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About multiple paths video-streaming: State of the art

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Abstract: With the best effort service presented by today's Internet network, video streaming applications face an important challenge, especially with the aim to keep the quality perceived by the end user high enough. Several approaches had been explored in order to cope with this aspect. These approaches still suffer from many problems like scalability, complexity, and deployment.

Recently, many research groups started to explore a different promising idea, in order to overcome the limitations of the existing approaches: That is to stream the video over multiple paths, instead of the classical streaming method over a single path.

The objective of this report is to highlight the different existing approaches of multi-path video streaming. Furthermore, it presents the elements that have to be considered regarding video streaming, from a video point of view (video compression standard, type of streaming application), and from transmission point of view (positioning according to network architecture, different multipath streaming strategies).

Key-words: Video streaming, Internet, Multiple paths, Redundancy

(Résumé : tsvp)

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A propos de la transmission vidéo multichemins: Etat de l'art

Résumé : Avec le service 'best effort' fourni par le réseau Internet, les applications de transfert de vidéo sont confrontes à un defi important. Spécialement , pour maintenir une bonne qualité perçue par l'utilisateur. Plusieurs approches ont explorées, mais elles se confrontent à des problèmes, d'extensibilité, de complexité de mise en ouvre et de déploiement.

Actuellement, plusieurs groupes de recherche ont commencé à explorer une idée prometteuse pour surmonter les limites des approches existantes. L'idée est de transmettre des copies différentes du flux vidéo sur plusieurs chemins.

L'objectif de ce rapport est de mettre en évidence les différentes approches existantes de transfert de vidéo multi chemins. . On présente les éléments qui doivent être pris en compte en étudiant la transmission de vidéo d'un point de vue de vidéo (la méthode de compression, le type d'application de transfert de la video), et d'un point de vue de la transmission (le positionnement en fonction de l'architecture des réseaux, les différentes stratégies de transmission multichemins).

Mots clés : transfert de vidéo, Internet, Multichemins , redondance

1 Introduction

Compared to non-streaming media (text, images, and graphics) transferring , video streaming applications are very sensitive to end to end delay, delay variations, while they can tolerate some data losses. Video streaming over Internet is still a very difficult task, especially if one wants to keep video quality as perceived by end user high enough.

Different approaches aim at solving the problems of low speed bottlenecks of many communication paths, and video streaming with respect to performance and without any guarantee. One possibility was to reserve resources (like RSVP [21]), or to protect specific flows (like DiffServ [21]). Such approaches suffer from scalability, complexity and deployment problems. Most of the existing video streaming approaches like [20] [24] [26] use a logical or physical single path over Internet in order to stream the video from sender to destination. Recently, several research groups all around the world work on a very promising idea, which is to get the benefit of the multiple existing paths over the Internet between any two nodes, and this will be by streaming the video over several paths instead of a single path [17] [10] [7] [13] [9] [29] [16]. These studies show how the term multipaths can open a wide range of possibilities according to the number of senders, receivers, and paths to stream the video. The goal of multipath video streaming researches is to develop or design new approaches that can be deployed over Internet network, on the upper layers without requirements of modifying the lower layers of the existing Internet infrastructure.

A general architecture of the video streaming process over Internet is shown in Figure 1. According to the video streaming application in demand, the original video flow will be pre-stored over the server or generated in real time. Furthermore, before passing over the existing Internet infrastructure, the frames holding video streams will pass through one or several transmission mechanisms and protocols. These mechanisms and protocols have been developed to adapt with the real-time-aware characteristic of multimedia streaming applications. Consequently, video streaming transmission application gives a kind of priority to the time-constrained video frames over the non-streaming frames through the Internet. In the other hand, in order to play-back the arrived video streams on a client, special treatment is also demanded from the video streaming recipient application, like buffering some frames before starting the play-back, or putting the arrived frames in their correct order, or recovering some losses, . . .etc. Each of the existing video streaming approaches over Internet tries to concentrate on the different components of the general architecture illustrated in the Figure 1.

This report is organized as following. In the next section we highlight the most important elements to be considered when streaming video over Internet, regarding the video itself. Methods of representing the video (coding methods), and the different video streaming applications according to the start up delay accepted by each of them. The third section presents another main aspect to be considered when dealing with multimedia streaming which is the way of managing the video transmission in the Internet architecture. This can be done from one hand, regarding the positioning of different video streaming solutions

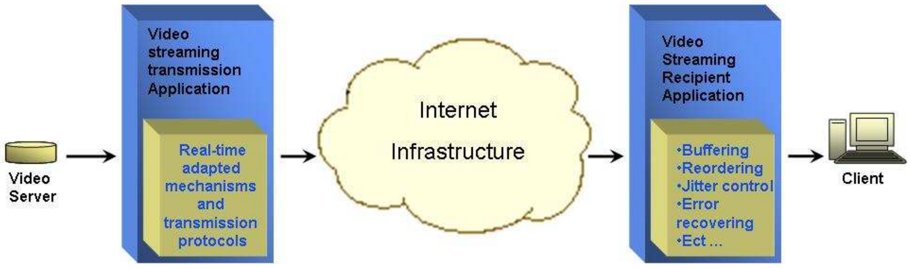

Figure 1: A general architecture of the video streaming process over Internet

over the different layers of the network architecture, and from the other hand, considering the number of used paths to transmit the video. The fourth section aims at classifying the different video streaming approaches into two main categories: single path and multiple path video streaming approaches. Even though most of the existing approaches use the single path method to stream the video over internet, this study clarifies the potential benefits of going forward the second category of video streaming, which is to stream video over multiple paths instead of the traditional single path method. Finally a conclusion is given in the last section.

2 Video in multimedia streaming

The term video-streaming over Internet holds two important aspects that most be considered, regarding the video itself, and the transmission over Internet network. In this section we present the elements that must be considered when streaming video over Internet, related to the video itself. These elements are: the method of representing the video and the different video streaming applications, according to the start up delay accepted by each of them.

2.1 Different video compression standards

Due to the huge sizes of video files, sending a raw video over Internet consumes non acceptable amount of bandwidth. Thus, beside saving disk space, video compression is also employed to achieve transmission efficiency. Video compression -like data compression- is a tradeoff between disk space, video quality and cost of hardware required to compress/decompress the video in a reasonable time. Thus, enhancing the compression rates and the quality of video has become a necessity. Two main standards committees are doing parallel development of video compression standards, the Moving Picture Experts Group (MPEG), and the International Telecommunication Union (ITU). The most important recent standards developed by these two committees are: MPEG-1, MPEG-2, MPEG-4, H.264/AVC and JPEG 2000[18]. Considering video streaming over Internet, some of these standards have been widely used. MPEG-2 is the current standard for High Definition Television transmission. It is widely used for video streaming over Internet, especially with its easy deployment and coding mechanism, but as it was mentioned in [21] MPEG-1 and MPEG-2 are still essentially linear and interactivity is limited to operations such as slow motion, frame-by frame or fast forward. Unlike its predecessors MPEG-1 and MPEG-2, the MPEG-4 standard does not focus only on the media compression aspect of multimedia technology. It also considers the media packaging and delivery parts and many of the tools that future multimedia applications might need. For that it can be considered as an object-based multimedia technology [1]. Furthermore, MPEG-4 has a very low coding bit-rate , and it offers high compression ratio and error resilience(more robust against errors) [14].

In the other hand, with its significant bandwidth saving, and with its flexibility to select from a number of reference frames for motion estimation for a given predicted frame [4], H.264/AVC (also called MPEG-4 part 10) is supposed to replace the use of MPEG-2 video compression. Especially that comparing to MPEG-2, H.264/AVC can produce a perceptually equivalent quality video at about half the bit rate [30]. However, there are huge differences between the MPEG-2 and the H.264/AVC coding algorithms. Thus, the transcoding between the two standards still hard to be implemented comparing to MPEG-2 to MPEG-4 transcoding.

2.2 Types of video streaming applications

In contrast with transferring, real time multimedia streaming has timing constraints. In the transferring mode, the user downloads the whole video file before playing it back. This may suffer long and unacceptable transfer time, especially, with the large size video files. However, streaming the video means that the user does not need to wait the whole media file to arrive before playing it back. Video streams will be played out while parts of the streams are being received and decoded.

Naturally, the transmission rate of a media stream is adjusted to its play-back rate, in order to keep the presentation running continuously [23]. But, this can not always be possible, because of the network delay. Therefore, video streaming applications need to buffer some frames and delay the first frame before releasing it to the player. According to [21] the time between the user's request for a video and the moment of starting the playing back of this video differs due to the type of video streaming application in demand. Here we mention the different possible types with the tolerated start-up delay for each of them.

Stored video streaming: In this class, client requests video files that are pre-recorded and stored by servers. Thus, users may pause, rewind, fast forward, or index through the video content. Here, a delay of 1 to 10 seconds between the request of the client and the execution of one of these actions is still accepted.

Live video streaming: This class of applications is similar to the traditional television broadcasting, except that transmission takes place over the Internet. Since the video to be transmitted here is not stored, the user cannot fast forward through the media. Delays up to tens of seconds from when the user requests a live streaming to when play-out begins can be tolerated.

Interactive Real time video streaming: This class of applications allows people to communicate between them in real time. Thus, according to [21] delays between 150 and 400 milliseconds are still acceptable.

Next in this report we will see how most of the existing studies lend to stream stored video. Even though the time delay constraints for the stored video streaming are less stringent than those for live and interactive video streams, it still has critical delay constraints on data delivery. Thus, the transmission of the video over best effort Internet network is still a critical issue as we will illustrate in the next section.

3 Solutions for multimedia transmission

The second main aspect to be considered when dealing with multimedia streaming is the way the transmission is managed in the Internet architecture. Indeed, to cope with the time delay constraint, solutions for multimedia streaming over the Internet architecture are provided either directly in the IP level or up to the applications layer as shown in Figure 2.

Here we present the different levels where solutions for multimedia streaming can be provided and the existing solutions for each level of the Internet architecture.

3.1 Positioning of video streaming applications

In the 'best effort' IP protocol, all packets are treated equally at the routers. As explained above, multimedia streaming are delay sensitive. For multimedia packets, it is then necessary to provide additive mechanisms that are needed to give a kind of priority to those packets. The first level where these mechanisms can be implemented is in the IP layer.

3.1.1 IP layer solutions for multimedia transmission

The solution for multimedia over IP is to classify all traffic, allocate priority for different applications and make reservations. Two main solutions are RSVP and DiffServ.

RSVP (Resource Reservation Protocol): [21], described in RFC 2205, is designed to reserve resources across a network for an integrated services Internet. RSVP does not transport application data but is rather an Internet control protocol. Resources will be reserved in each node along the transmission path. Some of the major disadvantages of the RSVP protocol are that, resources reservation over all the routers along the entire path of transmission makes RSVP scale poorly. Thus RSVP presents a considerable overhead in large networks like Internet. These considerations had led to the so-called 'DiffServ' architecture.

DiffServ (Differentiated Services): Diffserv is a computer networking architecture that specifies a simple, scalable mechanism for classifying, managing network traffic and providing Quality of Service (QoS) guarantees on modern IP networks [21]. The disadvantage of this system is the complexity; it needs a complete different architecture of the Internet.

3.1.2 Transport layer solutions for multimedia transmission

Multimedia applications developers in the most of recent studies [12] [4] [30] [29] [25] [5] often choose to run the applications over UDP instead of TCP as the transport protocol for several reasons: the non connection establishment, the non connection state of UDP, the small segment header overhead - UDP has only 8 bytes of header overhead - and the unregulated send rate of UDP- the speed at which UDP sends data is only constrained by the rate at which the application generates data. More over TCP cannot be employed with multicast.

Even though TCP provides a reliable data transfer service compared to UDP, TCP is conventionally regarded as inappropriate for multi-media streaming, since its back off and retransmission mechanisms may lead to long delays which violate the real time requirement of multimedia streaming. [27] [26] defying the conventional wisdom, study an approach that relies on TCP for video streaming. This is motivated by the wide use of TCP for streaming in practice and commercial streaming products (e.g., Real Media and Windows Media). Furthermore, recent measurement studies have shown that, for both stored-video and live streaming, a significant fraction of the traffic uses HTTP underlying with TCP.

3.1.3 Application layer solutions for multimedia transmission

As it was mentioned over Internet all packets are treated equally at the routers, including delay sensitive audio and video packets. For this reason in any multimedia application, the sender will append the video or audio chunks with header fields containing sequence number, timestamp, and packet deadline time. This packet will be sent to the transport layer. Then all Internet applications can make use of this header to give some priority to the video or audio packets over the non-streaming media packets. Now to make sure that all the applications can easily get the benefit of this information it is convenient to have a standardized packet structure that includes a packet sequence number, timestamp, and deadline time, in addition to other important data. Here we mention some solutions that cope with this issue.

RTP (Real-time Transport Protocol): RTP defines a standardized packet format for delivering audio and video over the Internet. It was developed by the IETF and first published in RFC 1889. RTP is implemented over an adaptation layer between the transport and the application layers Figure 2. The services provided by RTP include time reconstruction, loss detection, security and content identification [2]. RTP is primarily designed for multicast of real-time data, but it can be also used in unicast. RTP is typically run on top of UDP to

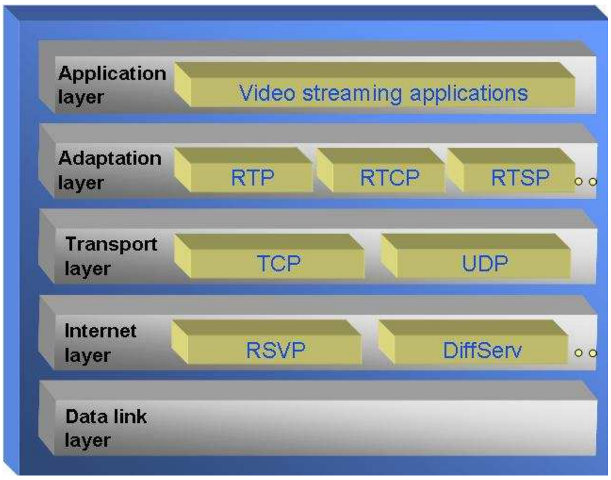

Figure 2: Video transmission: Solutions positioning over Internet architecture

make use of its multiplexing and checksum functions. It provides timestamping, sequence numbering, and other mechanisms to take care of the timing issues.

RTCP (Real-Time Control Protocol): RTCP is the control protocol designed to work in conjunction with RTP. It is standardized in RFC 1889 and 1890. In an RTP session, participants periodically send RTCP packets to convey feedback on quality of data delivery and information of membership.

RTSP (Real-Time Streaming Protocol): An application-level client-server multimedia presentation protocol to enable controlled delivery of streamed multimedia data over IP network. It provides 'VCR-style' remote control functionality for audio and video streams, like pause, fast forward, reverse, and absolute positioning.

Considering the benefits and the disadvantages of the above approaches, a promising idea will be to design an application-level approach that can be deployed over Internet network. The idea is to overcome the problems of scalability and deployments of the existing approaches, by pushing the complexity of QoS provision to the network edge.

3.2 Different schemes for video streaming

Internet is defined as a global data communications system that provides connectivity between its nodes. Thus, unlimited number of computers can act as video senders or receivers while they can be linked by different types and numbers of connections. This general architecture leads to another important element for classifying the different video streaming approaches, which is the number of used paths from sender to receiver. A new promising method will be to get the benefit of the diversity of the different existing paths between any sender and receiver, by streaming the video over multiple paths instead of a single path. The word multipath video streaming opens a wide range of possibilities differ between them according to the number of used paths, the number of senders, the number of receivers and the number of sent copies over each of the paths. Next in this report we will see how many approaches tend to stream the video from several senders to the receiver like in [17] [12] [29], instead of sending it from a single sender like in [24] [19]. Another important point is to couple the redundancy with video streaming. Thus, in order to keep the quality of the perceived video high enough, more than one copy of the same video flow can be sent over the different paths. Terminal thus may receive a redundant from the video file, or from some of its parts.

4 Path oriented classification

As we have mentioned in Section 3.2 another point to be taken in to consideration when analysing the different existing approaches of video streaming over Internet, is the number of paths used to deliver the video streams from the sender to the receiver. Next in this section we categorize the video streaming approaches into two grand categories: the most known and used category *the single path video streaming*, and the 'new' category *multiple paths video streaming*. For each of the existing studies, we will try to highlight its classification as it was mentioned in Section 2 from a video point of view (the type of video coding and of video streaming application), and from a transmission point of view Section 3 (the positioning in the network architecture and the scheme of video streaming).

4.1 Single path video streaming

Most of the existing video streaming approaches use the single path video streaming method, that is to stream the video from a sender to a destination over one path of the existing paths between these two nodes, through the Internet Network. Single path video streaming is a very vast domain that makes a first step toward the aim of our report, which is studying the multiple path video streaming. Here we try to highlight some of the most important studies in this domain from our point of view.

In [20], which can be considered as a survey, authors highlight six key areas of streaming stored video over Internet: video compression, application-layer QoS control, continuous media distribution services, streaming servers, media synchronization mechanisms, and protocols for streaming media. For each area, the study addresses the particular issues and review major approaches and mechanisms. It also discusses the tradeoffs of the approaches and point out future research directions.

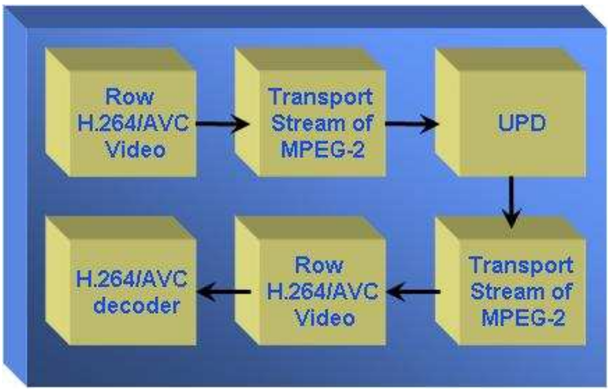

Figure 3: System architecture of H.264/AVC streaming over MPEG-2 system

Considering the positioning in the network architecture, we will start with the approaches that are deployed over the transport layer. One of these approaches is presented in [30]. The work is based on streaming real time video from a single sender to a single destination over IP and UDP transport protocol. The objective of this approach is to find a solution for transcoding H264/AVC video to MPEG2 video. H264/AVC is capable of providing good video quality at substantially lower bit rates than previous standards (e.g. half or less the bit rate of MPEG-2, H.263, or MPEG-4 Part 2), without increasing the complexity of design so much that it would be impractical or excessively expensive to implement. An additional goal was to provide enough flexibility to allow the standard to be applied to a wide variety of applications on a wide variety of networks and systems. In the other hand, the wide scale use of MPEG-2 in the market place today makes the complete migration to H264/AVC so difficult. Thus, the solution in [30] was driven by the important needs for the compatibility between H.264/AVC and MPEG-2 in the domains of DTV(Digital TeleVision), Mobile Video Communications and Network Video Streaming. According to the paper, as it shown in the Figure 3 the proposed system structure has fulfilled the expected functions so as to transport, in real time style, an H.264/AVC video stream over MPEG-4. But the results did not show the exact gain of this system over the other existing solutions.

Another transport layer solution for video streaming using TCP transport protocol to deliver CBR (Constant bit rate) video flows over a single path, is proposed in [11]. Despite the conventional wisdom that TCP is not desirable for video streaming and the large body of literature on UDP-based streaming, TCP is widely used in commercial streaming systems. Furthermore, a recent measurement study has shown that a significant fraction of commercial streaming traffic uses TCP [11]. This study found that 72% and 75% of the on-demand and live streaming traffic, respectively, used TCP. Motivated by this reason, the main objective in [26] was to find under what circumstances can TCP streaming provide satisfactory performance. To answer this question this study developed analytic performance models to systematically investigate the performance of TCP for both live and stored media streaming. It studies a baseline streaming scheme - similar to HTTP streaming - this scheme uses TCP directly for streaming. The results of this work show that direct TCP streaming generally provides good performance when the available network bandwidth, and thus the achievable TCP throughput, is roughly twice the video bit-rate, with only a few seconds of start-up delay. However, the performance of TCP streaming improves as the achievable TCP throughput is becoming higher and higher than the video playback rate. In the other hand, studies show that the video quality drops sharply when the packet loss ratio exceeds 10^{-4}. Therefore, for large RTTs, high loss rates and timeout values, to achieve a low fraction of late packets, we have even to tolerate a large start-up delay or to achieve a very high TCP throughput according to the video playback rate.

A lot of studies tried to cope with the problem of [11] of the long start-up delay. One of these studies is [23], which proposes a new mechanism to eliminate or reduce the initial start-up delay of Internet video streaming applications. This study uses UDP transport protocol to stream a stored MPEG4 video over a single path. While the transmission rate of a media stream is adjusted to its play-back rate, in another word, if the inter-frame time is T, the sender should transmit (at least) 1/T frames per second to keep the presentation running. However, the discrepancy in network delay does not allow for this ideal case. Thus, some frames are buffered before starting the play-back, and that may cause some larges start-up delays. The general idea of the work presented in [23] was a fast start streaming rate controller that transmits frames at a higher rate at the beginning of a presentation, but later slows down the transmission rate to the natural rate of play-back, 1/T frames per second. The results show that for low jitter environments, where the jitter buffer size does not have to be large, fast start does not generate much burst. The fast start traffic is, however, more demanding than the natural traffic pattern of a stream. Especially, the initial burst requirements of the mechanism are taken into account.

Another category of video streaming solutions is to use a real time adaptation protocol over the application layer Figure 2. One of the single path video streaming approach that uses RTP over TCP network layer protocol is presented in [19]. The study is an implementation of an end to end application for streaming stored MPEG4-FGS (Fine-Grained Scalable) video over Internet where the video is encoded into one base layer (BL) and one or several enhancement layers (EL). Over RTP the video flow is devised to BL and EL packets, but over the TCP a single packet contains the two RTP packets is sent from single sender to the receiver over a single path Figure 4. The experiments in this work show that the system gives good visual performance despite low efficiency of current FGS-encoding.

Staying over the application layer, and although we are considering the single path video streaming scheme, another promising is to allow content providers to choose between several network paths towards a given receiver. Such path diversity gives video streaming one

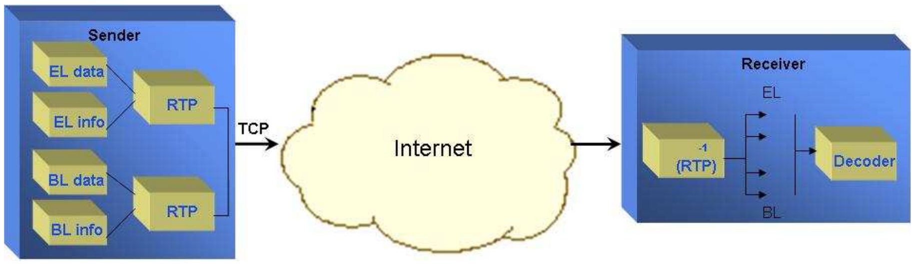

Figure 4: Implementation of Adaptive Streaming of Stored MPEG-4 FGS Video

more adaptation option: to dynamically switch from one path to another depending on the observed (or predicted) performance in the candidate paths. The studies in this area rely on loss rate, delay or TCP throughput measurements. [6] streams an MPEG-2 clip, from one sender to one receiver over a selected path. The study considers an overlay-based video streaming architecture in which the objective is to maximize the perceived video quality through dynamic overlay path selection. A novel aspect of this work is that the path selection process rely on available bandwidth (avail-bw) estimation. The avail-bw of a network represents the maximum additional load that the path can carry before it becomes saturated. The reason of focusing on avail-bw is because this metric can determine whether a path has enough capacity to carry a video stream before we switch the stream to that path. Other network-layer metrics, such as jitter or packet loss rate, can only determine whether a path is already congested, causing degradation in the video quality at the receiver. In order to prove the efficiency of avail-bw path selection, evaluations for four path selection schemes were done. These schemes are distinguished based on the choice of the key measured network performance metric.And they are: (1) Loss based path selection (LPS): The path with the minimum loss rate is selected. (2) Jitter based path selection (JPS): The path with the minimum 90th percentile of jitter measurements is selected. If the minimum jitter is practically the same in more than one paths, then JPS selects the path with the lowest loss rate. (3) Avail-bw based path selection (APS): This scheme has two variations, the average avail-bw (A-APS), and (4) the lower bound of the avail-bw variation range (L-APS). The path with the highest avail-bw estimate is selected. The comparing between the different schemes was done based on three criteria: video quality, user-abort probability and path switching frequency. The results show that the JPS and L-APS schemes have comparable performance and in some cases L-APS is slightly better than JPS, and they are clearly better than A-APS and LPS. This is because both JPS and L-APS are able to detect the onset of queuing delays in the currently selected path, before that path becomes congested. In the other hand, L-APS has the lowest path switching frequency. JPS causes significantly more path changes. Finally, even for the user-abort probability L-APS also improves the performance.

Since there is no way to guarantee a stable bandwidth over a best-effort network, streaming systems usually would just estimate the most appropriate amount of bandwidth according to its knowledge about the current network condition, and base that bandwidth estimation to stream the video data that are needed by the user for the playback. To this reason [24] proposes an efficient mechanism, namely piggyback prefetching, to improve the overall utilization of the precious network bandwidth for the single path streaming systems over wide-area networks. The tests had been done streaming stored MPEG-1 video files over a single path from the sender to the receiver. Assuming the server adopts certain bandwidth estimation mechanism (like the one discussed in [6]) to estimate the bandwidth, then it will roughly know how much bandwidth will be unoccupied when it allocates certain bandwidth to deliver a segment of video data. Piggyback prefetching mechanism wants to make use of such 'spare space' in the allocated bandwidth by having the server deliver some prefetched data using this unoccupied bandwidth, as it is shown in Figure 5. According to the prposed approach, and since the data delivered using the unoccupied bandwidth are just prefetched data (i.e. some data for future use), they are not critical to users' current playback, therefore, it will not cause any problem to users' playback if they are delayed or even lost. Furthermore, the prefetched data do not need to be stored in clients' buffers. They will be stored in proxy servers, where huge storage space can be utilized. The results show that the amount of data can be prefetched is substantial. For example, when the bandwidth allocated is 50% of the peak data rate, as much as 33.7 minutes of extra video data can be prefetched using the unoccupied bandwidth. However, these results still preliminary and need some extra use cases to prove them.

4.2 Multiple paths video streaming

4.2.1 Introduction

Single path video streaming over Internet still suffers from several limitations. For example, some approaches need resource reservation. This will limit their scalability and deployment over Internet. Other approaches are very complex and need a complete different architecture of the Internet. Furthermore, transferring over a single path will absolutely suffer from congestion problems. In order to overcome these limitations, and many other ones, a new solution has recently been proposed by several research groups. It consistes in streaming the video from the sender to the destination over multiple paths on the Internet network, instead of using a single path. As we have seen in Section 3.2 the term multipath holds many definitions differ between them according to the number of senders, number of receivers, number of used paths, in addition to the number of the sent copies of the video flow. The main objective of multipath video streaming is however to offer an optimal quality of service in delay-constrained video applications.

Certainly, using more than one path to stream video raises a vast range of questions that have to be considered. For example, how to determine if the paths are completely separated

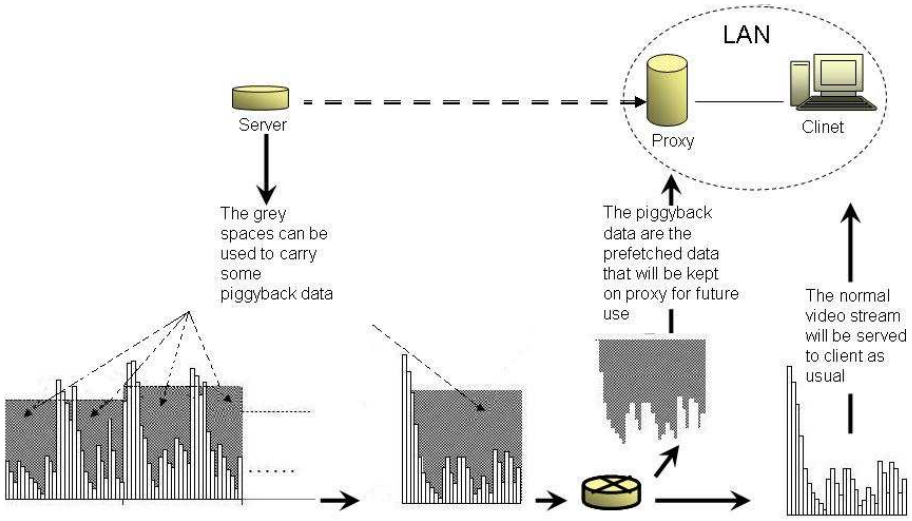

Figure 5: The piggyback prefetching mechanism

or if they have some joint points of congestions. In the other hand, how to adapt with network changing conditions, consequently, how to redistribute the loads over the multiple paths. Another important point is to decide whether or not the support from lower layer protocols is required. Then, how to get the benefit of the redundancy and error erasure scheme on the receiver side.

Next we will demonstrate the advantages of using multipaths in video streaming instead the traditional single path streaming. Then, we will give an analytical study about the different approaches that have employed this methodology, and how each of them handles the questions enumerated above.

4.2.2 Multipath video streaming advantages

Multipath video streaming allows increasing the streaming bandwidth and throughput. In another word, when the video has to be streamed over single path with a small bandwidth it must be more compressed which leads to a more bad quality. Using multiple paths will give an aggregate bandwidth leading to enhance the quality of transmitted video. In the other hand, it the continuous media traffic between the multiple available paths. Another potential benefit of multipath video streaming as it has been mentioned in [17] [15] [16] is to reduce burst lengths of packets, which means reduce correlations between consecutive losses. Additionally, using multiple paths will increase the robustness and the ability of adaptation with the variations of congestions patterns. Video streaming is one of the long lasting applications over the changeable not robust Internet network. Thus, sending data over multiple paths will increase the ability to adapt with the changes of network conditions. Finally, combined with error resilient streaming strategies, multipath video streaming provides means to limit packet loss effects.

4.2.3 Multipath video streaming approaches

Recently, multipath video streaming has attracted much attention. Here we give a state of the art about the different existing approaches exploit multipath streaming. Like we did in Section 4.1, the approaches will be listed regarding the positioning in the network architecture. In the other hand, each of the existing studies will be categorized according to Section 2 and Section 3.

As we have already seen in Section 4.1, one of the earlier proposed solution to transmit video over TCP as a transport layer protocol, was presented in [26]. The approach has studied the benefits of single path video streaming over TCP motivated by the widely support of TCP streaming in commercial products, and with some recent measurement studies that show that for both stored-video and live streaming, a significant fraction of the traffic (around or above 50%) uses HTTP/TCP. The experiments had shown that single-path TCP streaming performance is generally satisfactory when the achievable TCP throughput is roughly twice the media bit rate, with a few seconds of start up delay.

As a continuing of [26], and by applying the multipath video streaming aspect, [27] comes to answer two important questions: Under what circumstances can multipath TCP-based live streaming provide satisfactory performance? And what are the benefits from using multiple paths, compared to using a single path, in TCP-based live streaming? The scenario considered to answer these questions is the following: a single video server generates a CBR video content in real time and streams it via TCP to a single client over several paths which may or may not share bottleneck links. Authers in [27] propose a simple and practical TCP-based multipath streaming scheme that is called Dynamic MPath streaming (DMP-streaming). DMP-streaming dynamically distributes packets over multiple paths by implicitly inferring the available bandwidths on these paths. The study shows that using two paths, each with half of the achievable TCP throughput of a single path, can support the same (even higher) video bit rate supported by the single path. In the other hand, two paths, each with the achievable TCP throughput of the single path, can support videos with twice (even more than twice) the bit rate supported by the single path. Hence, in addition to economical reasons (subscribing to multiple low-bandwidth access links is cheaper than subscribing to a single high-bandwidth access link), it is also advantageous to use multipath for TCP-based streaming due to performance reasons. Thus, as an answer about the two main questions presented as an objective of this work, [26] says: The performance is generally satisfactory when the aggregate achievable TCP throughput is 1:6 times the video bit rate, with a few seconds of start up delay. Moreover, the performance of DMP-streaming is not sensitive to path heterogeneity, so it can be used when the multiple TCP flows share or do not share bottleneck links. Again this study does not handle the long start up delay problem. Furthermore, it does not indicate the loose when using throughputs less than 1:6 times the video bit rate. In the other hand, TCP will be always regarded as inappropriate for multimedia streaming, since its back off and retransmission mechanisms may lead to long delays which violate the real time requirement of multimedia streaming.

An improved transport layer solution was proposed by [28] and [29]. In the objective achieving higher throughput, increasing tolerance to packet loss and delay due to network congestion, the work uses TCP-friendly instead of TCP to stream H.263 video flows. TCP-friendly is designed to be fair with TCP traffic, results in less fluctuation in sending rate than TCP does. The proposed approach designs a receiver-driven protocol for simultaneous video streaming from multiple mirror senders to a single receiver. This protocol employs a novel rate allocation algorithm (RAA) that runs on the receiver to specify the sending rate for each sender in order to minimize the total loss rate. Also it employs a packet partition algorithm (PPA) that runs on each sender to partition packets. Thus, to insure that every packet is sent by one and only one sender and to minimize the start up delay.

As it is illustrated in Figure 6 the protocol presented by [28] functions as following: Each sender estimates and sends its round trip time to the receiver. Then the receiver uses the estimated round trip times and its estimates of sender's loss rates to calculate the optimal sending rate for each sender. When the receiver decides to change any of the sender's sending rates, it sends an identical control packet to each sender. Finally, using the specified sending rates and synchronization sequence number, each sender runs a distributed packet partition algorithm to determine the next packet to be sent. The experiments have demonstrated the effectiveness of distributed video streaming framework in reducing overall packet loss rate. But in the other hand, the study assumes that the multiple paths do not share congestion links, which is not always possible. Additionally, it considers the possibility of changing the sending rates among a fixed set of senders and not in a dynamic way.

One reason why multipath streaming has not been widely explored in the Internet is that multipath streaming is hard to provide at the low layers. However, the recent trend in networking to consider routing or other networking activities at the application layer [17] [16] [8] opens new opportunities. Especially that an application layer multipath streaming mechanism is far easier to implement and to deploy than one within IP layer. However, the trend towards application-layer implementation, for the communication protocols facilitates the deployment of a multipath streaming in the current Internet. A lot of the existing studies have presented this promising trend. One of these studies is [9], in which two approaches between transport and application layers where proposed. The main objective of using these two approaches is to combine adaptation and multipath streaming. in the aim of solving the problem of multipath-streaming unknown resources. Moreover, to cope with the dynamic changes of these resources over time. To this reason continuous adaptation is needed even for a multipath streaming setup. Both approaches have been implemented in the context of an adaptive MPEG application. The first approach in this study implements the

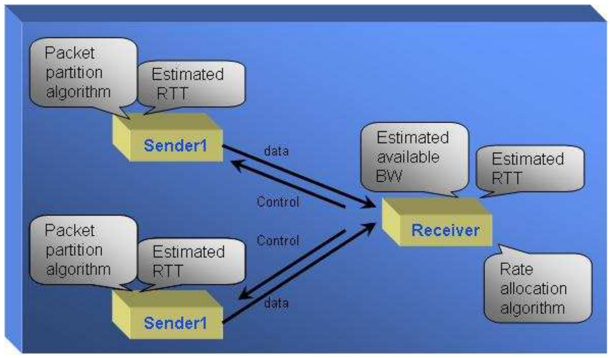

Figure 6: High level description of Distributed video streaming framework

multipath streaming completely transparent to the application. This approach follows the layering approaches of the traditional Internet design and current overlay networks by separating applications from the data transport. it iscalled transport-layer multipath streaming (TLMS), because it hides the splitting behind the transport-layer socket API.

In contrast, the second approach focuses on an integration of the data transport into the application context and allows a combination of adaptation and multipath streaming. This approach is called application-layer multipath streaming (ALMS). ALMS setup is similar to a multicast setup with individual stream adaptation, with two exceptions. First, all sub streams reach the same destination. Second, multipath streaming and filtering must be combined to ensure that (i) not the same data is sent in the different sub streams and (ii) the most important frames are transmitted first (over any path). TLMS, which is closely related to the network, works with UDP packets. In contrast, ALMS works with application-specific data types (MPEG frames). From an engineering point of view, TLMS has the advantage that no changes are needed in the application code. The splitting can be hidden from the application, which makes TLMS also portable to different applications. TLMS also has a low overhead and a low impact on the system performance. However, TLMS has significant disadvantages. Every path in a best-effort network has its own dynamic behaviour, e.g., different latencies or error rates. Any asymmetry in this behaviour must be addressed by an application, e.g., to maintain synchronization. TLMS only sees the effects of the asymmetry on the whole stream, but it is neither able to identify the misbehaving path nor can it take appropriate reactions. In contrast, ALMS integrates the multipath streaming and the adaptation into the application context. Because it first splits the data and adapts every sub stream individually, it is able to deal with path asymmetries. Although TLMS has a lower overhead and is more efficient, ALMS is able to deliver a better video quality because the adaptation and the splitting mechanisms use application-layer metrics. In the other hand, the mapping of networking metrics (bandwidth) onto application-layer metrics

(MPEG frames) is not easily performed.

Over the application layer, a new approach called path diversity has been addressed in several recent studies in order to achieve better end-to-end loss behaviour. The first study we will mention is [5], this study stays in the single sender single receiver scheme and addresses some main issues that arise with path diversity, and the results of the study show:

- It is advantageous to increase the number of paths up to a certain (application-specific) threshold value, but not beyond in order to decrease the probability of a long burst. If we increase the number of paths, the probability of a long burst decreases, but the probability of a small burst may increase. Since the probability of a very long burst is generally very small, it may not be efficient to perform path diversity over many paths.
- Path diversity is still beneficial when paths share a common bottleneck. The gains obtained depend heavily on the loss processes of the shared segment and those for the independent segments in the paths. The effect of a shared bottleneck link in the overall performance of the method may vary drastically, and the method may be either worse or better than interleaving, but it is still superior when compared to the single path case.
- Increasing the number of applications that employ the technique introduces correlation among the loss processes. On the other hand, distributing the load over many paths smooth the overall traffic, this causes a reduction on the packet loss probability. Experiments show that the best scenario for an application is that when only a few of them use path diversity.

[25] also deploys path diversity concept, over the application layer. In contrast, it uses another scheme which is to send data from different senders to a single receiver. This work proposes a system that improves the performance of streaming media by exploiting the path diversity provided by existing content delivery networks (CDN) infrastructure coupling with multiple description (MD) coding of MPEG-4 and H.263 to provide resilience to losses.

CDNs have been widely used to provide low latency, scalability, fault tolerance, and load balancing for the delivery of web content and more recently streaming media. They improve end-user performance by caching popular content on edge servers located closer to users, thus reducing the request response time, the probability of packet loss, and the total network resource usage. CDN provides path diversity by the different network paths that exist between a client and its nearby edge servers. Static CDN solutions to cope with packet loss like retransmissions are not quite possible for streaming context. Thus new mechanisms are required. Multiple description coding (MDC) with path diversity is one of the proposed mechanism to cope streaming real time criteria.

With MDC each description can be decoded independently to give a usable reproduction of the original signal. Thus, the multiple descriptions contain complementary information so that a useful reproduction of the signal is allowed when any description is received; the quality of the decoded signal improves with the number of descriptions that are correctly received. Designing the MD-CDN architecture some issues have to be studied: How to distribute the MD streams across the existing nodes of CDNs and how to select for each client multiple neighbour nodes with complementary descriptions. The results of this work show that distortion reduction by about 20% to 40% can be realized even when the underlying CDN is not designed with MDC streaming in mind. Also, for certain topologies, MDC requires about 50% fewer CDN servers than conventional streaming techniques. to achieve the same distortion at the clients.

Another path diversity approach is presented in [4], in the same scheme as [25] from several senders to a single receiver, but this time not coupled with multiple description video coder, instead, H.264 is used. H.264 is a new standard of video coding that offers many coding flexibilities for better coding and streaming performance. One of these flexibilities is flexible motion-estimation support where each P-frame can choose among a number of frames for motion-estimation. This study proposes an optimization algorithm using dynamic programming that exploits this flexibility for multipath simultaneously streaming with real-time playback over two transmission paths with different bandwidths and loss rates. Using multiple paths simultaneously means larger combined transmission rate in the case when each path is rate constrained. The question to be answered by this approach is: what is the jointly optimal selection of reference frame and transmission path for optimal performance. To this objective and in contrast with other existing work, [4] optimizes the selection of both the reference frame and the transmission path simultaneously, and not consecutively. In the other hand, it has an advantage over the MDC approach presented in [25] in the assumption that simultaneous failure in both paths is probable. MDC always assumes that transmission errors typically occur in one of the two transmission paths and not in both of them at the same time. Thus, the proposed algorithm can simply consider the MDC as a special case of many other possibilities.

Not away from path diversity, another approach of multipath video streaming over the application layer, using the multiple sender single receiver scheme, was presented in [17] [16] [8] and [12]. The objective of this approach is to explore high quality streaming, which means significant bandwidth requirements, of relatively long duration, and without information loss or hiccups in data delivery. To this purpose an pre-stored MPEG1 video flow was fragmented into packets, then different packets take alternate routes to a single receiver. This is illustrated in Figure 7 where three servers are used to stream data over a wide area network. Any server can send any part of data; specifically, server i sends fraction α_i of the data expected by the receiver, where $0 \leq \alpha_i \leq 1$ and $\sum_i \alpha_i = 1$. This can be achieved by determining a sending pattern for each server, as in Figure 7 where each sender only sends packets depicted by the solid rectangles. As each packet is sent by only one of the senders, the total amount of data sent is the same as in a single path case. Thus, the overall load on the network will not increased using this approach.

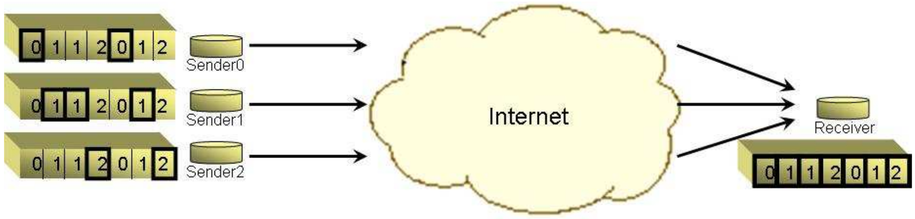

Figure 7: Different packets from multiple sender to the receiver

An advantage of this approach (as compared to approaches that require support of lower layers) is that the complexity of QoS provision can be pushed to the network edge hence improve the scalability and deployment characteristics while at the same time provide a certain level of QoS guarantees. The results as mentioned in [12] indicate that in general, multipath streaming exhibits better loss characteristics than single path streaming with or without the use of an erasure code. In the other hand, [16] adds a study about the load distribution problem. It focuses on determining an appropriate optimization objective for computing the load distribution. Then it conducts a performance study to understand the goodness of these optimization objectives. In contrast, [8] lists some limitations of the approach, starting with the importance of considering the potential costs or detrimental effects of multi-path streaming. Then, the overheads associated with sending data over multiple paths and then assembling it into a single stream at the receiver should also be considered. Finally, the approach assumes that the multiple paths have always disjoint bottleneck nodes, which is not always reliable over real Internet.

Finally, staying over the application layer, one new scheme of path oriented classification as it is mentioned in Section 3.2, is to stream video from a single sender to multiple clients. This can be simply done by giving the different addresses of the clients to the sender. The sender in his turn will stream several copies of the same video over these addresses. Another possibility is to use multicast. Multicast video streaming solution came to cope with the increasing demand of interactive and teleconferencing video applications over Internet. In the other hand, supporting multicast communications of multimedia applications introduces important challenges unaddressed in traditional network architectures: first, multicast groups consist of heterogeneous receivers, and second, multimedia data streams are often multi-resolution.

Traditional solutions have proposed a layered multicast approach for multicasting prioritized data streams. But the main limitation of such an approach is that, it decomposes the multi resolution data stream into component single resolution streams, establishing a distinct multicast group for each component stream using standard IP multicast. Then, it leaves each receiver decides which multicast groups it wants to join based on the perceived connection quality or on the coordinated control mechanism. This approach has several

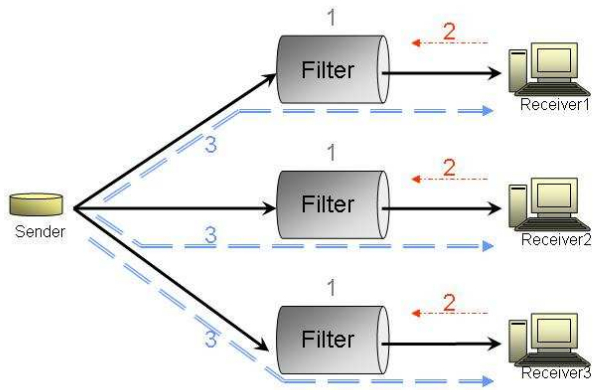

Figure 8: The multicast tree of MHPF: protocol steps

inherent limitations imposed by slow reaction to network dynamics, destructive interference by concurrent adaptation by multiple receivers, and overhead of decomposition and resynchronization of the multimedia stream at the end hosts.

To overcome these limitations, [3] presents an overlay network architecture called MHPF (Multicast Heterogeneous Packet Flows) that supports multicast communications of heterogeneous data for small to medium multicast groups without decomposing them into component homogeneous data streams. The simulations were done using CBR live video flows from one sender to multiple receivers, in order to prove the efficiency of this application-level multicast architecture. MHPF used an adaptive transport protocol called HPF (Heterogeneous Packet Flows) which was presented by a previous work of the same laboratory in [22]. MHPF extends the idea of HPF to the multicast domain.

Figure 8 illustrates how MHPF works. For each session MHPF abstracts a multicast tree T composed only of MHPF servers and multicast tunnels between them, which constitutes the overlay network on top of the IP multicast infrastructure. The HPF protocol at the sender interleaves packets with different priorities and transmits them in a single heterogeneous data stream. The MHPF servers implement a specialized packet forwarding behavior (rate adaptation and priority-based filtering on each multicast tunnel) so that only the highest priority packets that can be accommodated on a path downstream are transmitted along the path. In the other hand, each receiver periodically generates feedback that contains the information on the bandwidth on the path leading to itself. The feedback from the receivers travels upstream along the multicast tree T and gets aggregated at the MHPF servers.Finally, the network of MHPF servers performs rate adaptation and packet filtering so that only the highest priority packets are forwarded downstreams. Two important results can be taken from the tests: when network condition is dynamically changing, MHPF effectively adapts to the condition, and as the number of receivers increases the performance of MHPF degrades gracefully.

5 Conclusion

In this report we presented a state of the art about multipath video streaming over Internet. We illustrated how video streaming application over Internet face an important challenge, especially if one wants to keep the perceived quality high enough.

We presented the most important elements that must be considered when streaming video over Internet, related to the video itself and to the transmission mechanism. Beside video compression standards, and video streaming application types, the second main aspect to be considered when dealing with multimedia streaming is the way the transmission is managed in the Internet architecture. We showed some of the different solutions for multimedia streaming over different levels in the network architecture (IP layer, transport layer, and application layer solutions). Then, we illustrated the different schemes of video streaming according to the number of used paths to deliver video streams from senders to receivers. From this last point, we categorized video streaming approaches into two grand categories: single path and multipath video streaming.

Important single-path video streaming approaches have been analyzed. In the other hand, more details about the existing approaches using multipath video streaming, have been also studied and illustrated. The noticed results of using multi-path video streaming are quite encouraging and warrant further promising studies. In the aim at developing or designing new scalable multipath video streaming approaches that can be deployed over Internet network.

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6 Annex

6.1 Different video compression standards in more details

6.1.1 MPEG-1

- The first lossy compression scheme developed by the MPEG committee
- Used for CD-ROM video compression
- Use Discrete Cosine Transform (DCT) algorithm as a first step converting the image into the frequency domain.
- The current wildly popular MP3 (MPEG-1, Part 3) audio standard is actually the audio compression portion of the MPEG-1 standard.

6.1.2 MPEG-2

- Evolved to meet the needs of compressing higher-quality video.
- Used in today's video DVDs and digital broadcasts via satellite and cable.
- Use bit rates ranging from 5 to 8 Mbits/s.
- Use DCT transforms, but it also provides support for interlaced video (the format used by broadcast TV systems).
- With some enhancements, MPEG-2 is the current standard for High Definition Television transmission.
- Like MPEG-1 compression, MPEG-2 audio and video compression are still essentially linear and interactivity is limited to operations such as slow motion, frame-by frame or fast forward.

6.1.3 MPEG-4

- MPEG-4 has emerged as much more than a video and audio compression and decompression standard.
- A single standard covering the entire digital media workflow from capture, authoring and editing to encoding, distribution, playback and archiving.
- The MPEG-4 file format, based on Apple Computer's QuickTime technology, was developed by the MPEG committee as a standard designed to deliver interactive multimedia and graphics applications over networks and to guarantee seamless delivery of high-quality audio and video over IP-based networks and the Internet.

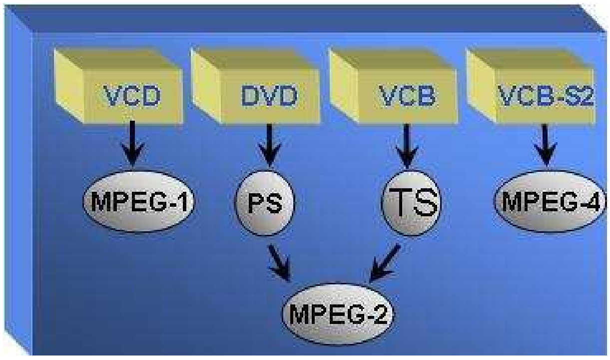

Figure 9: Different uses of MPEG standards

- The goal of MPEG-4 standard was trying to solve two video transport problems: " sending video over low-bandwidth channels such as the Internet and video cell phones, " And achieving better compression than MPEG-2 for broadcast signals.
- Use bit rate ranging from 64 Kbits/s to 1,800 Mbits/s.
- MPEG-4 is in the range of 15% better at compressing video data than MPEG-2, this has not been enough of an advantage to convert the whole broadcast industry to MPEG-4.

6.1.4 H.264/AVC

- H.264 is able to achieve a 2:1 improvement over MPEG-2 on full-quality SDTV and HDTV, and it is expected to come into wide use in satellite and cable TV over the next decade.
- H.264/MPEG4-AVC is a jointly developed standard by the ITU-T Video Coding Experts Group (VCEG) and the ISO/IEC Moving Picture Experts Group (MPEG) and has been standardized by the ITU under the H.264 name. It is also called MPEG-4 Part 10 AVC (Advanced Video Compression), even though it is unrelated in operation to MPEG-4.
- The main goals are to provide significantly enhanced compression performance and provision of a 'network-friendly' packet-based video representation addressing 'conversational' (video telephony) and 'non-conversational' (storage, broadcast or streaming) applications.
- Use a bit rate ranging from 40 Kbits/s to upwards of 10 Mbits/s

6.1.5 JPEG 2000

- JPEG(Joint photographic Expert Group) is distinct from MPEG
- JPEG was designed in the first place for still picture use.
- Specify the codec, which defines how an image is compressed into a stream of bytes and decompressed back into an image, and the file format used to contain that stream.
- The file format is known as 'JPEG Interchange Format'.
- JPEG/JFIF is the format most used for storing and transmitting photographs on the World Wide Web.
- JPEG 2000 is mentioned here because the Part 3 of the JPEG 2000 standard -Motion JPEG 2000- provides for motion video.
- Use 'wavelet' compression technology rather the DCT technology used in the MPEG and JPEG standards.
- The advantages of using Motion JPEG 2000 for video are:
  - Low latency compared to MPEG streams.
  - For DVR applications, every image is self-contained and complete; there is no need to reconstitute frames.
- The disadvantages are:
  - Lower compression ratios than MPEG algorithms.
  - Requires more computing power for decoding.