were taken from the IGMP proxy. In Fig.6, for several buffer sizes set on the proxy, we plotted the mean value of the number of the missing TSP units per GOP. Based on 200 trials where IGMP-Join messages were randomly sent for each buffer size, Fig.6 shows clearly that the missing TSP units are located only in the first received GOP. That means that, when using our proposal, the video distortion may only appears during the first GOP time line (around a half a second in our tests cases). Additionally, we can see that, when the buffer size exceeds 1200 RTP packets, no more packets are lost. We clearly establish, as it can be seen in Fig. 8.a, that it depends on the ratio between the size of the GOP and the buffer. When each line of the buffer, with a width of $d$ packets (cf. section III.2) contains at least an entire GOP, our ordering rules will be based only on the GOP number of the TSP packets and so the packet’s number, as in [10].

Fig.7 shows the average number of the missing TSP units of the first received I-Frame and the First I-Frame Position after the buffer ordering. The top value of the missing TSP units of the 1st I-Frame is located where the proxy buffer has 900 RTP packets size. In this case, 32 % of the first I-Frame is missing. With this buffer size, each $d$ line of the buffer can contain 90% of the GOP. As depicted in Fig.8.b, this leads to have a minimal efficiency interval where we have in a Process Unit (cf. section III.2) at least two I-Frame TSP units pertaining to different GOPs. Whereas in the 10% left, each Process Unit, will have the TSP Header of an I-Frame, and another TSP content type of the same GOP. Whereas the 10% left, each Process Unit, will have the TSP Header of an I-Frame, and another TSP content type of the same GOP.

Fig.7 shows that the minimal values of missing I-Frame data in the first GOP are reached when the proxy buffer size is equal to 600 RTP packets or more than 1200 RTP packets. Indeed, as presented in Fig.8.c, in the maximum efficiency interval, each Process Unit used for creating the secondary stream should have at least one I-Frame with a different content’s type of the same GOP. That leads to have a minimal I-Frame missing packets and a good position of the First I-Frame after packet’s ordering.

Finally, when the buffer has a 500 RTP packets size, another peak is reached and the missing I-Frame packets reaches 12%.

Based on the results of Fig.7, we conclude that the optimal buffer size to have the minimal number of the missing packets and the best I-Frame position after the packet’s ordering is equal to 600 RTP packets. In other words, for the best results in terms of First I-Frame Delay and TSP missing packets number, the STB buffer size must be set at the maximum efficiency interval expressed by the test case c of Fig.8. Note that with this buffer size, more than 50% of the 200 trials have First I-Frame content at first buffer’s position.

In Fig.9 and Fig.10 are plotted the missing TSP Packets and I-Frame position for respectively $r=2$ and $r=4$. As we can see, for each value of the transmission ration $r$, the buffer sizes for a maximum and a minimal efficiency are still under the intervals expressed by the test cast b and c of Fig.8. This leads to conclude that the intervals expressed in Fig.8 and with which we can define the optimal buffer size are independent from the transmission ratio $r$ of the secondary stream.
With a transmission ratio \( r \) equal to 3 and an optimal buffer size of 600, we ran 200 new trials to measure the zapping delay and the video/audio quality on a STB emulator. As we can see it on Fig. 11, our solution reduces the zapping time by around 250 ms from the approach proposed in [10] and more than 1.3 second compared to the standard approach. Note that the maximum reduction compared to the proposed approach in [10] was measured equal to 0.5 second, which is the main value of the time size of the used GOP.

However, when the content of the buffer is finally displayed, the quality of the audio/video during the first GOP time line can be less than in the standard case. It varies from 70% up to 90% and may be under human perception [11].

V. CONCLUSIONS AND PERSPECTIVES

In this paper, we proposed an enhancement of a rapid channel change in IPTV multicast systems. To accelerate the buffering process and at the same time reduce the First I-Frame Delay, we add to the main stream a secondary multicast stream, which is a time back shifted from the main one. With new ordering rules that prioritize both the I-Frame and the content of the most recent GOP, we prove with our evaluations that for an optimal buffer size, our solution reduces the zapping time meanly by a half GOP time size. We also prove that our proposal is robust and can be used with different additional stream rates.

As perspective, we will propose an analytical model with which an operator can easily define the optimal buffer size for a given GOP/I-Frame size, and the transmission ratio of the secondary stream. Also, the available IPTV stream of the provider should be analyzed to know if they have mainly the same GOP/I-Frame size. If not, a dynamic STB configuration of the buffer size should be proposed too. Finally, the attachment point of the accelerator server should be also discussed.

REFERENCES