Scalable Internet Video-on-Demand Systems

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Kurzfassung

Im Rahmen dieser Dissertation zum Thema *"Scalable Internet Video-on-Demand Systems"* werden Lösungen entwickelt, die skalierbares Streaming im heutigen Internet ermöglichen. Die vorgestellten Lösungen ermöglichen sowohl die Optimierung existierender Video-on-Demand Anwendungen als auch die Integration neuer Mechanismen im Bereich Video Streaming.

Begründet durch die zunehmende Popularität von Video- und Audiodaten in packetvermittelten Netzwerken und der gleichzeitig mangelnden Unterstützung von Dienstgüte, müssen Mechanismen, die die Skalierbarkeit von Video-on-Demand Anwendungen erhöhen, zur Verfügung gestellt werden. Die in dieser Dissertation vorgestellte Skalierbarkeit für Video-on-Demand Anwendungen ist zweidimensional. Die erste Dimension wird als System-Skalierbarkeit bezeichnet und durch die Integration von Caches in die Verteilinfrastruktur realisiert. Inhalts-Skalierbarkeit, die zweite Dimension der Skalierbarkeit, ermöglicht, dass sich ein Videostrom in seiner Bandbreite an die existierenden Bedingungen im Netzwerk anpassen kann. Dies wird durch die Anwendung skalierbarer Kodierformate für Video erreicht.

Im Verlauf der Dissertation wird demonstriert, wie diese beiden Dimensionen von Skalierbarkeit kombiniert werden können, um die Gesamt-Skalierbarkeit und die Güte von Video-on-Demand Anwendungen zu erhöhen. Als erster Schritt werden die Probleme, die durch die Kombination beider Dimensionen von Skalierbarkeit entstehen, identifiziert. Diese Problemfindung führt zur Entwicklung einer neuen Architektur, die geeignet ist, die identifizierten Probleme zu lösen. Da diese neue Architektur auf mehreren neuen Mechanismen basiert, ist der größte Teil der Arbeit der Entwicklung und Untersuchung dieser Mechanismen gewidmet. In der initialen Phase der Arbeit stellte sich bei der Entwicklung erster neuer Mechanismen ein weiteres Problem heraus. Dies ist der Einfluß von Qualitätsschwankungen in einem skalierbarem Video auf die wahrgenommene Qualität beim Betrachter, welche bisher nicht im Rahmen einer subjektiven Beurteilung untersucht wurden. Daher wurde im Rahmen dieser Arbeit eine solche Untersuchung durchgeführt. Die Ergebnisse dieser Beurteilung sind die ersten, die substantielle Informationen für das Layer-Encoded Video Format in Zusammenhang mit Qualitätsschwankungen bieten. Diese Ergebnisse bilden die Grundlage für die Entwicklung eines neuen, objektiven Qualitätsmaßes, welches zur Beurteilung des Einflusses von Qualitätsveränderungen in skalierbaren Videoformaten benutzt werden kann. Es wird gezeigt, dass dieses Qualitätsmaß besser zur Beurteilung von Qualitätsveränderungen geeignet ist als eine prominentes, in der Bildverarbeitung etabliertes Maß. Darüber hinaus wird das neue Qualitätsmaß zur Evaluierung der in dieser Arbeit neu entwickelten Mechanismen benutzt.

Ein wichtiger Bestandteil einer Verteilinfrastruktur für Video-on-Demand sind Caches. Aus diesem Grund wird hier untersucht, wie neue Mechanismen (die die schon existierende Funktionalität eines Caches nutzen) die Skalierbarkeit der Verteilinfrastruktur und die Qualität des zum Empfänger gesendeten Videostroms erhöhen können. Mit Hilfe von Simulationen wird gezeigt, dass diese neuen Mechanismen geeignet sind, die identifizierten Probleme im Bezug auf Skalierbarkeit und Qualität zu lösen. Die Untersuchungen an neuen Mechanismen für *Skalierbare Internet Video-on-Demand Systeme* werden durch die Erweiterung existierender Transport- und Signalisierungsprotokolle ergänzt. Dadurch wird eine erweiterte Kommunikation zwischen Server, Cache und Endgerät in einer Verteilinfrastruktur ermöglicht, die nötig ist, um die neu entwikkelten Mechanismen optimal zu unterstützen. Simulationen und Messungen, die auf einer neu entwickelten Streaming-Plattform basieren, zeigen, dass die Kombination von neuen Mechanismen und die Erweiterung existierender Protokolle die Skalierbarkeit von Video-on-Demand Systemen und die am Empfänger empfundene Qualität eines Videostroms erhöhen. Zusätzlich wird gezeigt, dass diese neue Architektur (Scalable Adaptive Streaming Architecture) zur Unterstützung heterogener Endgeräte geeignet ist. Video-on-Demand Anwendungen, die auf den in dieser Arbeit entwickelten Mechanismen basieren, sind somit nicht auf eine Klasse von Endgeräten beschränkt und zeichnen sich durch eine noch höhere Gesamt-Skalierbarkeit des Systems aus.

Abstract

In this thesis on *Scalable Internet Video-on-Demand Systems* solutions are provided that allow scalable streaming in today's Internet. These solutions allow the optimization of existing Video-on-Demand applications, as well as the integration of new mechanisms in the video streaming area.

With the increasing popularity of video and audio data in packet switched networks and the lack of quality of service support, mechanisms that increase the scalability of Video-on-Demand applications must be provided. The scalability introduced in this thesis is two-dimensional. The first dimension is characterized as system scalability which is realized by the integration of caches into the distribution infrastructure. The second dimension, the content scalability, allows the adaptation of the video stream to existing conditions on the network by applying scalable video.

In the course of this thesis, it is demonstrated how these two dimensions of scalability can be combined to increase the overall scalability and performance of a Video-on-Demand application. First of all, the problems that arise from the combination of both scalability dimensions are identified. This problem identification results in a new architecture that is suitable to solve the problems identified. Since this new architecture is based on several new mechanisms, most of this thesis is dedicated to the development and investigation of these new mechanisms. In the initial phase, when first developing these mechanisms, an additional problem was identified. That is, the influence of quality variations in scalable video on the perceived quality of the viewer had not been investigated so far. The results of a subjective assessment, which was conducted in the scope of this thesis, are the first that provide substantiated information for the layer-encoded scalable video format. These results build the basis for the development of an objective quality metric, which is used to measure the influence of quality variation in scalable video. It is shown that this metric is better suited than an existing, well-known metric. In addition, this new metric is used to evaluate the newly developed mechanisms that are presented in this thesis.

An important element of a distribution infrastructure for Video-on-Demand is the cache and, thus, it is investigated how new mechanisms (which make use of the functionality offered by the cache) can increase the scalability of the system and the quality of the delivered video stream. Simulations which are based on these new mechanisms demonstrate their applicability and show that the identified problems can be solved. These investigations on new mechanisms for *Scalable Internet Video-on-Demand Systems* are enhanced by the extension of transport and signaling protocols. The latter allows the extended communication between servers, caches, and clients in a distribution infrastructure that is necessary to optimally support the new mechanisms. Simulations and measurements based on our own streaming platform show that the combination of both

increases the scalability of a Video-on-Demand system and the perceived quality for the viewer. Additionally, these results show that the Scalable Adaptive Streaming architecture is capable of supporting heterogeneous clients. Thus, Video-on-Demand applications making use of the new mechanisms presented in this thesis are not limited to a specific class of clients and are even capable of increasing the overall scalability of the system.

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"The journey is the reward" -Taoist Saying

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Abbreviations

ADSL	Asymmetric Digital Subscriber Line
AIMD	Additive Increase/Multiplicative Decrease
A/V	Audio/Video
AVT	Audio/Video Transport
BSD	Berkeley Software Distribution
CBR	Constant Bit Rate
CC	Cache Centric
CDI	Content Distribution Interworking
CFVC	Cache-friendly Viewer Centric
CDN	Content Distribution Network
CPU	Central Processing Unit
DAVIC	Digital Audio Visual Council
DCCP	Datagram Congestion Control Protocol
DCT	Discrete Cosine Transformation
DRM	Digital Rights Management
DSCQE	Double Stimulus Continuous Quality Evaluation
DSBV	Double Stimulus Binary Vote
DSIS	Double Stimulus Impairment Scale
DSL	Digital Subscriber Line
DSM-CC	Digital Storage Media Command and Control
DSS	Dynamic Stream Switching
DVB	Digital Video Broadcast
EZW	Embedded Zerotree Wavelet
ERA	Expanding Ring Advertisement

FEC	Forward Error Correction
FGS	Fine Granularity Scalability
FSC	Fair Share Claiming
FTP	File Transfer Protocol
GM	Graph Manager
HTML	Hypertext Markup Language
HTTP	Hypertext Transfer Protocol
IETF	Internet Engineering Task Force
ICP	Internet Cache Protocol
IP	Internet Protocol
ITU	International Telecommunications Union
KDE	K Desktop Environment
LAN	Local Area Network
LC-RTP	Loss Collection RTP
LD	Layer Dummy
LRMP	Light-weight Reliable Multicast Protocol
MAN	Metropolitan Area Network
MCL	Maximum of Cached Layers
MDC	Multiple Description Coding
MMUSIC	Multiparty Multimedia Session Control
MOL	Maximum of Original Layers
MP3	MPEG-1 Audio Layer III
MPEG	Motion Picture Expert Group
MTU	Maximum Transferable Unit
NVoD	Near Video-on-Demand
NoVoD	No Video-on-Demand
OS	Operating System
PDA	Personal Digital Assistant
POSIX	Portable Operating System Interface
PPV	Pay-per-view
PSNR	Peak Signal-to-Noise Ratio
PSTN	Public Switched Telephony Network
QoS	Quality of Service
QVoD	Quasi Video-on-Demand

RAP	Rate Adaptation Protocol
RAM	Random Access Memory
RDTSC	Read Time Stamp Counter
RFC	Request For Comment
RLM	Receiver-driven Layered Multicast
RMTP	Reliable Multicast Transport Protocol
RSVP	Resource Reservation Protocol
RTP	Real-time Transport Protocol
RTCP	Real-time Transport Control Protocol
RTSP	Real-time Streaming Protocol
RTT	Round Trip Time
SAS	Scalable Adaptive Streaming
SC	Stimulus Comparison
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SH	Stream Handler
SNR	Signal-to-Noise Ratio
SPEG	Scalable MPEG
SRM	Scalable Reliable Multicast
SR-RTP	Scalable Reliable RTP
SSCQE	Single Stimulus Quality Evaluation
TCP	Transmission Control Protocol
TEAR	TCP Emulation at Receivers
TFMCC	TCP-Friendly Multicast Congestion Control
TFRC	TCP-friendly Rate Control
TRM	Transport Protocol for Reliable Multicast
TTL	Time-to-live
TVoD	True Video-on-Demand
U-LLF	Unrestricted Lowest Layer First
UMTS	Universal Mobile Telecommunication System
U-LL-SGF	Unrestricted Lowest Layer Shortest Gap First
U-SG-LLF	Unrestricted Shortest Gap Lowest Layer First
UDP	User Datagram Protocol
VBR	Variable Bit Rate

VC	Viewer Centric
VoD	Video-on-Demand
WLAN	Wireless LAN
W-LLF	Window-based Lowest Layer First
WWW	World-Wide Web
XDSL	Protocols under the DSL umbrella (e.g., ADSL)

Chapter 1: Introduction

Caused by the increasing popularity of the Internet and especially one of its major applications, the WWW, not only in research and professional environments but also in broad public, the provision of new types of content is increasing rapidly. In the beginning of the WWW, the available content was mainly based on text (hyperlink) documents and still images. With the increasing popularity for private users content which is usually provided by the entertainment industry becomes more and more interesting. Such content is usually characterized as multimedia data [1] like audio, video, and the combination of both. With the need for such types of content new services and applications are offered in the Internet. An example of new services is radio station programmes which are offered, in addition to the traditional terrestrial broadcast, via the Internet.

Nevertheless, it is obvious that these services are only popular in the cases of no alternatives to obtain this content or where the offered quality is comparable to existing alternatives. For example, listening to a radio station via the Internet can be compared to listening to the same station on a car radio. The quality is definitely not equivalent to CD quality, in both, Internet and car radio, interruptions and changes in the quality might occur. A client who is used to listening to the radio in his or her car accepts these quality degradations and, thus, is willing to use a service like Internet radio. The explicit provision of live events in the Internet is an example for the lack of alternatives. In this case users accept a degradation in quality, since there is only one possibility for receiving the live event. The broadcast of live events, like pop concerts or sport events, even lead to partial collapse of the service caused by the high user demand.

The situation is completely different in cases in which alternatives in a much better quality are available. Since the mid 90s several Video-on-Demand (VoD) trials were performed but none of them resulted in a major success. In fact, there are only a few VoD services available in the Internet. Despite the failure of these trials a huge effort has been put in the goal to overcome the problems that avoid VoD and video streaming from becoming a successful service in the Internet. Recent developments show that VoD is, at least in some areas, gaining popularity. This tendency is certainly supported by new technologies that allow users at home to receive data at a higher bandwidth and, thus, better quality. Yet, there are still open issues that have not been solved so far. Two of these issues, which are considered in this thesis, are the absence of Quality of Service (QoS) in the Internet and the heterogeneity of the clients. Both require new mechanisms that allow an adaptation of the streaming rate to available network and client resources. Furthermore, an integration of these new mechanisms with a video distribution architecture is necessary to increase the overall scalability of VoD services. In this thesis, these new mechanisms and their integration in a video distribution architecture are presented.

1.1 Motivation

In the last few years, the Internet has experienced an increasing amount of traffic stemming from the emergence of multimedia applications which use audio and video streaming [2]. This increase is expected to continue and be reinforced since access technologies like Asymmetric Digital Subscriber Line (ADSL) and cable modems enable residential users to receive high-bandwidth multimedia streams. One specific application which will be enabled by future access technologies is Video on Demand (VoD). True VoD (TVoD) [3] is a subtype of VoD which allows users to watch a certain video at any desired point in time while also offering the same functions as a standard VCR (i.e., fast forward, rewind, pause, stop). The challenges of providing TVoD in the Internet are manifold and require the orchestration of different technologies. Some of these technologies like video encoding (for example, MPEG-1) are fairly well understood and established. Other technologies like the distribution and caching of video content and the adaptation of streaming mechanisms to the current network situation and user preferences are still under investigation.

Existing work on TVoD has shown caches to be extremely important with respect to *scalability*, from the network, as well as from the video servers' perspective [4]. Scalability, of course, is an important issue if a TVoD system is considered to be used in the Internet. Yet, simply reusing concepts from traditional Internet Web caching is not sufficient to suit the special needs of video content since, for example, popularity life cycles can be very different [5].

In addition to scalability, it is very important for an Internet TVoD system to take the "social" rules implied by Transmission Control Protocol's (TCP) cooperative resource management model into account, i.e., to be *adaptive* in the face of (incipient) network congestion. Therefore, the streaming mechanisms of an Internet TVoD system need to incorporate end-to-end congestion control mechanisms to prevent unfairness against TCP-based traffic and to increase the overall utilization of the network. Note that traditional video streaming mechanisms rely on open-loop control mechanisms, i.e., on explicit reservation and allocation of resources. As it is debatable whether such mechanisms will ever be used in the global Internet, e.g., in the form of RSVP/IntServ [6], we do not assume these but build upon the current best-effort service model of the Internet which is based on closed-loop control exerted by TCP-like congestion control. Yet, since video transmissions need to be paced at their "natural" rate, adaptiveness can only be integrated into streaming mechanisms in the form of quality degradation and not by delaying the transfer as is possible with elastic traffic such as File Transfer Protocol (FTP) transfers. An elegant way of introducing adaptiveness into streaming is to use layer-encoded video [7] as it allows dropping segments (the transfer units) of the video in a controlled way without high computational effort of, e.g., adaptive encoding as described in [8]. Thus, it overcomes the inelastic characteristics of traditional encoding formats like MPEG-1 or H.261. In addition, adaptive streaming in combination with an adaptive encoding format like layer-encoded video can avoid uncontrolled losses and, thus, increase the perceived quality of a video in contrast to an uncontrolled streaming. A side-effect of adaptive streaming is the fact that heterogeneous clients and access networks can be supported more efficiently.

Little work has been performed so far on the aspect of combining both, scalability for VoD systems and adaptive streaming. Thus, the focus of this thesis is on new mechanisms that combine the benefits of both approaches in order to maximize the quality of the video stream that is delivered to the client. However, while the combination of caching and adaptive streaming promises a scalable and TCP-friendly TVoD system, it also creates new design challenges. One drawback of adaptive transmissions is the introduction of variations in the number of transmitted layers during a streaming session. These variations affect both the end-users' perceived quality and the quality of the cached video and, thus, the acceptance of a service that is based on such technology.

The motivation for this thesis can be expressed by the following question: Can the benefits of system scalability and adaptive streaming be combined to create new systems that can increase the performance of VoD services?

1.2 Goals

The goal of this thesis is to answer the aforementioned question by extending existing mechanisms and creating new ones to increase the performance of VoD systems in today's Internet (an Internet without Quality of Service support). The validity and applicability of these mechanisms are proven through investigations based on assessment, simulation, and a prototype implementation which in combination lead to the final results of this thesis. These new mechanisms should be usable as building blocks for scalable Internet VoD systems. Next to the development of the individual mechanisms it should also be shown how these mechanisms can be orchestrated to build a well suited distribution infrastructure for VoD services which is in contrast to approaches where only isolated parts of the distribution infrastructure are investigated. Nevertheless, the mechanisms should also be usable independently from each other to allow VoD operators to tailor a service based on these mechanisms according to their specific needs. For example, the mechanism that reduces quality variations, which is located on the cache, should be independent from the transport mechanisms between server and cache. It is certainly not the goal of this thesis to build a specific VoD application.

It is a fact that in an Internet without QoS support quality variations and data loss during a streaming session cannot be avoided. Thus, those quality variations should be kept to a minimum in order to increase the acceptance of VoD services. The minimum of quality variations can also be seen as the maximum number of variations that the viewers tolerate. As a consequence, an intolerable number of variations would lead to the fact that users do not accept the offered service. Investigations on the subjective impressions of quality variations in layer-encoded videos were not performed so far, which might reveal such information. Thus, this thesis tries to gain better insight in how variations in layer-encoded video affect the viewers' perceived quality by conducting such a subjective assessment.

Based on this newly gained knowledge it is investigated if caches can be used to improve the quality of a layer-encoded video stream that is transported from or through the cache to the client. In other words, how can these layer variations, with the aid of caches, be kept to a minimum. As a constraint, the mechanisms at the client used to receive and display the video should be kept

unchanged. This decision is based on the fact that the establishment of new mechanisms are far easier to perform on a manageable number of servers and caches compared to the vast amount of uncontrollable clients. Next, the new mechanisms should be designed in a way that allows for heterogeneous clients and access networks.

Since scalability in a VoD service means, among other things, to increase the number of simultaneously served clients, it is important that server load is reduced by streaming data directly from caches to the client. Thus, it is also important not only to minimize layer variations in the stream delivered to the client but also to minimize these variations in the cached version of the video to allow the delivery of a high quality video from the cache. Figure 1.1 shows the elements of a VoD service and the systems that are in the focus of this thesis.

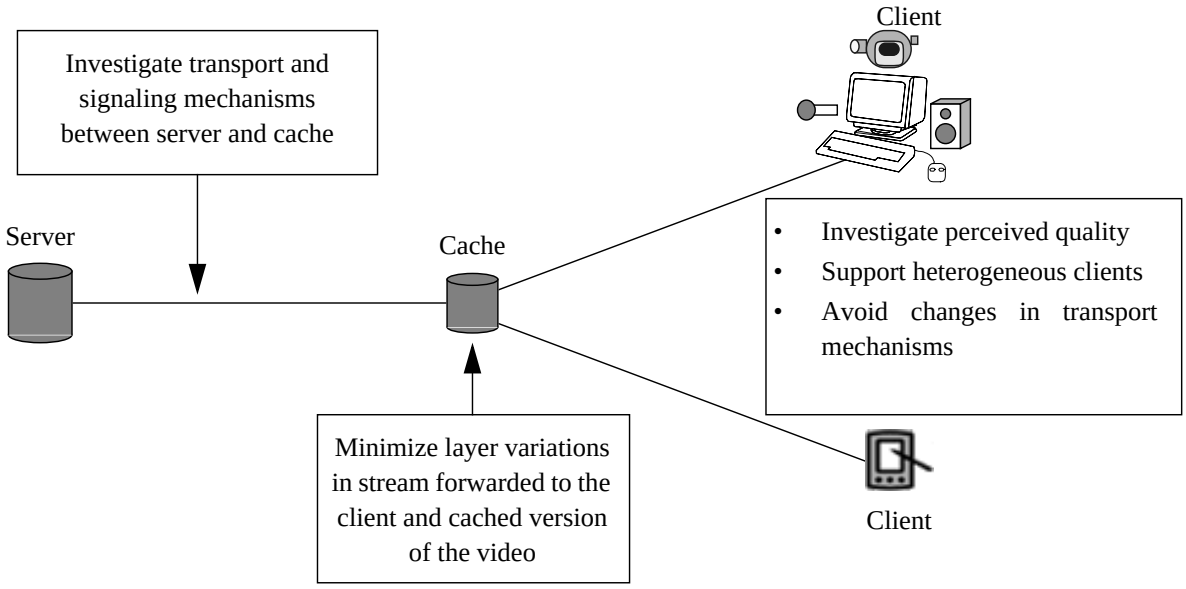


Figure 1.1: Goals of this thesis

1.3 Outline

Chapter 2 gives an overview of the Scalable Adaptive Streaming (SAS) architecture which combines system and content scalability in order to allow VoD services in a best-effort Internet in combination with heterogeneous access networks and clients.

In Chapter 3 an overview of related work in the area of video streaming and distribution is given.

A survey on related work in the area of retransmission scheduling revealed a lack of subjective investigations on how layer variations in layer-encoded video influence the viewer's perceived quality. Existing work on retransmission scheduling is based on speculative assumptions. Thus, as a first consequence a subjective assessment is performed in the scope of this thesis to get better insights in how layer variations affect the perceived quality. The subjective assessment is presented in Chapter 4.

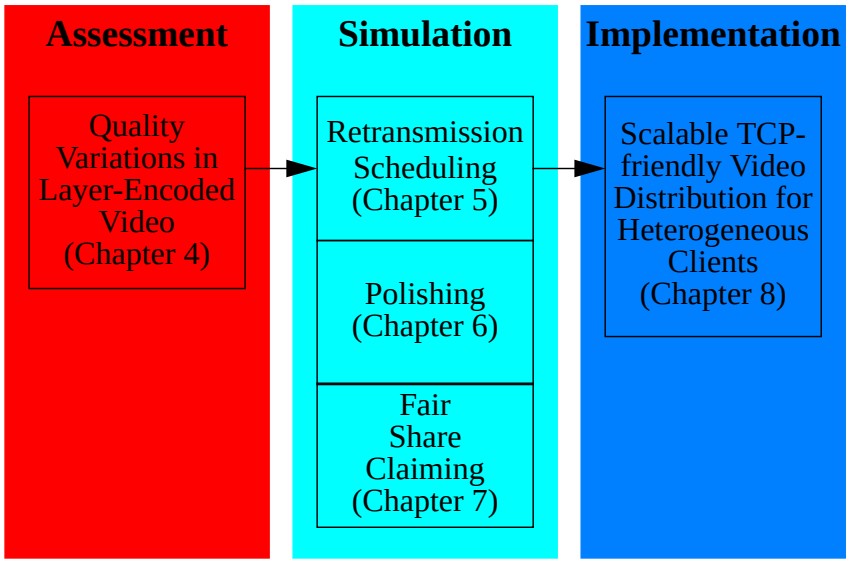


Figure 1.2: Investigation areas of this thesis

The results of this investigation are used to confirm the applicability of an objective metric which is developed in order to evaluate heuristics for retransmissions scheduling. The investigations on the latter (presented in Chapter 5) show that an optimal solution, given reasonable computing power, is computationally infeasible.

Additionally, the results of the subjective assessment reveal that an increase in the amount of stored data for a cached layer-encoded video object does not necessarily increase its perceived quality (see Section 4.5). That means, dropping certain segments of a layer to reduce the amount of variations can increase the perceived quality. Based on this knowledge reducing layer variations by dropping certain segments seems to be an additional option to improve the perceived quality, leading to a new mechanism called *polishing* which is presented in Chapter 6. Polishing can either be used for cache replacement or during playout from the cache to the client. In the first case, segments are deleted from the cache, based on the polishing algorithm, in order to free storage space for new video objects, while in the second case certain segments are not streamed from the cache to the client.

With a simulative environment solely built to investigate the newly created mechanisms for retransmission scheduling and polishing, a series of simulations are performed. The goal of the simulations is to show the applicability of both mechanisms and their dependence on certain parameters (e.g., the available bandwidth for retransmissions). The results obtained by the simulations were satisfying and showed, in the case of retransmission scheduling, a significant improvement compared to already existing mechanisms.

In subsequent work, which is presented in Chapter 7, a new mechanism allowing the transport of segments requested for retransmission is developed leading to a combination of TCP-friendly streaming and retransmission scheduling. This approach has a side-effect allowing a TCP-friendly transport stream to claim its fair share on the network path, although layer-encoded video is trans-

mitted. Based on the mechanism for fair share claiming an implementation design for an already existing streaming platform is made.

In Chapter 8, this design is extended to allow scalable TCP-friendly video distribution for heterogeneous clients. Therefore, the cache is extended by gateway functionality enabling standard clients in the SAS architecture. Based on this extended platform experiments are performed to demonstrate the applicability of the newly created mechanisms which are building blocks of the SAS architecture.

Finally, a summary of the contributions created in this thesis is given and final conclusions are drawn.

Chapter 2: Scalable Adaptive Streaming Architecture

This chapter gives a general overview of the Scalable Adaptive Streaming (SAS) architecture. SAS allows VoD services in today's Internet which is characterized by best-effort service and a wide heterogeneity regarding end-systems and access networks.

In Section 2.1 it is shown that a video distribution system is a special subclass of distributed systems. In a more general sense scalability issues of distributed systems are discussed and mapped to the specific scenario of video distribution. Since replication and its subclasses are means to increase scalability in distributed systems, they are introduced and discussed within the scope of video distribution systems in Section 2.2. Terminology for elements used in a video distribution system is not consistent in literature. To avoid confusion, the terminology used throughout this thesis is introduced in Section 2.3. Section 2.4 gives a detailed presentation of the architecture that is based on the new mechanisms presented in this thesis. It is shown how video distribution without these mechanisms has been performed so far. Additionally, the drawbacks of such an approach are emphasized. Furthermore, the advantages of caching based on the access characteristics for videos which lead to load reduction on the origin server and an increased system scalability are presented. Subsequently the two major building blocks of the SAS architecture, system and content scalability, are introduced, and the benefits of combining both, system and content scalability, are presented. A typical video distribution scenario, reflecting the situation in the current Internet, is given in Section 2.5 to clarify why it is important to combine system and content scalability in the SAS architecture. To emphasize the positive effects of combining both, system and content scalability, an example application for SAS is presented in Section 2.6.

2.1 Distributed Systems

In this section, it is motivated that a video distribution system can be seen as a subclass of distributed systems as they are known in traditional computer science. A distributed system can be defined as a *"collection of independent computers that appears to its users as a single coherent system"* [9]. The applicability of this definition to video distribution systems can be demonstrated by the following example. If we assume a user wants to watch a specific video, the user might access that video through a link on a web page. By choosing that link the video client application is started, which transparently to the user, decides where to retrieve the video data from. Thus, the user is completely unaware of whether the video is located on a video server, a cache, or even its local disk.

Applications based on distributed systems are manifold and a video distribution system is only one possible example. Other prominent examples are the world wide web (WWW), distributed

file systems like CODA [10] or applications used for GRID computing [11]. In a more general way one can speak of a distributed system, if several computing devices are cooperating with each other.

One major issue of distributed systems is scalability. A distributed system that scales can deal with an increasing amount of users while its growth keeps performance degradation and administrative complexity to a minimum.

In the case of video distribution systems there are three major classes of scalability as shown in the following:

- **User scalability:**
  A user scalable distributed system allows one to add more users and resources to the system. A good example for *user scalability* is the world wide web. The tremendous increase in number of users in the WWW necessitated the use of caches that reduced the load on web servers. Therefore, it is possible to satisfy the higher demand caused by the increase in users. If this rise continues, additional caches can be installed to keep the systems scalable.
- **Geographical scalability:**
  The WWW is also a good example for *geographical scalability.* Information is offered from virtually any location in the world and users can access this information no matter where their location is. A distributed system is geographically scalable if effects caused by a wide geographical distribution of the system are hidden. A user who requests a web document and is served by a local cache receives the document with the same performance as if it had been requested from a local server, although the original document might be located somewhere else.
- **Content scalability**:
  Content scalability is usually applied to multimedia content like audio and video or even pictures. It characterizes content whose storage size and bandwidth requirements for transmission vary. MPEG-1 Audio Layer III (MP3) objects are a relevant example for this type of scalability. The standard [12] allows different types of encoding (the sampling frequency is varied) and, thus, the resulting MP3 files and the required bandwidth for transmission are of different sizes. In combination with multimedia objects, content scalability is also related to the quality of the object. An MP3 audio file generated with a higher sampling rate has a better quality than an MP3 file generated from the same original with a lower sampling rate.
  Content scalability can contribute to the overall scalability of a distributed system. A specific example for content scalability in a distributed system is discussed in more detail in Section 2.4.5.

2.2 Replication

Replication is one technique to increase scalability in a distributed system:”*Replication not only increases availability, but also helps to balance the load between components leading to better*

performance." [9]. Therefore, a distributed system should make use of replication in order to increase its scalability.

One special form of replication is client initiated replication [9] which is also called *caching*. It is distinguished from server initiated replication by the fact that the decision to create a replica is made by the client of a resource and not by the owner. Scalability is increased in a similar way as with replication.

In the following sections, the terms replication and caching are discussed within the focus of video distribution systems. The distinction between replication and caching in the scope of video distribution systems is necessary because a certain class of content can make the application of one or the other technique more suitable. The distinction between replication and caching can also be observed in the WWW where content is actively distributed (replicated) or autonomously cached. For the first case mainly overlay networks like CDNs (content distribution networks) are used for the distribution while in the second case no such overlay networks are necessary.

2.2.1 Server Initiated – Replication

Replication is more efficient than caching in the case of a low *read-to-update* ratio. That is, it is very likely that few read requests are made to a replicated content until the replica must be updated. In addition, replication provides consistency by actively updating the replicated content. News is a popular example for content that should be replicated, i.e., actively pushed into local storage nodes. The content of a news video might change several times during a day and it might be requested by many users during a certain time interval. If, in addition, several news videos are created that are only of regional (geographical) relevance, the distribution system can determine to which storage nodes a specific news video should be pushed.

The main advantage of replication is load leveling which can be performed to reduce the origin server's load. For example, the popularity of a news video might be very high for a short duration. Without replication the origin server might not be able to handle all requests for that video. Pushing the video into local storage nodes reduces the possibility of overloading the origin server. Thus, applying the replication technique to news videos is efficient because the amount of network traffic is reduced. Additionally, it is scalable because server load is reduced and, thus, more clients can be supported, and the reliability of the system is increased. For example, in the case of a server failure the news video can still be streamed from the local storage node, although if the failure period exceeds a certain amount of time, consistency cannot be guaranteed anymore. It was shown, in this context, that the location of the replica (i.e., the storage node chosen to store the replica) has an important influence on the scalability of the distribution system and the availability of the content [13]. Other types of content that can be distributed by replication are all kind of objects with a very high popularity, e.g., the latest block buster movie in a VoD system.

Replication has the drawback that the scalability of the system is constrained by the way updates for an object are performed. In contrast to caching, an additional mechanism is needed that keeps track of the existing replicas in a content distribution system in order to be able to update those

replicas, if necessary. It might also occur that a replica is created on a storage node but is never requested by a client thus, leading to unnecessary resource consumption.

2.2.2 Client Initiated – Caching

When the *read-to-update* ratio is relatively high or hard to predict, an approach that is based on the caching technique is more efficient. In this case, a copy of the original object is only created on the cache, if a client requests this object and the local caching policy on the cache decides that this object should be cached. This technique is efficient for objects which are of a lower popularity and are updated rarely or never. Objects that fit into this category are, e.g., less popular movies, on-line lectures, or recorded sport events. The latter might be interesting for users who did not have the chance to take part in the live broadcast of the event. As in the case of replication the amount of network traffic can be reduced if a cached object is requested more than once from the cache. In contrast to replication, caching has the advantage, that objects are only distributed to a cache at which a request for this object has been made from a client. A distribution architecture that is based on caching increases the scalability of the system, since server load is reduced and more caches can be added in case the amount of users increase. Reliability can be increased with caching because the content of a cached object rarely changes. Therefore, server outages or link failures between server and cache can occur for a longer period of time and still clients can be served from the cache. Caching is not as constrained by the update procedure as it is the case for replication because the server does not have to keep track of the caches that keep a copy of its content. The cache itself is responsible for the invalidation of cached content. Depending on the behavior of the cache a server might even not recognize that a copy of its content is stored on a cache. Yet, it is not clear if content providers would allow autonomous caches in a distribution infrastructure. This decision is mainly based on copyright issues [14] and is not in the focus of this thesis.

Table 2.1: Comparison of replication and caching

Replication	Caching
owner initiated	client initiated
low *read-to-update* ratio	high *read-to-update ratio*
increases system scalability	increases system scalability
increases reliability	increases reliability
suited for content that is of high popularity for a short duration, e.g., news	suited for content with lower or unknown popularity, e.g., on-line lectures

2.3 Video Distribution System Terminology

Before the presentation of the specific video distribution architecture that was designed in the scope of this thesis the building blocks that are necessary to build video distribution systems are introduced in more detail. Since the terms to identify the elements that constitute a video distribution system are not named identically in literature, this introduction is necessary in order to avoid confusion. A possible configuration of a video distribution infrastructure is shown in Figure 2.1.

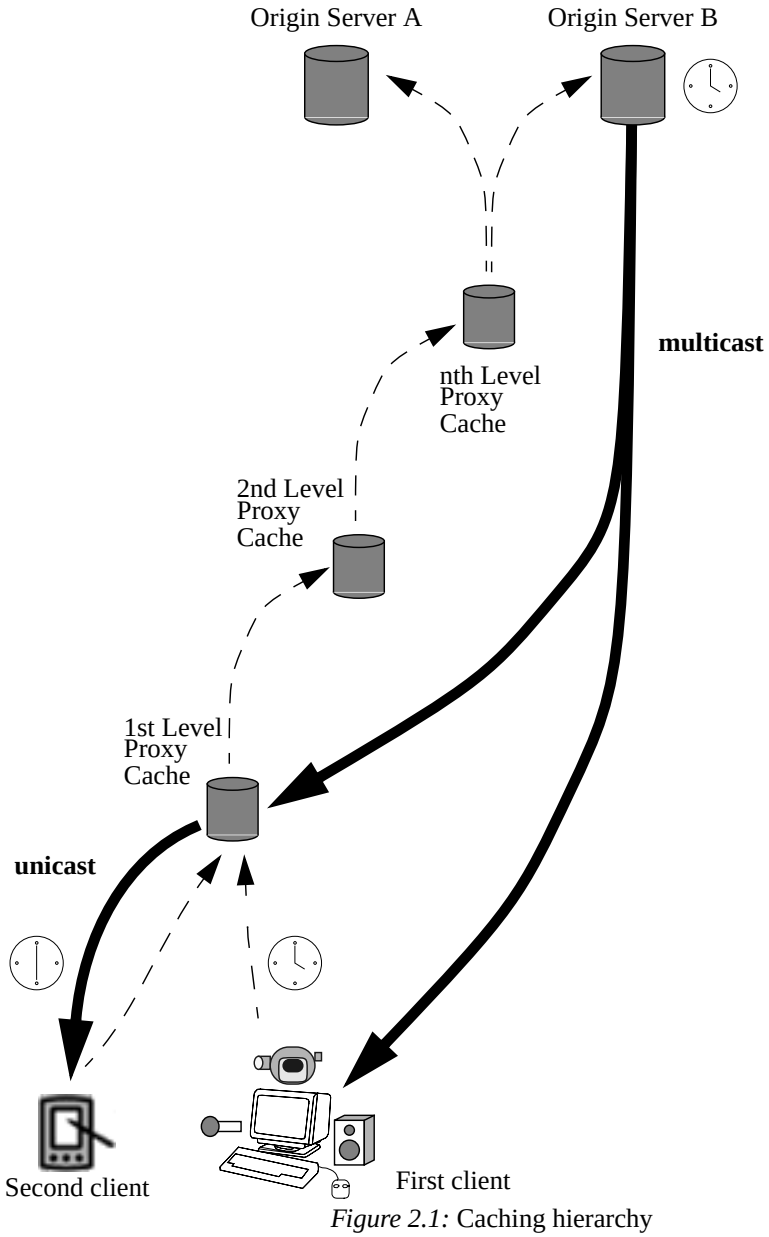


Figure 2.1: Caching hierarchy

2.3.1 Origin Server

Origin servers store the original version of a video object. Those servers are in general controlled by content providers. That means, the content provider decides which content it offers and at

which point in time new content is offered by actively storing this content on the origin server. The content provider might also grant or deny a cache to store a copy of the original content on its local storage. In the scope of this work it is assumed that origin servers and proxy caches are owned by a single or cooperating CDN operators. Therefore, proxy caches can obtain copies of the caches without any restrictions. For reasons of simplicity the term *server* is used to denote an origin server.

In the case of a hierarchical distribution infrastructure (see Section 2.3.5), the origin servers are located at the top level of the hierarchy.

2.3.2 Proxy Cache

A request for a specific video object is directed from the client to its nearest proxy cache. The proxy cache has then two options to deal with this request: a) if the requested video object is already cached it simply starts streaming this object to the client, b) if not, it forwards the request to another proxy cache or an origin server. Content discovery mechanisms like the ones provided by the Internet Cache Protocol (ICP) [15] or the Real Time Streaming Protocol (RTSP) [16] might be used to forward the client's request to a node of the distribution infrastructure that stores the requested object. Depending on the local caching strategy the proxy cache decides if it caches the requested object or not. The outcome of the caching decision causes two different scenarios:

- **Object is cached:**
  The cache informs the originator of the content that a local copy of the requested object is created on the cache. Based on the distribution mechanisms the sender of the stream might either set up a multicast stream, which is joined by proxy cache and client or a unicast stream that is forwarded through the proxy cache to the client (see Figure 2.1). The issue of how a reliable transport between server and cache can be achieved is addressed in Section 3.3.4.
- **Object is not cached:**
  In this case, there is no general need for sending the stream to the cache. Simply streaming the data from the server to the client is sufficient.

In addition to the functionality that must be provided by a proxy cache (e.g., cache replacement, extended signaling, etc.), it also offers the same functionality as an origin server. Thus, in terms of provided functionality, a proxy cache can always be seen as an extension of the origin server.

In literature *proxy cache* and *cache* are used interchangeably, while always this definition of a proxy cache is meant. Throughout the remainder of this document the term *cache* is used.

2.3.3 Cache Replacement

Since the available storage space on the cache is limited, cache replacement has to be performed on a cache to increase the efficiency of a cache. Cache replacement is not limited to video caches but is also applied in all kinds of caches, like memory caches or web caches [17]. Cache replacement mechanisms were originally developed for memory cache. With the upcoming of web

caches, replacement strategies for those caches became an interesting topic. Unfortunately, these strategies cannot be applied directly to video caches since the characteristics of the objects that are cached are different. Video objects usually have a high read-to-update ratio compared to typical web objects, like web pages. In addition, transfer times are much higher and the ratio of cache size to object size is much smaller. In the case of a web cache and a video cache with the same storage space the first can store a larger number of objects than the latter due to the difference in object size. An introduction to cache replacement for video caches is given in [4].

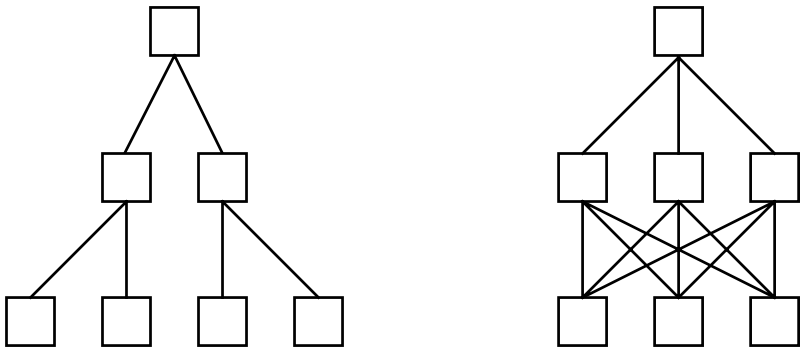
2.3.4 Client

Throughout this thesis, it is assumed that all clients are connected to the Internet. This is achieved via different access technologies, like Local Area Networks (LANs), ADSL, cable modems, wireless networks, or modems. The clients can be all kinds of devices that are able to deal with multimedia data (in this specific case video streams). For example, clients are set-top boxes, standard PCs, PDAs, and mobile phones. This implies a very heterogeneous environment concerning characteristics such as access bandwidth, computing power, and display capabilities of the client. Each client obtains the information to which cache its requests for a video object should be directed. This can be done manually by setup parameters or, if the object is requested via a web server, through additional HTTP [18] information.

2.3.5 Logical Overlay

Another important issue is the placement and interconnection between servers, caches, and clients. One traditional approach is a hierarchical distribution approach as shown in Figure 2.2. In this case, the servers are located at the top level of the hierarchy while caches are located in the intermediate levels and clients always on the lowest hierarchy level. Client requests are directed to caches of the lowest hierarchy level and only forwarded to caches in the next higher level, if necessary. The distribution hierarchy can consist of a different number of levels with a minimum of at least two levels which would lead to a video distribution system only consisting of servers and clients. Another approach is often described as *cooperative caching* [19] in which the forwarding of requests is not as restricted as in the case of a hierarchical distribution infrastructure. With this type of infrastructure caches can cooperate with each other independent from their location in the overlay. Thus, requests can be forwarded to any cache or even server in the infrastructure. A comparison between the two types of distribution infrastructure is shown in Figure 2.2. The links between the single entities of the infrastructure represent the possible path of request messages and do not reflect the physical layout of the infrastructure.

The new transport mechanisms presented in this thesis are suited for both kinds of hierarchies. For reasons of simplicity, further examples and measurements were only made on the basis of a hierarchical infrastructure. This does not restrict the presented transport mechanisms, since the performed investigations and design decision for these mechanisms are not limited to a certain type of distribution infrastructure as it is shown throughout the thesis. Nevertheless, cache replacement mechanisms can depend on the logical overlay. For example, with cooperative content a new



Hierarchical Cooperative

Figure 2.2: Logical overlay

object must not necessarily be delivered from a cache or server that is located in a higher hierarchy level. The cache replacement presented in Chapter 6 is independent from the distribution infrastructure, since only local decisions about content that should be discarded from the cache are made.

2.3.6 Video Object

Any video content that can be streamed via the Internet is defined as a video object. An example for a video object is an MPEG-1 [20] encoded movie that has a duration of approximately 90 minutes No restrictions to the encoding format, the duration, and the actual content of the video object are made. Thus video objects could also be short music video clips or lectures as they are provided in distributed learning systems like *eTeach* and *BIBS* [21].

2.3.7 Video-on-Demand (VoD)

The term VoD is widely used for systems that allow one to watch a certain video content at any point in time via communication systems like cable TV, satellite, or the Internet. VoD is used to describe on the one hand services like pay-per-view that allow viewers to watch a movie on a digital TV channel (either cable or satellite) with the restriction that those movies are only offered at certain points in time. On the other hand VoD describes services like True-VoD (TVoD) that allow a client to watch a video content immediately after its request. According to [3] there are five different types of services which are classified on the level of interactivity allowed:

- **Broadcast (NoVoD):**
  There is no interactivity in NoVoD. One example for this type of VoD are traditional TV channels.
- **Pay-per-view (PPV):**
  Viewers sign up and pay for a specific program but still cannot make any influence on the way, when or how the content is shown.

- **Quasi VoD (QVoD):**
  Viewers can be grouped together based on the threshold of interest. For example, viewers interested in sport events are joined in a group at which soccer games are broadcasted. The viewer has rudimentary control by the ability to change the group and, thus, receiving a different type of content but interactivity is not supported.
- **Near VoD (NVoD):**
  Interactivity like fast forward and reverse is only allowed in certain time intervals. Even when a viewer decides to start watching it might take a certain amount of time until the content is displayed. One possibility to realize NVoD is the broadcasting of the same content on different broadcast channels while the start of each broadcast is shifted in time. For example, a movie with a total length of 60 minutes is broadcast over 12 channels with a time shift of 5 minutes. In this scenario, a viewer might have to wait up to 5 minutes until the playout starts, and the granularity for fast forward and rewind is limited to 5 minute segments as shown in Figure 2.3. The limitation to 5 minutes for fast forward and rewind is caused by the fact that these operations are realized by switching to a different channel.

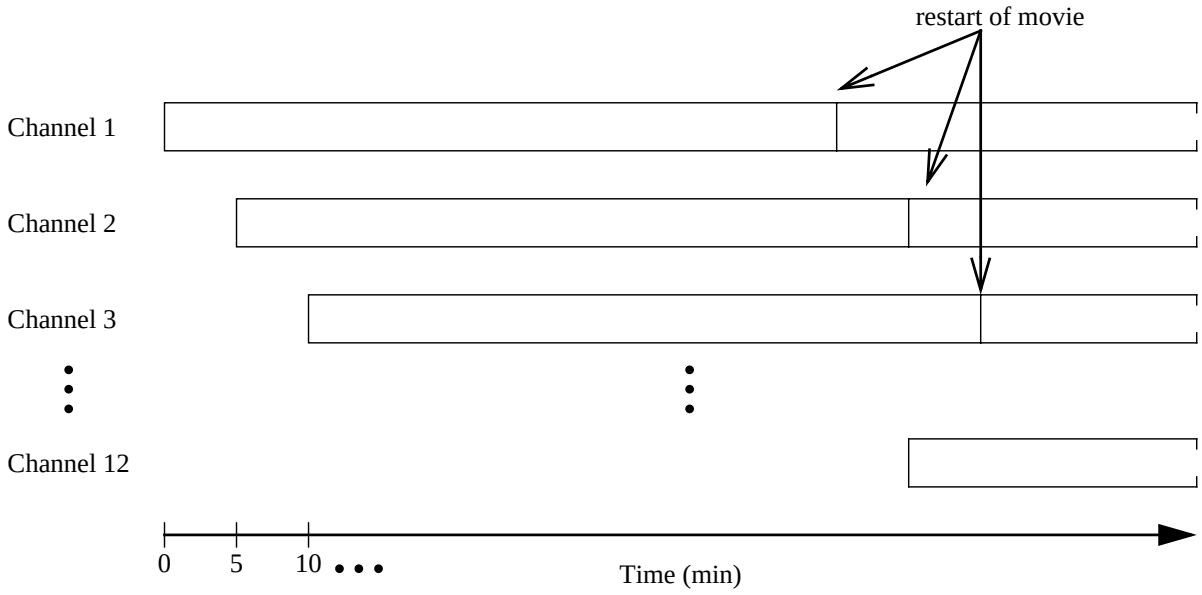


Figure 2.3: Near VoD

- **True VoD (TVoD):**
  In this case, the viewer has complete control over the session presentation. The viewer can jump to any position and perform operations similar to the ones that are offered by a VCR. In the case of TVoD one channel cannot be shared by several viewers. From all of the five types TVoD is the most challenging, since for each single request a new channel (i.e., a new stream) must be set up and it is not guaranteed that the video object is streamed linearly. The viewer might perform non-linear operations like fast-forward, fast-rewind, or pause.

In the scope of this thesis the Internet is regarded as the underlying communication system for VoD, with the focus on True VoD. The decision to focus on TVoD was driven by the fact that it offers the best possible service in terms of interactivity and, therefore, increases the attractiveness of VoD in the Internet compared to other services which are already available via traditional broadcast media (e.g., PPV). In Section 2.6, an application is presented where the level of interactivity is increased by the offered TVoD service.

2.4 Architecture

2.4.1 Situation

After the identification of the elements that form a video distribution infrastructure, the *Scalable Adaptive Streaming* (SAS) architecture is presented in this section.

The main goal of this architecture is to support the efficient distribution and streaming of video data in today's Internet. Most of the existing approaches require techniques that are not fully established in the Internet so far or might probably never be established. One example is the work that has been performed on broadcast and multicast schemes in order to reduce load on video servers (see Section 3.3.4). All of the existing approaches assume that the bandwidth between client and server is sufficient and no congestion on those links occurs. Thus, those approaches can only be reasonably applied in environments where bandwidth reservation between the different entities of the distribution system can be performed, e.g., as with open-loop control mechanisms like RSVP [22]. Unfortunately, those mechanisms are not deployed in the Internet and it is rather debatable if those mechanisms will ever be used in the global Internet. *Prefix caching* (see Section 3.3.4) is an exception that assumes that bandwidth between server and cache might not be sufficient. It has the drawback that only (temporal) parts of a video are cached and, thus, in the case of a link or server failure, the streaming of the video cannot be continued.

The architecture presented in this thesis is based on the techniques that are available in the Internet of today but is also designed in a way that allows adaptation to possibly new mechanisms and techniques in a future Internet and increases to fault tolerance of video distribution systems.

Although there has been an immense amount of work on quality-of-service (QoS) [23], the only service that is currently offered in the Internet is best-effort. That means, resources in the Internet cannot be reserved and, thus, no guarantees for a certain service (e.g., a guaranteed amount of bandwidth, packet loss, and latency on the link between two nodes) can be offered. Since traditional streaming approaches make use of UDP as the underlying transport protocol, one might naively assume the lack of service guarantees in the Internet less problematic as it would be the case for TCP. Yet, the lack of congestion control mechanism in UDP leads to an unfair behavior against TCP traffic. So far, UDP traffic in the Internet makes up only a small fraction of the total amount of traffic. This could change in the near future if, e.g., streaming applications become more popular, leading to a situation in which the performance of TCP-based applications is heavily affected due to non congestion controlled traffic. Several investigations on streaming media in the Internet have shown that VoD is becoming more popular, and it is very likely that this

increase in popularity continues in the near future. For example, a study in 1997 [24] did not report on any remarkable amount of streaming media, while a study in 1999 [25] reported that 14% of the total amount of transferred web traffic belonged to audio and video content [26].

One solution to overcome this problem of TCP-unfairness is to perform streaming also in a congestion controlled manner. In recent years, there have been several proposals on such TCP-friendly mechanisms [27] which are presented in Section 3.3.2.

Recent developments in the end-system market also increased the heterogeneity of end-systems and access links that are used by those systems. In the beginning of the Internet, end-systems were mostly located in local area networks built on Ethernet or Token-Ring technology thus, access networks were mostly homogenous. With the increasing popularity of the Internet, caused by the success of the WWW, and new access technologies like wireless LAN (WLAN) or DSL, the situation has changed. Future scenarios might even include all-IP based wide-area wireless networks, as it is discussed for 4th generation wireless networks. Therefore, a video distribution architecture must be able to not only provide mechanisms for TCP-friendly streaming, but also allow adaptation to a wide spectrum of access link capacity and end-system characteristics.

In the following sections the benefits of using caches in a video distribution infrastructure and two classes of scalability on which the SAS architecture is based are presented. A Scalable Adaptive Streaming architecture allows one to perform video distribution in the Internet of today that is characterized by best-effort service and a wide heterogeneity regarding end-systems and access networks.

2.4.2 Advantages of Caching

Studies on access patterns and file characteristics of video objects [28, 29, 26] have shown that caching can be advantageous for the following reasons:

- Most video applications follow the write-once-read-many principle, hence, cache consistency is not a major issue.
- Accesses to videos exhibit a strong temporal locality. That means, it is highly probable that an object that was accessed recently with a high frequency will be requested again soon.
- Broadcast (see [30] for an overview) and multicast mechanisms (e.g., [31], [32], [33], and [34]) used to reduce the load of the original server can only be used for QVoD or NVoD, but not for TVoD.

Additionally, caches for video objects share the well known benefits introduced by web caches:

- The fault tolerance of the system is increased since a failure of the server does not necessarily lead to a service interruption for the client, if video objects are cached completely.
- Caches store content closer to the user and, thus, reduce start-up latency and network load.
- The load generated at the server is reduced and distributed over several caches.

Yet, in contrast to web caches the characteristics of the data to be stored are very different. High quality video files are much larger than most web pages and, therefore, different caching strategies are used in caches for VoD systems [4]. The distribution process for video files is further complicated by the fact that the transmission is much more time and bandwidth consuming. Finally, the popularity models for video caches are different to the ones for web caches [5, 26]. Nevertheless, investigations on the popularity of video objects have shown that approximately 80% of the user requests are concentrated on 20% of the total amount of available videos ([35], [36], [5]). These results indicates that caches can be a very effective means in relation to scalability and fault tolerance in a video distribution system.

It is not the goal of this thesis to optimize video distribution by investigating several caching concepts (hierarchical, cooperative) or to investigate cache replacement strategies like it was done in [4]. The focus here is on transport issues related with TCP-friendly streaming and caching which can be used by either hierarchical or cooperative caching and does not effect cache replacement strategies. With the aforementioned benefits of caches in video distribution systems in mind, the SAS architecture was designed with caches being one major building block of the infrastructure.

2.4.3 VoD without Scalable Adaptive Streaming

Before introducing the SAS architecture a short description of a video distribution infrastructure that is not supported by SAS is given. This description is given to clarify the differences between a traditional and a SAS-based distribution infrastructure.

Without SAS, video is streamed via standard UDP which does not allow an adaptation to the actual network conditions on the route between server and client. In addition, the content offered in such a VoD system is not scalable. This results in two possible scenarios depending on the available bandwidth on the links between server and client. As long as the bandwidth on the links is sufficient the service can be offered to the viewer in full quality. In the case of insufficient bandwidth the service quality decreases to a point where it is unacceptable for the viewer to watch the video. Due to uncontrolled data losses the resulting quality of the presented video is either very bad or, even worse, complete frames or sequences might not be rendered due to missing data (see Figure 2.4). Thus, such a VoD service can be seen as a *binary* service: either the quality of the client is at its best or it becomes very bad. Any steps in between do not really exist. It is obvious that the resulting quality depends on the amount of lost data, but it cannot be controlled which of the data is lost during the transmission. Investigations on MPEG-1 video sequences [37] have shown that selective losses of only 1% of the total amount of data can decrease the quality to a level where the content is not recognizable anymore. An investigation by Boyce and Gaglianello [38], which measured the losses of UDP-based streaming sessions in the public Internet, has shown that loss rates as low as 3% can lead to frame error rates as high as 30%.

An additional problem is the support of clients like mobile devices which might not have sufficient computing power to render and display the content if offered in a non-scalable format.

Another approach to avoid these uncontrolled losses is the combination of a TCP-based transport with a sufficient buffer at the client. This solution has the drawback that the start-up latency can be high (the buffer has to be filled initially) and that re-buffering might occur in the case of a buffer underrun. Both effects are quite annoying for the viewer.

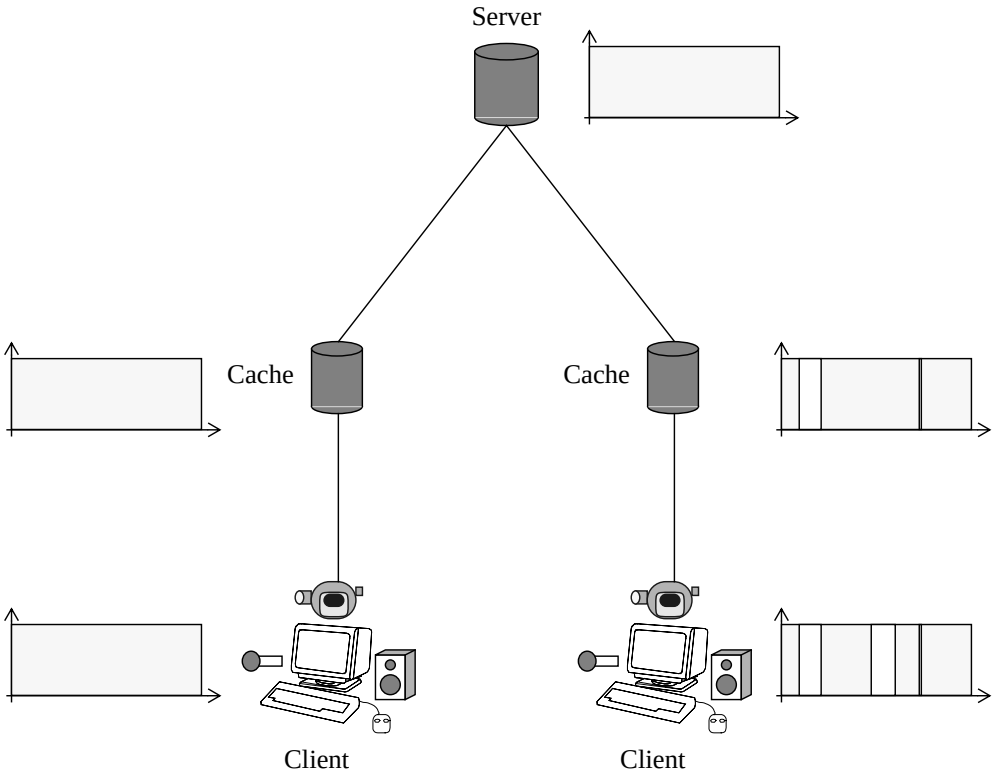


Figure 2.4: VoD without Scalable Adaptive Streaming

2.4.4 System Scalability

A distribution system based on SAS combines *user* and *geographical* scalability (see Section 2.1) in what is referred to as *system* scalability. The decision to combine both scalability concepts into one is caused by the fact that the introduction of caches affects both, as is shown in the following.

As with traditional web caches, caches for TVoD systems allow one to store content closer to users, reduce server and network load and increase the system's fault tolerance. It is important to mention that system scalability here means the scalability of the whole video distribution system, in contrast to approaches that increase the scalability of a single component. For example, an enormous amount of work has been performed on the issue of single server scalability. Although analysis of media streaming workload has shown that the application of techniques like *Patching* [39] or *Bandwidth Skimming* [40] are justified, they are not sufficient. First of all, the fault tolerance of the system is not increased because without caches the server would still be a single point of failure. In addition, video objects are less likely to be stored near the client if no caches are involved. Thus, start-up latency might be increased and quality degradation of the stream might occur because the potential of a link failure or congestion is more likely on a WAN connection.

Figure 2.5 depicts an example for system scalability where hierarchical caching is used as a caching concept. As caching method so-called write-through caching[1] is employed, where a requested stream is either forwarded through the cache or is streamed via a multicast group which client and caches join, if the cache replacement strategy decides to store the requested video on the cache.

Subsequent clients can then be served from the cache (see Figure 2.5). This technique has a lower overall network load in a TVoD system than a method where the video is transported to the cache in a separate stream using a reliable transmission protocol (e.g., TCP) [41]. On the other hand, write-through caching requires a reliable (multicast) protocol to recover from packet losses. The design and implementation of such a protocol, called Loss Collection RTP (LC-RTP), which fits particularly well in a TVoD architecture and employs write-through caching is presented in Appendix A. As can be seen in Figure 2.1 the SAS architecture can increase fault tolerance in a video distribution system. Especially, if we keep in mind the temporal locality characteristics that were observed for video traffic in the Internet. Caching complete video objects would allow one to still serve a large amount of clients despite a server or network (between server and cache) failure.

In addition both, the usage of caches and write-through caching, reduce the server and network loads and, thus, increase the number of users that can be supported by the system. In the next section, content scalability, which allows congestion control for streaming applications, is introduced.

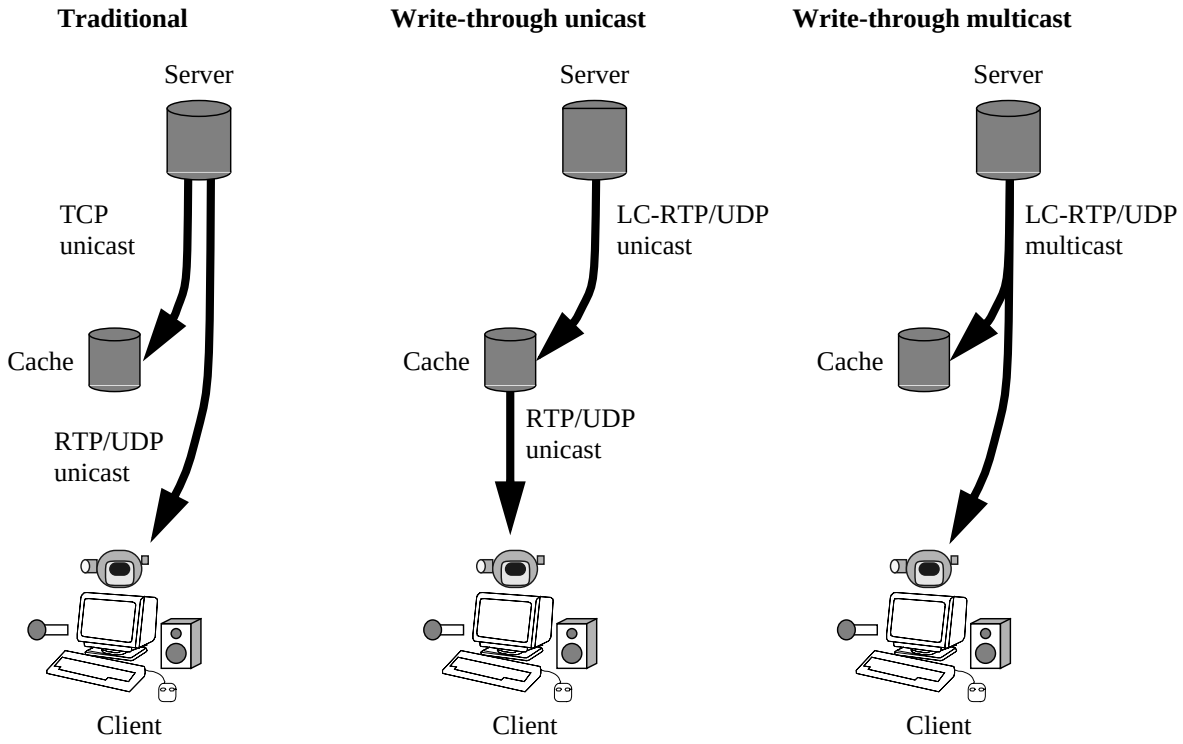


Figure 2.5: Comparison of transport methods for caching

[1] Adopted terminology from memory hierarchies.

2.4.5 Content Scalability

In contrast to elastic applications whose traffic can be spread over time in order to adapt the transmission rate to the available bandwidth on the network, quality adaptation is needed to enable congestion control for inelastic applications like streaming.

For elastic applications like FTP, data may not arrive at the receiver at a certain point in time. This is different from inelastic applications where the delivery of the data is time-critical. For example, in the case of a MPEG-1 video that is streamed to the client certain parts of the video must arrive at the client at specific deadlines to allow correct representation at the client.

However, quality adaptation does not solely serve congestion control purposes but also satisfies the needs of the large variety of heterogeneous clients that exist in the Internet. Even in cases where, due to the provision of network QoS mechanisms in the Internet, congestion control must not be performed, clients might be connected via access networks that have different bandwidth characteristics. Thus, clients are able to receive the requested content with different rates. A client which is directly connected to an Ethernet LAN might be able to receive an object with a rate of several Mbps, while a client connected to XDSL might be limited to receive an object with a rate of several hundred kbps. This could not be realized if the content offered on the server would only be a monolithic encoded file, such as an MPEG-1 file.

In today's video distribution systems the heterogeneity problem is solved by offering only low-bandwidth streams, thus, punishing clients which could receive higher bandwidth streams leading to a better perceived quality of the video. Investigations on RealVideo over the Internet [42, 43] state this assumption, since none of the requested video objects had a bandwidth higher than 500 kbps.

Hierarchical layer-encoded video, i.e., video that is encoded in base and enhancement layers which have hierarchical relationships, represents a suitable method to allow for this quality adaptation. Besides the layer-encoded format there are other alternatives like adaptive encoding or switching between different encodings of one original content [44] (also described as *dynamic stream switching (DSS)* or *Simulcast*). If both techniques (DSS and layer-encoded video) are compared with each other, it becomes obvious that layer-encoded video has the advantage that it can adapt to network conditions in finer granular manner. With DSS that would require to store a larger number of different encodings (regarding the transmission bandwidth) of one original content on the server or the cache. For example, to offer the same quality levels with DSS as a layer-encoded video consisting of 3 layers, 3 independent video objects, each encoded with a different rate, must be available. In addition, the switching between objects of different rates at the client or the cache cannot be performed as easily as in the case of layer-encoded video. For example, in the case of a video that is encoded in several MPEG-1 bitrates (alternatives) a switching between these alternatives can only be performed at the next I-frame. Switching on a random position in the video is not possible due to the intra-coding characteristic of MPEG-1 [1].

Recently, a new scalable encoding scheme, called *multiple description coding* (MDC) [45] has been developed. It can also be categorized as a layered encoding scheme with the advantage that the layers (i.e., descriptions) are independent from each other which is in contrast to hierarchical

layer encoding. Since MDC is a relatively new research area, no investigation on congestion controlled streaming of such data have been performed. Thus, it is not clear if MDC can be used in combination with this kind of data transport. A known disadvantage of MDC is the fact that the coding efficiency is not as good as with layer-encoded video due to the higher amount of redundancy that is introduced in MDC [46].

2.4.6 Combining System and Content Scalability

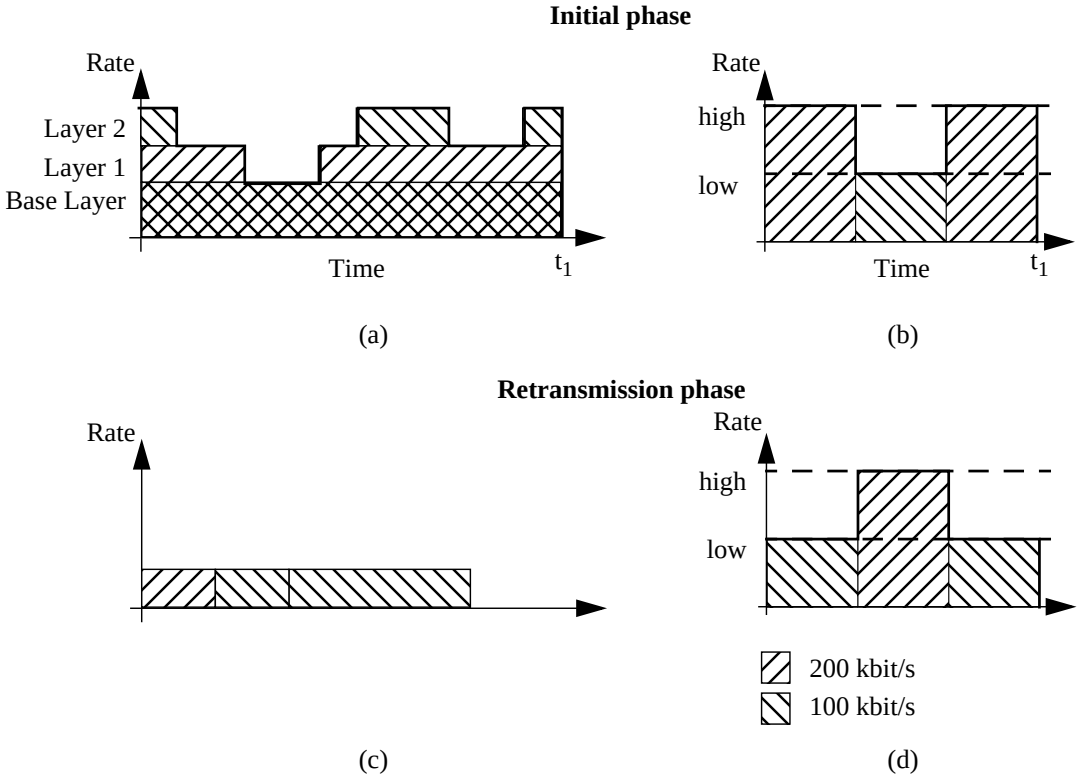


Figure 2.6: Initial cached video quality.

As already mentioned, TCP-friendly streaming is a major issue in SAS. There have been several proposals on how to achieve TCP-friendly congestion control using hierarchically layer-encoded video transmissions, e.g., [47], [48] or [49]. Developing yet another TCP-friendly protocol for video streaming is out of the scope of this work but existing proposals are used since they are already very effective.

Figure 2.6 (a) and (b) show two possible versions (layered and DSS) of a cached video if a TCP-friendly video transmission is combined with write-through caching. Obviously, in both cases, the cached copy of the video exhibits a potentially large number of missing segments from different layers (or complete encodings as in the case of DSS). Note that the exact shape of a cached video content is a function of the congestion control mechanism being used by the TCP-friendly protocol.

Clients that request the same content at a later point in time and would be served from the cache do not have the chance to receive the video in full quality if no extra measures are taken. Thus, a mechanism is required that improves the quality of the cached content. Figure 2.6 (c) and (d) depict the parts that would be identified by such a mechanism and be transmitted from the server to the cache leading to a full quality copy of the video object. The benefit of using layer-encoded video instead of DSS becomes even more clear if we compare the amount of data that has to be additionally transmitted for both encoding techniques. In the case of DSS complete parts of an encoded video must be transmitted instead of only missing segments of certain layers as it is the case for layer-encoded video (see Figure 2.6 (c) and (d)). Thus, with layer-encoded video less network resources and storage space at server and cache is consumed for the transmission of missing segments.

Due to the distinct advantages of layer-encoded video compared to the other methods that allow adaptive streaming presented here and in Section 2.4.5, only layer-encoded video is regarded in the remainder of this thesis.

In the following transmissions of missing segments from the server to the cache caused by the cache's request for these segments are called retransmissions. At first, this definition might be confusing because some of the missing segments might have never been transmitted at all (due to congestion). On the other hand, the cache cannot distinguish between packets which were not sent at all and dropped packets (e.g., queue overflow on one of the intermediate routers). Thus, for the cache every packet that was not transmitted initially appears as a retransmitted packet.

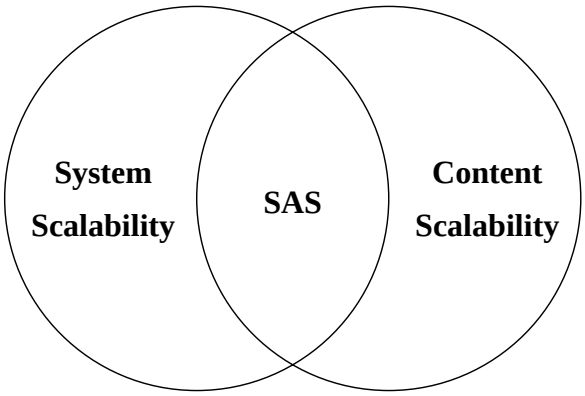


Figure 2.7: Combination of system and content scalability

2.4.7 VoD with Scalable Adaptive Streaming Support

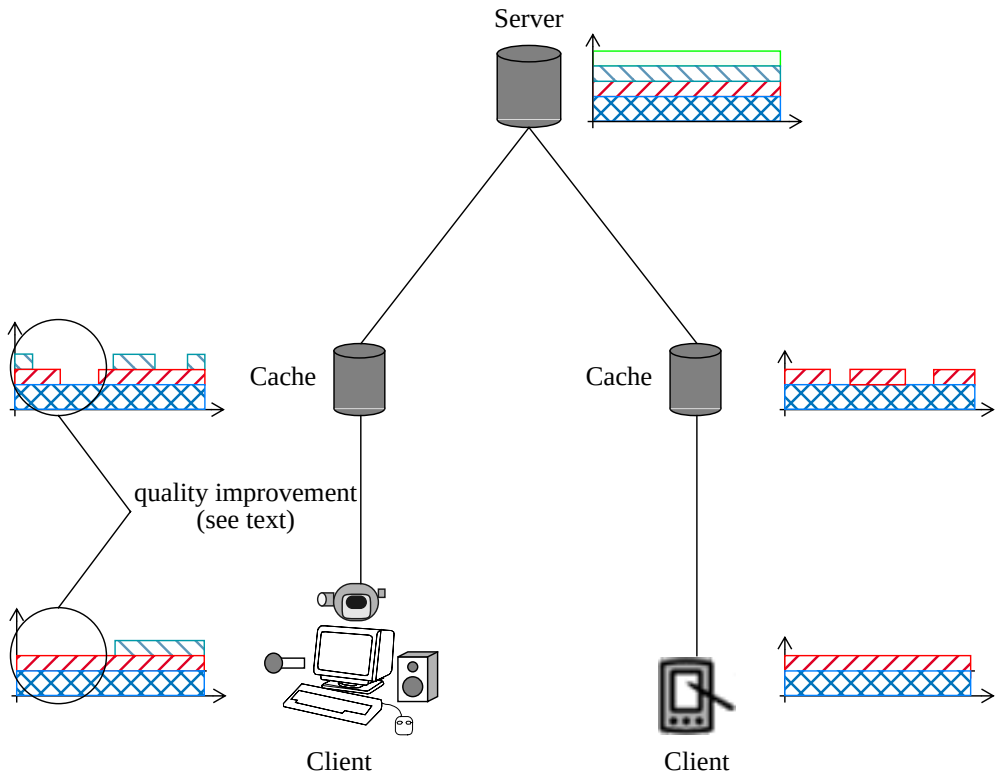


Figure 2.8: VoD supported by scalable adaptive streaming

Figure 2.8 shows the same basic architecture as in Figure 2.4 with additional Scalable Adaptive Streaming support. In contrast to the binary service as it is described in Section 2.4.3, this new video distribution architecture allows more than two quality steps. This is mainly caused by the fact that the multimedia objects (e.g., videos) are encoded in a scalable format and, in addition, the streaming is performed adaptively to the conditions in the network. As it is shown in Figure 2.8 video distribution with SAS support allows to deliver a video stream in more than two quality steps to the client. In addition, the scalable content is also well suited to support a large variety of clients (PCs, handhelds, ...). This thesis presents new mechanisms, which are part of the SAS architecture, that also try to deliver the video in the best possible quality to the client. The two sections in Figure 2.8 which are marked by a circle give a simple example that demonstrates how the quality improvement can be achieved. The initially cached version of the video misses some segments and, thus, would lead to a lot of quality variations if the video would be streamed to the client as it is stored on the cache. With the new mechanisms presented in this thesis the amount of quality variations can be reduced as it is shown for the video that is finally streamed to the client. The mechanisms used to reduce the amount of quality variations are presented in Chapter 5 and Chapter 6.

2.5 Scenario for SAS

To clarify why it is important to combine both kinds of scalability in a video distribution system that suits well for today's and the future Internet, a scenario that uses the SAS architecture is presented in the following section. The goal is to present how a video distribution infrastructure can benefit from the integration of system and content scalability. The example tries to reflect a typical scenario as it can occur in today's Internet.

The scenario shown in Figure 2.9 depicts a heterogeneous distribution system consisting of two subnets that are connected to the Internet backbone. In each of these subnets a cache is located to which all client requests are directed. Subnet A has a wireless infrastructure in which only homogeneous clients (in terms of link bandwidth) are connected, while Subnet B has a heterogeneous infrastructure. In the case of Subnet A it becomes clear that the maximum of two layers is sufficient for all clients since all clients are connected via a homogeneous access network. It is assumed that the original videos stored at the server consist of four layers, as shown in Figure 2.9.

In Subnet B, the content might be first requested by a handheld (client 2) in a lower quality due to its restricted access bandwidth. A subsequent client requesting this content might be a high-end PC which would like to receive the content in a better quality and has an access link with high bandwidth characteristics. Its capabilities allow for additional transmissions from the server to the cache in order to improve the quality of the cached content. Depending on the order client 1 and client 2 are requesting the video, it might occur that client 1 receives a stream from the cache that has reduced quality because no higher quality is available at the cache. This can be caused by two reasons, either the stream was originally requested by client 2 which did not request more than two layers or, because of space constraints the cache replacement strategy might have dropped the two upper layers. If a retransmission mechanism is used, this situation could be circumvented. After initially delivering the stream to the client 2 the cache starts to request missing segments of certain layers or even complete layers if upon a request from client 1. This would be the case for segments from the second and third layer as shown in the scenario in Figure 2.9.

In some cases it might not be useful to cache all layers of a video, as it was already mentioned for the case of Subnet A. It is assumed that a certain amount of video objects are already stored on the cache and none of those videos was requested in a higher quality than two layers by the clients which are served from this cache. Now the cache can presume that none of its clients can receive a video in a better quality and for the caching process of a new video no more than two layers would be cached. In addition, the clients can also give hints about the maximum rate they are able to receive data with (as shown in Chapter 8). In the retransmission phase only missing segments of these two layers would be requested. In addition, this information could be used for the cache replacement strategy. Storage space needed for the caching of new content could be gained by deleting all layers above the second layer instead of deleting complete videos. Applying such a mechanism can lead to a 2-dimensional cache replacement where one dimension is the quality (number of layers) and the other dimension is the time (e.g. as in prefix caching (see Section 3.3.4)).

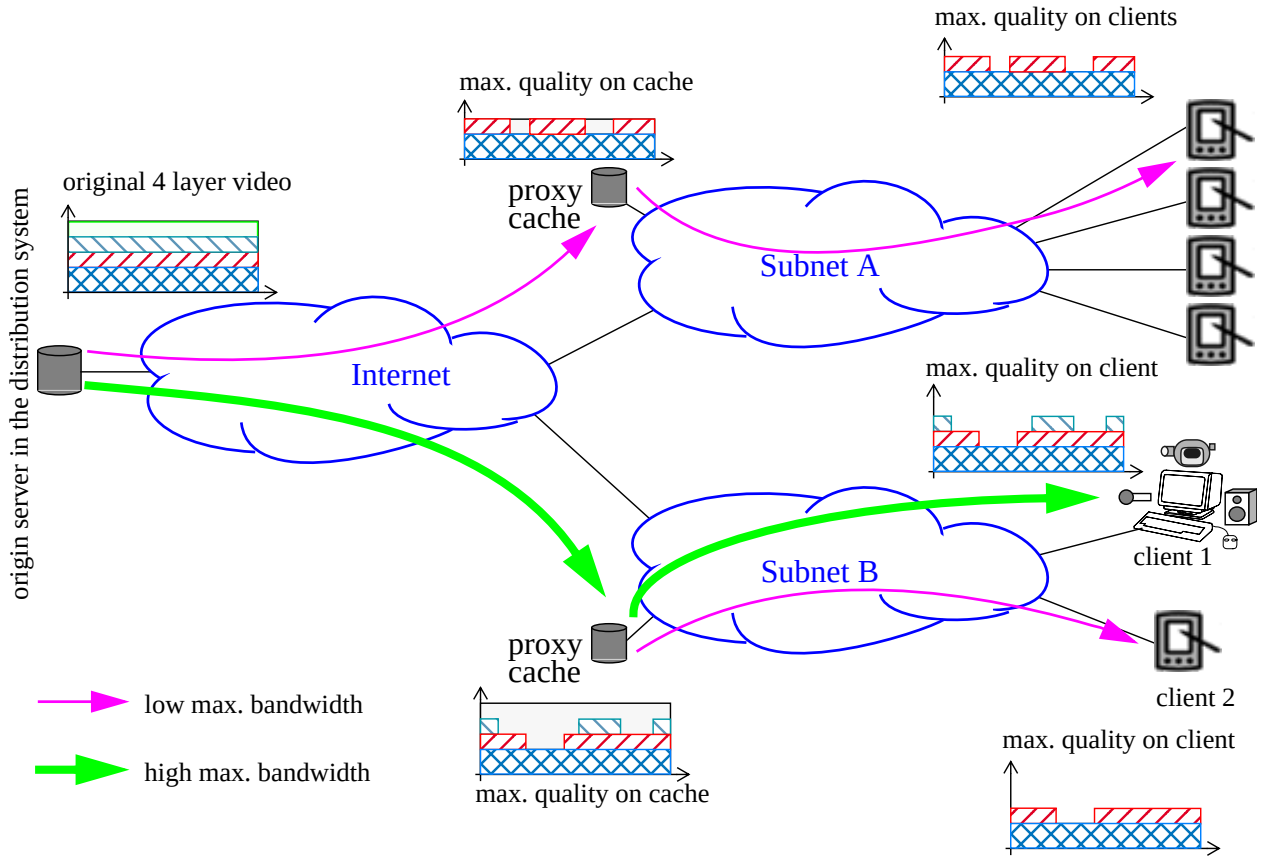


Figure 2.9: Scalable video distribution system for heterogeneous clients

2.6 An Example Application for SAS

In this section, an example application that could benefit from the incorporation of the SAS video distribution architecture is given. The goal of this section is to motivate the application of the SAS video distribution architecture in the Internet. Note, that the goal of this thesis was not to develop a complete system that allows video streaming. This example is rather thought to give the reader more detailed information about the context of this work. Also, the application described below is not the only application that could benefit from the SAS architecture. Other possible applications are TVoD service that offers a large variety of video objects having a large variance in their popularity (e.g., not only the latest block buster movies but also classics are offered) or business TV of a large company that has locations in different regions, countries or even continents. In this section, distance learning is the application scenario that is used to demonstrate an application of the SAS architecture.

In Europe, it is very common that students spend a certain amount of time at another European university as part of their studies. Unfortunately, not all students get the opportunity to attend the official european exchange program (ERASMUS) because of many reasons. Offering on-line courses would allow students to study with loosened time and location constraints. A person having a regular job could attend an on-line course at a university because there is no necessity to

attend the regular courses during day time. One way to offer on-line courses is to record the regular course that is given on a regular (mostly weekly) basis and make this course available for all subscribed students. There is also an advantage for the lecturers which might record a course in advance having in mind that at the regular date another important event might happen (e.g., a conference the lecturer wants to attend). The course can be made available on-line at its scheduled date and every student has the chance to "attend" it. Not only broadcasting the course (e.g., via multicast) as a live stream but making it also available for on-demand requests bears the advantage that students can "attend" the on-line course when it fits in their time schedule. Another advantage of this method is that on-line courses could be shared between universities. For example, two universities might agree to offer on-line courses for all students of both. An example for on-line course sharing between universities is the German *ULI* project [50]. Another example that simply offers courses on-line is the MIT Open Courseware project [51].

These on-line courses would offer students a broader variety of courses and allow them to train their language skills, if, e.g., lectures from universities located at other countries are provided. Yet, arranging time schedules between universities might be hard or even impossible. This could lead to the situation that there might be an interesting on-line course at a remote university but this one is given at exactly the time a local lecture is given which the student wants to attend. Thus, the student would not be able to attend the on-line course if it is only distributed as a live stream. The need for on-demand on-line courses is even higher if one imagines that such courses are offered across the borders of continents (e.g. Europe and North America) due to the different time zones. Also the beginning and end of a semester are not identical between different universities. Some of the on-line courses might become quite popular at a certain university because students might recommend them to each other or the lecturer is a well known person in a specific research area. Thus, several students from one university will watch this on-line course, but it is very unlikely that all of them will watch it at one point in time. Measurements of user access behavior in a distance education application [29] have revealed a high temporal locality. That means, several requests for the same video object occur often within a short time span. Caching the on-line courses at a cache located in the campus network of the local university would be beneficial in this case because:

1. Load on the origin server at the remote site is reduced since, after caching the lecture at the local site, subsequent requests are not directed to the origin server.
2. Fault tolerance is increased because students might still be able to watch the lecture although the original server or the path to it has failed.
3. Overall network load is reduced since only one complete lecture has to be streamed across the Internet backbone. In all other cases only local resources are consumed.
4. Once the file is cached the need for an update is highly improbable considering that a specific lecture is given at the earliest in the next semester. Until then the video object representing the lecture should be discarded from the cache a long time ago.

Even students at the university might use different access technologies to connect to the campus network and the Internet. For example, students in dormitories might be connected directly to the

campus network via LAN technologies like Ethernet while some universities already offer wireless access on campus (e.g., WLAN) to their students. Finally, students might use ADSL, cable, or traditional modems to get access to the campus network. This is very similar to the scenario as shown for Subnet B in Figure 2.9. An additional important fact is that students might be unwilling to pay for a certain service class that would ensure QoS guarantees on the link between the server, the cache, and the client. Thus, an architecture for the distribution of on-line courses across universities should be well suited for the best-effort service class, which is so far the only available class. This example also shows that such a system is not outdated if other service classes become available in the future Internet. Although service guarantees might be offered in the future students might still be connected via access networks that do not allow the reception of the video stream in full quality.

Chapter 3: Related Work

As the SAS architecture combines several research areas there is a large amount of related work to this field. Therefore, this chapter is structured in three main parts. The first part (Section 3.1) is concerned with available products in the area of video distribution systems while the second part (Section 3.2) regards the related work of standardization bodies. The work on video distribution has gained a lot of attention which is shown by the fact that products have become available and the IETF inaugurated a new working group for content distribution (Content Distribution Internetworking (CDI)). Nevertheless, both available products and the work in the IETF mainly aim at the goal to simply move content closer to the receiver, increase fault tolerance and reduce server load.

The last and most important part is presented in Section 3.3 and covers the related work that has been performed in research. Here, a lot of effort was put in the issue of single server scalability, which includes topics like disk access, memory management, and techniques that increase the delivery capacity by efficiently using network resources. Since these topics are not in the main focus of this thesis, the interested reader is referred to [30].

Layer-encoded video is an important component of the SAS architecture and, thus, an overview on related work on scalable encoded video is given in Section 3.3.1. Related work on TCP-friendly streaming which allows in combination with scalable encoded video adaptive streaming, is presented in Section 3.3.2. Section 3.3.3 shows existing approaches of adaptive streaming. The existing work on caches is very broad. The most important areas of this research work are presented in Section 3.3.4. It shows the work on partial caching which can be subdivided in time-based and bandwidth-based caching. This work is relevant for SAS since layer-encoded video can be cached partially both in the time-based and bandwidth-based domain. Finally, reliable transport of video data and cache clustering to increase the scalability of caching are additional related topics.

In some specific cases it seems appropriate to discuss the related work in the context of the new approaches presented in this thesis. This allows one to show the differences between existing and new approaches in more detail. Thus, additional related work is also mentioned in the following chapters of this thesis as necessary.

3.1 Products

Caused by the increasing popularity of video streaming in the Internet more and more products are available that offer streaming applications. These products can be divided into three major categories:

- Server and client application

- Server, client, and cache application
- Cache application only

The first category is not of much interest, since it does not offer mechanisms for scalability as they are included in the SAS architecture. One example for the second category are the products offered by Real Networks [52] which offer a complete solution for video distribution in the Internet. There is very little information available about Real's products but this information reveals that neither TCP-friendly congestion control nor caching in the traditional sense is performed. In the case of a cache miss the data is streamed directly to the client and a second, reliable (TCP-based) connection is established to the cache that is used to transport the requested video to the cache. Although streaming can be performed in an adaptive manner by using a flavor of DSS, called *SureStream*, the mechanism to adapt is based on RTCP feedback information and, thus, can hardly be considered as congestion control. The restrictions on the amount of RTCP feedback information that can be sent during a certain period is restricted. Therefore, the adaptation to changing conditions in the network takes longer than with TCP or a TCP-friendly mechanism.

The third category is represented by products from Kassena [53], Network Appliance [54], Inktomi [55], Blue Coat [56], and Infolibria [57]. Unfortunately, there is no technical information available about these products, except in the case of Kassena's MediaBase XMP which reveals that they make use of prefix caching [58] in their cache.

Not included in these three categories, because they rather make use of the products offered in the third category, are CDNs like Akamai [59] or Cable & Wireless [60]. Usually, replication is applied in these CDNs as a mechanism to distribute the content into caches.

3.2 Standardization

In relation to the Internet the most important standardization body is the Internet Engineering Task Force (IETF). Virtually, every open standard protocol that is used in today's Internet has been standardized by the IETF. Also most of the available video streaming and distribution applications support IETF standards.

In parallel to the IETF, standards for video distribution were developed by the Digital Video Broadcasting Project (DVB) and the Digital Audio Visual Council (DAVIC).

3.2.1 IETF

Related work to SAS is covered by four different IETF working groups: *Audio/Video Transport, Datagram Congestion Control Protocol, Multiparty Multimedia Session Control,* and *Content Distribution Internetworking.* While the first three working groups belong to the *Transport Area*, the last one is part of the *Application Area*.

A) Audio/Video Transport (AVT)

IETF's *Audio/Video Transport* working group (AVT) is mainly concerned with the specification of real-time transport protocols for audio and video. Work in this group led to the development of the RTP [61] standard and the specification of a series of payload formats for RTP [62]. The standards

of this working group are well known and integrated into many commercial and open source video streaming and conferencing applications. Recently, the working group drafted a new version of the RTP RFC [63] which is mainly concerned with changes to rules and algorithms and leaves the packet format on the wire unchanged. Mainly the timer algorithm that calculates when to send RTCP packets is changed. Thus, these recent changes have no impact on the SAS architecture.

B) Multiparty Multimedia Session Control (MMUSIC)

The control of multimedia communication in the Internet is addressed by the *Multiparty Multimedia Session Control* (MMUSIC) working group. For the control of audio and video data, the *Real Time Streaming Protocol* (RTSP) [16] has been specified while multimedia sessions are described by the *Session Description Protocol* (SDP) [64]. Recent activities in this group are concerned with the creation of a new RTSP version [65] which is mainly concerned with fixing existing flaws in the standard and the next generation SDP (SDPng) [66] which deals with fundamental changes in SDP. Issues related to overcome security flaws in RTSP with the aid of firewalls are presented in more detail in [67] and [68].

C) Datagram Congestion Control Protocol

The relatively new *Datagram Congestion Control Protocol* working group (DCCP) deals with the development and specification of a *Datagram Congestion Control Protocol* [69]. Their goal is to develop a protocol that establishes congestion control for an unreliable packet stream. The way DCCP is designed it must be seen as a framework which determines the general rules of the protocol behavior and a general message format but allows the usage of different congestion control algorithms which are separately specified as profiles. One of these profiles [70] specifies the usage of *TCP-friendly Rate Control* (TFRC) as a congestion control algorithm in DCCP.

As an alternative the *Stream Control Transmission Control Protocol* (SCTP) [71] could be used. It was originally designed to transport Public Switched Telephony Network (PSTN) signaling messages over IP networks, but it is also capable of supporting other applications like video streaming. SCTP is a reliable transport protocol that operates on top of IP. The latter is the major drawback of SCTP. Making use of SCTP would require kernel extensions, since it is not intergrated in most standard operating systems.

D) Content Distribution Internetworking (CDI)

New standards that specify the interoperation of separately administered CDNs are defined by the *Content Distribution Internetworking* (CDI) working group. The goal of this working group is to specify the requirements for CDI rather than the development of new protocols. The group is mainly concerned with the definition of requirements for content distribution internetworking, namely interoperation of request-routing systems, interoperation of distribution systems, and interoperation of accounting systems. An introduction to CDNs, CDI and the specification of a common vocabulary, is given in [72]. A more detailed overview of the activities of this working group can be found in [73].

3.2.2 DVB and DAVIC

The Digital Video Broadcasting (DVB) project was started in 1993 with the goal to develop digital terrestrial TV in Europe. Over the years the work in DVB also covered satellite and cable networks. Since the work in DVB is mainly focussed on broadcasting it is not directly applicable for VoD services. Nevertheless, there exist VoD services that make use of the underlying DVB technology. For example VoD services, like the one offered by Kingston [74], make use of IP on top of DSL and use only the existing infrastructure like cable. To be able to support IP new routers must be added in these networks. This enhanced infrastructure allows the usage of streaming solutions which are devised for IP-based networks. Applications that make use of the new mechanisms presented in this thesis can be applied to such networks. Thus DVB can be seen as an underlying technology that offers an infrastructure which is used by the video distribution system.

The Digital Audio-Visual Council was founded in 1994 with the goal to specify open interfaces for interactive digital audio-visual services. Shortly after its foundation the interest in DAVIC from industry and research was high. With an increasing popularity of the Internet and the standardization of new protocols for streaming ([65], [61]) the focus shifted. This development led to the situation that DAVIC is no longer active. The goal to achieve open interfaces for audio-visual services could not be reached. Today, available video streaming systems that are used in the Internet are not fully compatible [75]. Some of them support a minimum set of compatibility by using the same control protocol (RTSP) but it is not assured that the client of vendor B can communicate with a server or cache of vendor A.

3.3 Research

The work that has been performed in the area of video streaming and distribution in the Internet is very extensive and mentioning all existing work goes beyond the scope of this thesis. The focus in this section is rather on work that is closely related to the work presented in this thesis. A more comprehensive overview is given by Dan and Sitaram [30].

The related work is categorized according to the requirements of the SAS architecture. Basic technologies to support SAS are scalable encoded video and TCP-friendly transport mechanisms which, in combination, allow an adaptive streaming of video data. Both technologies are needed to support SAS' content scalability. System scalability is achieved by the introduction of caches. Therefore, a separate section is dedicated to caching mechanisms for video objects including reliable transport of video data into caches and the work on cache clusters.

3.3.1 Content Scalability - Scalable Encoded Video

In Section 2.4.5 content scalability was introduced as one major component of SAS. It is required to adapt the transmission rate of a streaming session to the available bandwidth on the network. In this section an overview of the work on scalable encoded video is given.

Basically there are four different classes for scalability in the case of video, that is to say spatial scalability, temporal scalability, and SNR (signal-to-noise ratio) scalability [76] (see Figure 3.2).

Spatial scalability offers the functionality to decode images at different spatial resolutions, while temporal scalability allows the adjustment of the frame rate. SNR scalability is achieved, e.g., through the layered quantization of the DCT values [77] or other methods like the *embedded zerotree wavelet* (EZW) [78] algorithm. A new, fourth class of scalability, emerged with the development of the MPEG-4 standard which allows the composition of a video scene from several independent video objects. Since this also allows to decode a subset of objects, *object-based scalability* is introduced as shown in Figure 3.2.

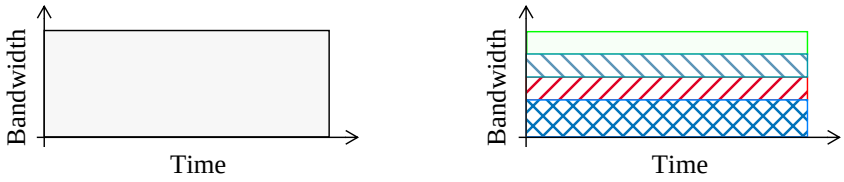


Figure 3.1: Unscalable vs. scalable content

In general, as with encoding formats like MPEG-1, only data loss up to a certain limit allows the reconstruction of a frame. This is different if scalable encoding schemes are applied. In this case, part of the video data is sufficient to reconstruct the video signal with the trade-off that the quality of this video signal is reduced. Additionally, the coding efficiency of scalable encoding approaches is reduced, since a higher amount of redundancy information is needed compared to non-scalable encoding formats. An encoding scheme that makes use of scalability is layered encoding. With layered encoding the video is split into one *base layer* and one or more *enhancement layers*. The base layer contains fundamental coding information and can be decoded without any additional information. Enhancement layers contain additional information that increase the quality of the reconstructed video signal. In contrast to the base layer, enhancement layers are not independent from other layers. To reconstruct the information included in layer, *n* all of the information of the lower layers (*0, ..., n-1*) are needed. If the base layer is missing, no video signal can be reconstructed at all. An introduction and overview about layered encoding techniques are given in more detail in [79]. One specific example for layer-encoded video is SPEG [80] which is used by me for the assessment of variations in layer-encoded video. A detailed description on SPEG is given in Section 4.3.1.

The concept of scalable coding was first introduced in the MPEG-2 [81] and H.263 [82] standards, which allow a two layer encoding (base layer plus one enhancement layer). This was extended with the H.263+ [82] standard which allows several layers. In MPEG-4, the layer scheme of *fine granularity scalability (FGS)* [76], which is basically a two layer scheme but

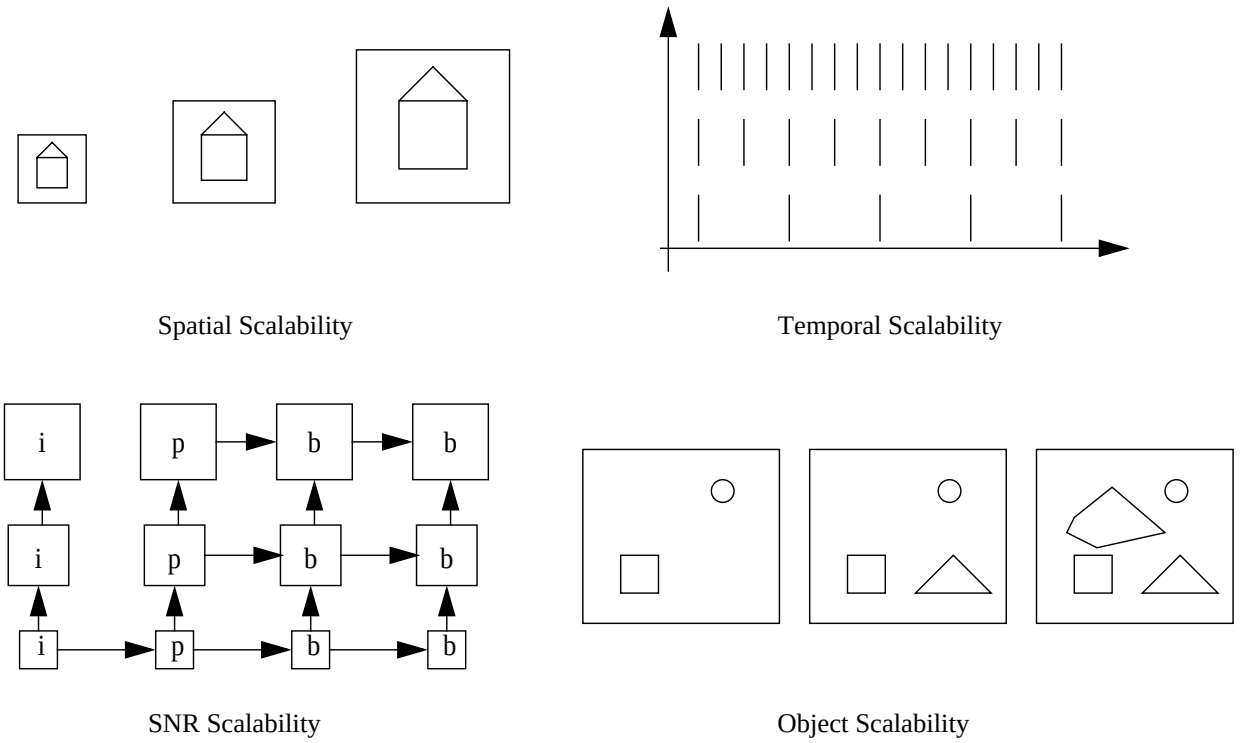


Figure 3.2: Classes of scalability

allows a variable rate enhancement layer (see Figure 3.3), is introduced. With FGS the bit stream can be truncated thus, it adapts exactly to the rate of the transmission channel.

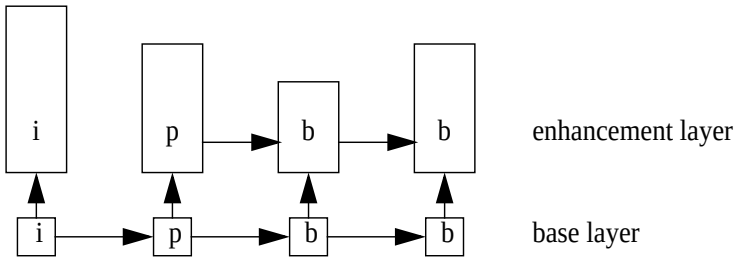


Figure 3.3: Fine granularity scalability

A new scalable scheme that gained attention recently is *multiple description coding* (MDC) [45]. MDC codes a video into two or more descriptions and either description can be used to decode baseline quality video. The quality of the decoded video increases with the amount of descriptions. It has been shown that MDC is well suited for networks that allow *path diversity* [83]. That means, the single descriptions can be sent via different paths to the receiver, thus, avoiding congestion on one path affects all descriptions. Path diversity in the Internet is hard to achieve, since source routing is not widely supported. Using MDC as a scalable streaming format for video distribution systems in the Internet can be beneficial, but only if path diversity is also supported. Future solutions might achieve path diversity by storing each description on a different server as proposed by Apostolopoulus et al. [83]. Yet, there are some issues that have to be investigated in further detail. Simply using different servers for each description does not necessarily

mean that both paths, between the servers and the client, differ much. In addition, the proposed scheme allows only a maximum of two descriptions which is rather restrictive. Figure 3.4 compares a possible scenario at which layer, FGS, and multiple description encoded video would be transmitted over a best effort link.

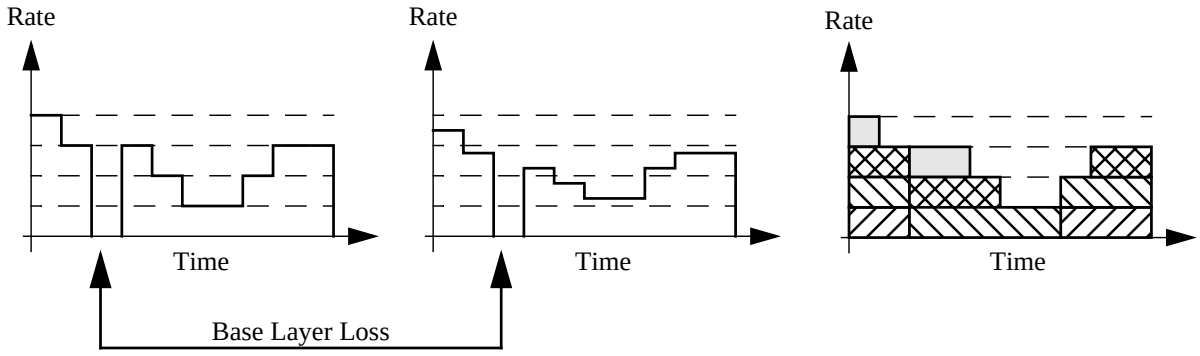


Figure 3.4: Layered, FGS, and MDC video transmission

3.3.2 Congestion Control - TCP-friendliness

Figure 3.5: Non-congestion controlled vs. congestion controlled

Besides a scalable distribution infrastructure, it is very important for an Internet TVoD system to take into account the “social” rules implied by TCP’s cooperative resource management model, i.e., to be adaptive in the face of an (incipient) network congestion. Therefore, the streaming mechanisms of an Internet TVoD system need to incorporate end-to-end congestion control to prevent unfairness against TCP-based traffic and increase the overall utilization of the network.

In recent years several protocols for the transport of non-TCP traffic that incorporate TCP-friendly congestion control were developed. Widmer et al. have published an overview of the approaches [27]. To be applicable for streaming these protocols have to meet the following requirements:

- Rate oscillations must be kept to a minimum
- Modification to the network infrastructure must be prevented (e.g. the protocol stack in the routers may stay as it is)

The existing approaches can mainly be separated in two major categories, window-based and rate-based, respectively.

A) Window-based

MTCP [84] and pgmcc [85] are two examples of a window-based congestion control approach. In this case a congestion window is maintained similar to the one used in TCP. The window size is increased if no congestion occurs and decreased in the case of congestion. Since this approach is quite similar to TCP congestion control mechanism, the resulting transmission rate of window-based TCP-friendly protocols is fluctuating (saw-tooth like) and, thus, the protocols are not very well suited for the transmission of audio and video streams which require a constant transmission rate. Even the use of layer-encoded video would not solve this problem, since the strongly oscillating transmission rate would lead to a high number of layer changes which impair the perceived quality at the client (see Chapter 4).

B) Rate-based

With rate-based congestion control the transmission rate is determined according to a network feedback mechanism indicating congestion. Rate-based congestion control can be subdivided in the AIMD (additive-increase/multiplicative-decrease) and the model-based congestion control schemes. The AIMD scheme (e.g., RAP [47]) behaves similar to TCP's congestion control and, therefore, the resulting transmission rate is oscillating leading to the same problems as described for window-based congestion control. In the model-based approach a TCP model is used to determine the TCP-friendly transmission rate. TFRC [48] is a TCP-friendly protocol which makes use of such an approach. The model is based on the TCP throughput equation as shown in (1).

$$Sendrate \approx min\left(\frac{W_{max}}{RTT}, \frac{1}{RTT\sqrt{4p/3} + Bmin(1, 3\sqrt{3p/4})p(1 + 32p^2)}\right) \quad (1)$$

B=*timeout*

p= *loss rate*

W_{max}= *maximum congestion window size*

The advantage of the model-based approach is the smoothness of the resulting transmission rate. This is caused by the fact that the sending rate is adapted to the long-term TCP throughput.

From our observations ([86] and Chapter 7) and the results presented by Widmer et al. [27], TFRC is very promising as a TCP-friendly protocol for streaming [48]. It is a rate based congestion control protocol with TCP-friendliness over longer time scales. The main advantage in combination with A/V streaming is that the rate is smooth in the steady-state case and, therefore, applications that rely on a constant sending rate are supported. In addition, the protocol is end-to-end based which does not require any modifications of the network infrastructure. TFRC has the advantage over TCP that its transmission rate is not limited by any window size. TFRC is the most prominent of all TCP-friendly congestion control approaches and has recently been considered to become a standard [70] in the IETF. TFMCC [87] is an extended version of TFRC that supports single-rate multicast.

Besides the approaches that were evaluated in [27] there is also related work done by Bansal and Balakrishnan [88] called binomial congestion control which is a non-linear generalization of AIMD. In the case of a packet loss the window is reduced from W to $W\text{-}bW^l$ and an RTT without loss leads to $W+a/w^k$. For $l=0$ and $k= 1$ the algorithm is equivalent to TCP's AIMD.

Additional work with the main aspect to investigate TCP-compatibleness by simulation and analysis is presented by Bansal et al. [89]. The results of this work show that all the investigated algorithms (RAP, TEAR, TFRC, and binomial congesting control) can be deployed in the Internet, since they behave TCP-compatible. In addition, the investigation revealed drawbacks of some of the algorithms and led, for the case of TFRC, to a modification that improves TFRC's behavior in the case of abrupt bandwidth reduction.

3.3.3 Adaptive Streaming - Streaming Layer-Encoded Video Without Caches

Figure 3.6: Non-adaptive vs. adaptive streaming

The work that has been performed on streaming of layer-encoded video can be subdivided in two categories. In the first, layer-encoded video is used in combination with congestion controlled streaming. Rejaie et al. [90] make use of RAP [47] to stream layer-encoded video in congestion controlled manner. In combination with a buffer that is located at the client they address the problem of absorbing short-term fluctuations in bandwidth that are introduced by the congestion control algorithm (sawtooth shape). Thus, buffering a few seconds of a stream can lead to smoother playout due to reduced layer changes. Quite similar to this work is the one presented by Nelakuditi et al. [91] that in addition to [90] uses quality metrics and present new algorithms that maximize those metrics. A similar investigation has been performed by Feamster et al. [92], with the difference that the binomial congestion control algorithm [88] is used. The authors show that their congestion control algorithm performs better in terms of buffer usage and average target bitrate in comparison to the one presented by Rejaie. In [93] the available bandwidth in the network is modeled as a stochastic process and in comparison to the work of Rejaie and Nelakuditi only layer-encoded video consisting of two layers is regarded and the client's buffer is unlimited. Another proposal for adaptive streaming is given by Tan and Zakhor [49]. In addition to their TCP-friendly transport protocol they also present a new scalable encoding scheme that overcomes the shortcomings of hierarchically layer-encoded video. This encoding scheme allows the decoding of upper layer data despite the absence of base layer information.

In addition to the work presented above, investigations on layer-encoded streaming in the context of multicast distribution were performed [94, 95].

One of the first applications of scalable content has been receiver driven layered multicast [94] which allows many heterogeneous receivers to take part in a multicast streaming session. This application was rather designed for the streaming of live events and, therefore, interactivity is not supported. To enable a higher level of interactivity, which is needed to support TVoD, multicast can only be applied in a limited scope. Thus, caches gain even more importance in a distribution system in order to reduce the server's load. In addition, making use of scalable content such as layer-encoded video is also a valid means in video distribution systems that support TVoD and have to adapt to current network conditions and heterogeneous clients.

3.3.4 System Scalability - Caches

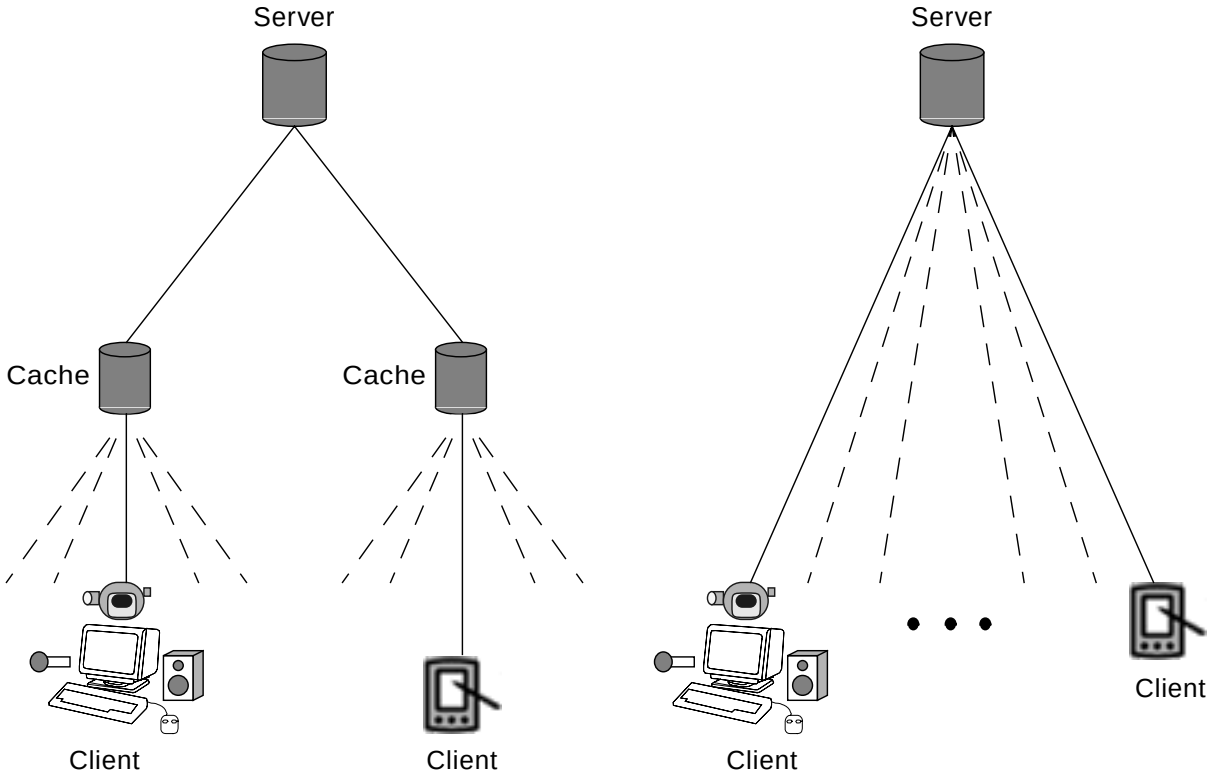


Figure 3.7: Distribution system with and without caches

The work on proxy caching for web objects has received much attention in recent years. This work was mainly concerned with the caching of HTML pages and still images. Further studies have revealed that there is a fundamental difference between the caching of conventional web objects and the caching of multimedia objects like audio and video streams [26]. This is caused by the well known facts that multimedia objects require more storage space, consume more network bandwidth, and have inelastic traffic characteristics. Therefore, a new research area emerged in recent years that is concerned with the caching of multimedia objects in the Internet. The most relevant work in this area is presented in the following.

This subsection reflects the requirements on caching and its related mechanisms in the scope of SAS. First of all, a general overview of the work on partial caching of video objects is given.

Adaptive streaming in combination with write-through caching (see Section 2.4.4) can lead to the situation that video objects are cached only partially. Also several proposals that are focussed on the caching of layer-encoded video are presented. Furthermore, mechanisms that achieve a reliable transport into caches are presented, since SAS requires a reliable transmission protocol (see Section 2.4.4).

Clustering of caches is an approach to increase the scalability of caches and, thus, increase the overall scalabitlity of the video distribution system. The work presented in this thesis does not focus on cache clusters, but existing related work shows how clusters can be used to increase system scalability.

A) Partial Caching of Video Objects

The existing approaches for partial caching can basically be divided in two sub-categories, since there are two dimensions for partial caching. The first dimension is the time and the second is the bandwidth. As shown in Figure 3.8, only a temporal part of the video is stored which is in contrast to a second approach where only a fraction of the bandwidth for the whole length of the video is cached.

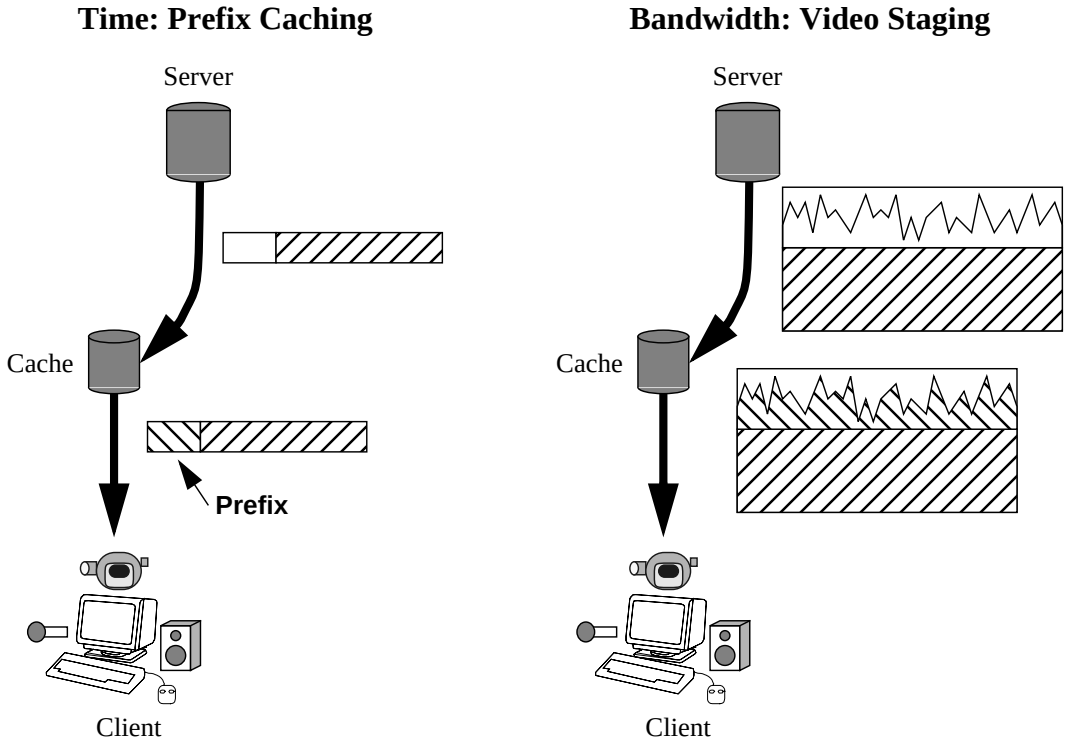


Figure 3.8: Prefix caching and video staging

Time-based Partial Caching.

This section discusses work on time-based partial caching.

Among the first who performed investigations in this area were Tewari et al. [96]. Their *Resource Based Caching* scheme is thought for non-layer-encoded video objects which can either

be cached partially or completely. The motivation to cache fragments of an object is to optimize the cache space usage based on the popularity and the interarrival times of requests for an object. Another scheme for partially caching monolithic video objects is the *prefix* caching scheme presented by Sen et al. [58]. It is the goal of prefix caching to reduce start-up latency and to smooth rate variations on the link between the server and the cache. The first goal is achieved by storing the beginning (prefix) of a video object at the cache. Immediately with the playout of the prefix the remaining part of the video is streamed from the server to the cache. Depending on the length of the cached prefix the suffix can be buffered at the cache and, thus, smoothing can be applied to achieve a constant transmission rate between server and cache. The length of the prefix that is stored on the cache increases with the objects popularity. *Segment-based* [97] caching is an extension of *prefix* caching. In this approach, the video object is split into segments which size increase exponentially with the distance from the start of the video. Thus, it can be assured that the prefixes of many objects are stored on the cache while only the most popular objects are cached entirely. Balafoutis et al. [98] investigate the impact of the replacement granularity on the overall performance of a cache by varying the chunk size, i.e., the smallest unit of a video that can be stored on a cache. Their simulations show that a smaller chunk size leads to a better performance of the cache. Chang and Tobagi [99] present an extension of prefix caching that is designed for a hierarchical distribution infrastructure. Their investigation assumes that part of the length of each video is temporarily stored in each cache of a hierarchical distribution system. In their *pre-storing* scheme, a request for video m_i that arrives at a cache at level one is answered by first sending the prefix of length v_i that is stored locally. The level one cache issues the request for the remaining portion of the movie to the level two cache, forwards this data to the client after playing out its own prefix, and caches the portion from the level two cache temporarily in a sliding window from which all other clients are served that have requested the same video. While this approach is theoretically highly efficient, it requires that several caches cooperate for a single streaming session.

While the *prefix* schemes and its variations presented above always store contiguous parts of the video on the cache, Miao et al. [100] present a method called *selective* caching that stores arbitrary segments in addition to the prefix on the cache. In the case of pre-encoded video, choosing the right packets can increase the robustness of the entire video stream against congestion on the link from the server.

A further extension of *prefix* caching is *MCache* [101] an approach that combines multicast transmission and *prefix* caching. It can also be seen as a modification of the *Patching* technique [39] where the patch is streamed from the prefix that is stored at the cache instead of the original version where both patch and multicast stream are sent from the server. An extension of *MCache* is presented by Wang et al. [102] who modify the caching mechanisms in a way that allows *MCache* to work also in backbones that do not support multicast transmissions. Request merging by a window-based caching approach is proposed by [103] and [99]. That is, requests for the same stream arriving relatively close to each other are merged by caching a sliding window of data belonging to a video on the cache. Another window-based approach is the one presented by Rexford et al. [104]. In their case, the cache is used as a buffer (smoothing window) to allow

smoothed transmission of VBR video in the access network between cache and client. Two further approaches that combine multicast transmission with caching are the schemes presented by Verscheure et al. [105] and *Gleaning* [4]. The goal of these two approaches is to reduce the number of streams that are sent to the client in parallel and to reduce the buffer requirements at the client. Thus, the buffering and sequentialization of the different streams (patch and multicast stream) are performed at the cache and only one stream is forwarded to the client. This is very efficient in environments where access bandwidth is scarce and the client buffer is very limited as it would be in the case of a wireless scenario. An analytical model to investigate optimal caching strategies for video objects in combination with segmented multicast delivery is presented by Eager et al. [106]. Their focus is on how the heterogeneity of client populations and cache capabilities influence the overall system performance. Results of this analysis show that it is almost always beneficial to store initial segments of many files than to store all segments of fewer files. This is caused by the applied multicast delivery technique where initial segments are repeatedly transmitted very frequently, and caching those segments reduces the load at the server. Somewhat surprisingly, even in the case of systems with heterogeneous features, it is efficient to store the same data set at all caches. A refinement of this work is presented in [107]. In this case, a new, more efficient multicast delivery protocol [40] is used in combination with several delivery scenarios. The overall results of this analysis state, in contrast to the previous work, that it is efficient to cache complete objects instead of segments. In cases of high object request rate multicast delivery without cache is more cost efficient.

It must be mentioned that all caching approaches that make use of some sort of multicast delivery mechanisms are not designed for the use of streaming and distributing content scalable video formats like layer-encoded video due to the lack of investigations of how to apply those schemes to content scalable video.

Bandwidth-based Partial Caching.

Another approach of partial caching is the one proposed by Zhang et al. [108] with their *video staging* system. In their case, a monolithic VBR video is split into two parts. As shown in Figure 3.8, the part of the video stream that exceeds a certain threshold rate is cached at the proxy while the lower rate part is stored at the server. In the case of a client request the lower rate part is streamed from the server and the higher rate part from the cache thus, the backbone bandwidth requirement is reduced and the transmission rate is constant. In comparison with the approaches presented above which can be seen as temporal partial caching, *staging* can be seen as an approach for the bandwidth dimension in partial caching. The staging approach is somewhat limited in terms of granularity. If the bandwidth on the link between server and cache should not be sufficient to transmit the lower rate part of the video the stream cannot be offered to the client.

It is quite obvious that layer-encoded video is well suited for this approach of partial caching. Nevertheless, the work on the caching of layer-encoded video has only received little attention compared to the work on time-based partial caching. Rejaie et al. [109] were the first who presented an approach for the caching of layer-encoded video. The video is streamed in a congestion controlled manner (using the RAP protocol [47]) from the server through the cache into the client.

Missing segments on the cache, caused by losses and rate adaptation, are prefetched in a demand driven fashion to improve the quality of the cached video. A cache replacement algorithm is presented that works on a fine-grained level which allows the dropping of single segments of a layer. Simulations reveal that the quality of a cached video is directly related to its popularity. Similar investigations have been performed by Paknikar et al. [110] with the difference that only complete layers can be dropped. In addition, their approach consists of a cluster of caches which is managed by a broker and is thought for a high-speed local area network.

An analytical investigation in this topic was performed by Kangasharju et al. [111]. Their main goal was to gain better insights on the effects cache space and link bandwidth have on the cache performance. In contrast to [109], only complete layers can be stored or removed from the cache in order to keep the problem mathematically tractable. Congestion control on the link between the server and the cache and the cache and the client is not assumed. For the streaming from the server a certain rate is allocated and the transmission of the requested stream starts if this rate is available and it is assumed that this rate is available for the remainder of the streaming session. Thus, this model can only be applied in an environment that gives bandwidth guarantees. In order to make it applicable for the best-effort service the model of the bottleneck bandwidth has to be modified.

A prototype implementation of an adaptive multimedia cache is presented in [112]. They modify the *Squid* web proxy cache [113] in order to perform this kind of caching and perform initial experiments on their prototype. These preliminary results tend to confirm the simulative results from [109]. Further experiments are necessary to examine performance and behavior of this prototype cache in more detail.

It must also be mentioned that bandwidth-based partial caching is not possible with available commercial clients.

The disadvantage of not caching video objects entirely is the reduced fault tolerance. The failure of the server or the link from the server to the client or the cache would lead to a complete failure of the streaming service. This is in contrast to approaches that store a complete video object with the drawback that more storage space is needed at the caches. There clearly exists a trade-off between fault tolerance on the one hand and the efficiency of a cache (in terms of storage space consumption) on the other. An additional drawback of partial caching in the temporal dimension is the limited support for interactivity like VCR functionality. For example, in the case of *prefix* caching a jump to a position further ahead in the video leads to increased latency, since the entire data has to be streamed from the server.

Approaches that make use of layer-encoded video can be seen as an exception if the single layers are always cached completely. In this case, a server or link failure leads not necessarily to a service failure but can lead to a quality decrease of the service since not all layers might be stored on the cache.

It should be mentioned that all of the presented approaches make assumptions on the reliability of the link between the server and the cache. The most popular prefixes used in *prefix* caching and

its variations, e.g., can be actively pushed from the server to the cache via a reliable transport protocol like TCP. The cache can also obtain the prefix by caching the first segments of a stream that is initially requested by a client if the stream is forwarded through the cache. In the latter case the transport would be unreliable and it cannot be assured that the prefix that is identical to the one stored on the server is cached. For reasons of simplicity, all approaches assume that losses between the server and the cache do not occur and, therefore, do not provide any loss recovery mechanisms. In today's best-effort Internet loss recovery mechanisms have to be applied in order to make the proposed approaches realistic. In the following section mechanisms to achieve reliable transport of video data into caches are presented.

B) Reliable Transport into Caches

One of the major goals in an environment for AV-caching should be to obtain a cached version of the content in the cache that is similar to the original content to avoid error propagation towards the client. With the use of standard RTP based on UDP, information that gets lost during transmission is also lost to the caches. The problem is that these errors would be transmitted with every stream that is forwarded from the cache server to a client. In any case that should be avoided since it has to be regarded as a degradation of the service quality.

The work on reliable multicast led to the development of a series of reliable mutlicast protocols on top of UDP. Using reliable multicast in a video distribution infrastructure bears the advantage that simultaneous transport of video objects into caches can be executed more efficiently than in the case of unicast. Some examples for reliable multicast protocols are SRM (Scalable Reliable Multicast, [114]), TRM (Transport Protocol for Reliable Multicast, [115]), RMTP (Reliable Multicast Transport Protocol, [116]) and LRMP (Light-weight Reliable Multicast Protocol as an Extension to RTP, [117]). TRM and LRMP make similar assumptions about loss detection and repair requests as SRM, so SRM can be discussed as an example for all three protocols. RMTP provides sequenced lossless delivery of bulk data (e.g. Multicast FTP), without regard to any real-time delivery restrictions. It is not applicable for streaming applications, because the retransmission of the missing data is done immediately after the loss detection.

SRM is a reliable multicast framework for light-weight sessions and application level framing. Its main objective is to create a reliable multicast framework for various applications with similar requirements on the underlying protocol. Each member of a multicast group is responsible for loss detection and repair requests. The repair requests are multicast after waiting a random amount of time, in order to suppress requests from other members sharing that loss. As it is possible that the last packet of a session is dropped, every member multicasts a periodic, low rate, session message including the highest sequence number. It must be mentioned that SRM needs a specific distribution infrastructure which is not widely available in the Internet at the moment.

A third class of reliable multicast protocols are the ones which include FEC (forward error correction) as a technique to achieve reliability [118]. Reliable multicast achieved through FEC is also applicable for streaming systems, since usually no retransmissions are necessary during the multicast transmission. The major drawback of this approach is that error correction information appropriate for the client with the worst connection must be included in each multicast packet.

This leads to a higher use of bandwidth thus leading to a reduced connection quality for the clients. In addition, a completely new protocol must be built in the case of layered FEC, since this model is not compatible with already existing protocols.

All of the aforementioned approaches either need a specific infrastructure and/or additional functionality in the clients and are not designed for distribution systems that encounter caches as part of the infrastructure.

SR-RTP [119] is a somewhat different loss recovery approach designed for unicast delivery of MPEG-4 video. Retransmissions of lost data are performed based on a certain priority. The priority depends on the content of the lost packet. If the missing data belongs to a reference frame (I-frame) it is requested with a higher priority than data that belongs to dependent frames (P- and B-frames). Based on the priority it might occur that some of the missing packets are never requested for retransmission. Thus, SR-RTP is well suited to improve the quality of a streamed video if only client and server are involved but cannot be used in combination with caches. With SR-RTP it is not guaranteed that an identical copy of the original video object is created on the cache.

An additional approach to achieve reliable transport of video data into caches is LC-RTP [120]. It is not only applicable to multicast transmission but can also be used in the case of unicast. On of the major design goals for LC-RTP was the realization of lossless transport of video data into caches while the stream is also concurrently received by the client. Since LC-RTP is a standard compliant extension of RTP, non LC-RTP capable client can also receive an LC-RTP stream. The SAS architecture builds on LC-RTP to achieve a reliable transport of video data into caches. A detailed description of LC-RTP is given in Appendix A and an extension to support layer-encoded video is presented in Section 8.4.1.

C) Cache Clusters

The goal of clustering (an example for a cache cluster architecture is shown in Figure 3.9) is to distribute the load introduced by client requests on to several single servers. The clusters are usually represented as one single entity. Thus, the client is not aware of the cluster. Next to the advantage of load balancing, video server clusters have the advantage of an increased fault tolerance. If one of the servers crashes, existing sessions can be redirected to other servers and, thus, allow a continuation of the streaming process. To allow load balancing or an increased fault tolerance either a shared storage for the servers of a cluster must exist or data must be replicated between the single servers.

Cache clusters do not necessarily offer the functionality mentioned above. They are mainly used for cooperative caching. Thus, data between caches is not necessarily replicated or they do not share the storage medium. Still, a certain level of load-balancing can be achieved depending on how single video objects are distributed for storage on the single caches.

In Section 3.3.4, an approach for cooperative caching was specified for layer-encoded video. Another approach that allows the storage of segments (belongs to the group of caching methods presented in Section 3.3.4) but not layer-encoded video is presented in the *MiddleMan* architecture [121]. As in [110] all cooperative proxies must be located in a single, well-provisioned LAN and a central *coordinator* is needed to determine where to cache a segment or from which of the

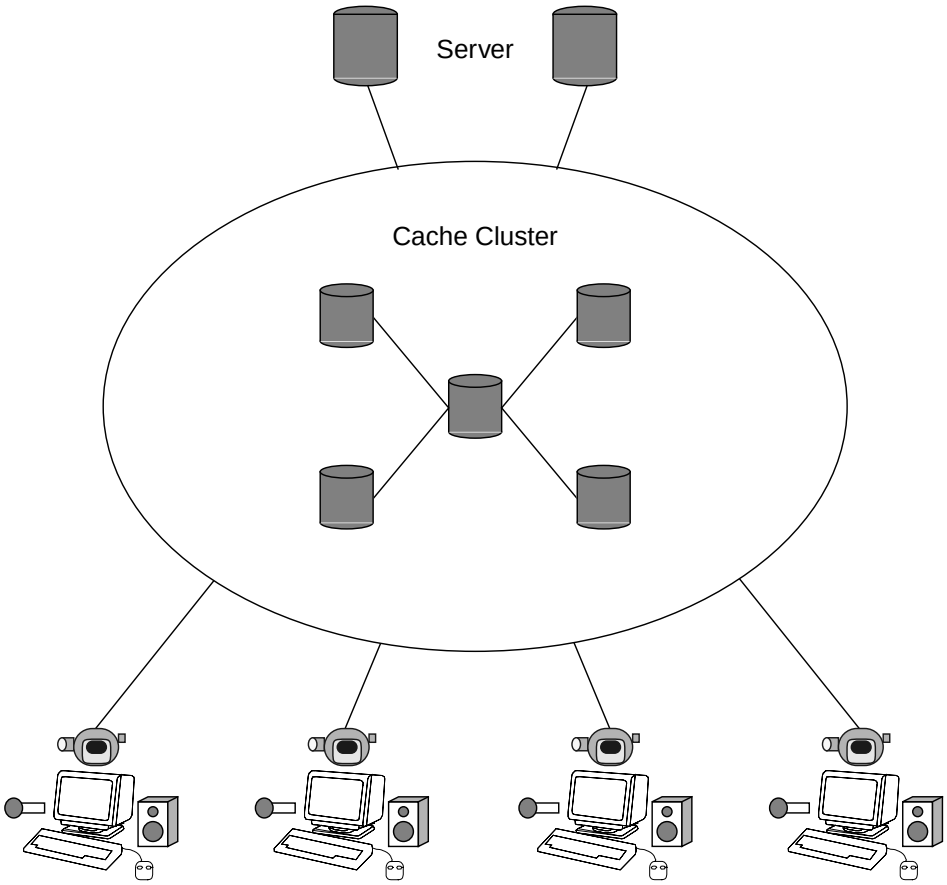


Figure 3.9: Cache cluster architecture

local caches to fetch a single segment from. A simulative analysis of the *MiddleMan* approach is performed that shows the applicability and benefits of such a cooperative caching method for video objects.

A distributed caching architecture that does not make use of a central coordinator or broker is the one presented by Chae et al. [122]. This approach is mainly thought for LANs and MANs since segments of a video object can be stored in a distributed fashion over several caches of the cluster. The segmentation of the video object and the distribution of the segments are based on their *Silo* algorithm. To minimize start-up latency the first segment is stored on all caches of the cluster while the probability to store further segments on a cache decreases. Thus, the higher the segment number the less the number of copies of the segment that exists in the cluster. It is guaranteed that at least one copy of each segment exists in the cluster. The minimized start-up latency is in trade-off with an increased storage requirement, e.g., in comparison to [121]. *Rainbow* and *Caching Token* are the cache replacement strategies that are used in the cache cluster. Analytical and simulation results show that the cache hit rate can be increased by a factor of up to eight in comparison to traditional web caching systems.

Finally, the approach presented by Hofmann et al. [103] is not limited to the usage in LANs and MANs but allows also caches to be distributed in the Internet. Also here the video object is divided into smaller segments and employs a window-based caching approach (see Section 3.3.4)

to merge requests. In addition, requests for a certain object can be redirected to other caches. Scalable state distribution between the caches is achieved by the *Expanding Ring Advertisement* (ERA), a multicast approach in which advertisements are sent with dynamic TTL values, achieving that caches with a further distance from the sender receive updates less frequently.

Castro et. al [123] combine a peer-to-peer overlay network with an application level multicast system, called *SplitStream*, which can be used for video distribution. In their approach a video is split into stripes which are distributed via separate multicast trees. The goal of SplitStream is to create a forest of separate multicast trees in such a way that a node is only an internal node for only one mutlicast tree and a leaf node in all other cases. This mechanism distributes the load equally over all nodes of the distribution system. This approach is well suited for the distribution of layer-encoded video, since each layer can be distributed via a separate multicast tree. Preliminary results of a performance analysis are promising, but further investigations are necessary to show the applicability of this approach for video distribution.

Chapter 4: Quality Variations in Layer-Encoded Video

In the area of video streaming layer-encoded video is an elegant way to overcome the inelastic characteristics of traditional video encoding formats like MPEG-1 or H.261. Layer-encoded video is particularly useful in today's Internet where a lack of Quality of Service (QoS) mechanisms might make an adaptation to existing network conditions necessary. In addition, it bears the capability to support a large variety of clients while only a single file[1] has to be stored at a video server for each video object. The drawback of adaptive transmissions is the introduction of variations in the number of transmitted layers during a streaming session. These variations affect the end-user's perceived quality and thus the acceptance of a service that is based on such technology.

Recent work that has focused on reducing those layer variations, either by employing intelligent buffering techniques at the client [91, 93, 90] or proxy caches [109, 112] in the distribution network, made various assumptions about the perceived quality of videos with time-varying number of layers. An extensive literature research was performed to investigate if these assumptions had been verified by subjective assessment. Yet, existing work that exactly meets the above mentioned conditions could not be found.

Based on this lack of in-depth analysis about quality metrics for variations in layer-encoded videos the decision was made to conduct an empirical experiment based on subjective assessment to obtain results that can be used in classifying the perceived quality of such videos.

4.1 What is the Relation between Objective and Subjective Quality?

The goal of the work presented in this chapter is to investigate if general assumptions made about the quality metrics of variations in layer-encoded videos can be verified by subjective assessment. The following example is used to explain the intention of this investigation in more detail: A layer-encoded video that is transmitted adaptively[2] to the client might have layer variations as shown in Figure 4.1. Several quality metrics that allow the determination of the video's quality are presented in Section 4.2.1. At first, the basics of these quality metrics are discussed. The most straightforward quality metric would be the total sum of all received segments (see Figure 4.1). However, common assumptions on the quality of a layer-encoded video are that the quality is not only influenced by the total sum of received segments but also by the frequency of layer variations

[1] In contrast to the dynamic stream switching [125] approach where for each quality level one specific video file is required.

[2] Adaptively in this case means that the number of layers transmitted to the client is based on some feedback from the network or the client, e.g., congestion control information.

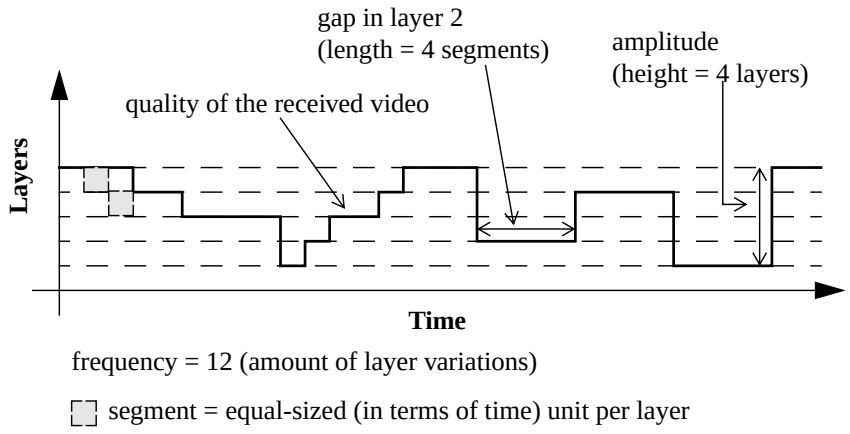


Figure 4.1: Quality of a layer-encoded video at the client

and the amplitude of those variations [91, 109, 126]. As shown in Figure 4.1, the amplitude specifies the height of a layer variation while the frequency determines the number of layer variations.

All known quality metrics are based on these assumptions. Verifying all possible scenarios that are covered by those assumptions with an experiment based on subjective assessment is hard to achieve. Therefore, the focus in this work was set on basic scenarios that have the potential to answer the most fundamental questions, e.g., are the sequences on the left in Figure 4.2 ((a1) and (b1)) more annoying than sequences on the right ((a2) and (b2)) for an end-user who views a corresponding video sequence. In this example, the first scenario ((a1) and (a2)) is focussed on the influence of the amplitude and the second ((b1) and (b2)) on the frequency of layer variations.

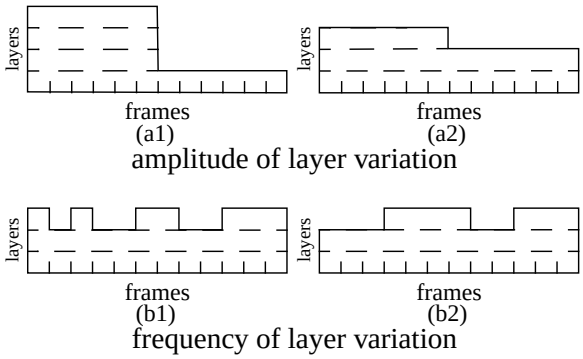


Figure 4.2: Quality criteria [91]

This chapter is structured as follows. Section 4.2 reviews previous work on retransmission scheduling for layer-encoded video and subjective assessment of video quality. The test environment and the subjective test method used for the experiment are described and discussed in Section 4.3. The details of the experimental setup are given in Section 4.4 and in Section 4.5 the results of the experiment are presented and discussed. Section 4.7 summarizes the major conclusions that can be drawn from the experiment. A new objective quality metric, called the spectrum, is presented in Section 4.6. It is shown that this new quality metric is more appropriate than the Peak Signal to Noise Ratio (PSNR) metric.

4.2 Quality Metrics for Video

Given that this work is influenced by the two research areas, quality metrics for layer-encoded video and objective video quality assessment, they are considered separately.

4.2.1 Existing Work on Quality Metrics for Layer-Encoded Video

During the investigation of favorable retransmission scheduling algorithms designed to improve the quality of layer-encoded video stored in a cache, it became clear that in related work specific to this aspect the quality metrics for layer-encoded videos are based on somewhat speculative assumptions only, and none of these assumptions are based on a subjective assessment.

Nelakuditi et al. [91] state that a good metric should capture the amount of detail per frame as well as its uniformity across frames; i.e., comparing the sequences of layers in a video shown in Figure 4.2 the quality of (a2) would be better than that of (a1) which is also valid for (b2) and (b1), according to their assumption. Their quality metric is based on the principle of giving a higher weight to lower layers and to longer runs of continuous frames in a layer.

The metric presented by Rejaie et al. [109] is almost identical to the one advocated in [91]. *Completeness* and *continuity* are the two parameters that are incorporated in this quality metric. *Completeness* of a layer is defined as the ratio of the layer size transmitted to its original (complete) size; e.g., the ratio of layer 2 in sequence (a2) in Figure 4.2 would be 1 while the ratio for layer 3 would be 0.5. *Continuity* is the metric that covers the gaps in a layer. It is defined as the average number of segments between two consecutive layer breaks (i.e., gaps). In contrast to the other metrics presented here, this metric is a per-layer metric.

4.2.2 Objective Video Quality Assessment

There has been a substantial amount of research on methods for subjective assessment of video quality, e.g., [127] and [128], which contributed to form an ITU Recommendation [129]. This standard has been used as a basis for subjective assessment of encoders for digital video formats, in particular for MPEG-2 [130, 128] and MPEG-4 [131] but also on other standards like H.263+ [132]. The focus of interest for all these subjective assessment experiments was the quality of different coding and compression mechanisms. The work presented here, in contrast, is concerned with the quality degradation caused by variations in layer-encoded video. The work presented in [133] is also concerned with layer-encoded video and presents the results of an empirical evaluation of four hierarchical video encoding schemes. The focus of their investigation is on the comparison between the different layered coding schemes and not on the human perception of layer variations.

In [134], a subjective quality assessment was carried out in which the influence of the frame rate on the perceived quality is investigated. Elasticity in the stream was achieved by frame rate variation and not by the application of a layer-encoded video format.

The effects of bit errors on the quality of MPEG-4 video were explored in [135] by subjective viewing measurements, but effects caused by layer variations were not examined.

Chen presents an investigation, which is based on a subjective assessment, on an IP-based video conference system [136]. The focus in this work is mainly auditorium parameters like display size and viewing angle. A layer-encoded video format is not used in this investigation.

Lavington et al. [137] used an H.263+ two layer video format in their trial. This is probably closest to the work presented here, although they were rather interested in the quality assessment of longer sequences (e.g., 25 min.). As opposed to using identical pregenerated sequences that were presented to the test candidates, videos were streamed via an IP network to the clients and the quality was influenced in a fairly uncontrolled way by competing data originating from a traffic generator. The very specific goal was to examine if reserving some bandwidth for either the base or the enhancement layer improves the perceived quality of the video, while the investigation presented here is rather concerned with the influence of variations in layer-encoded videos and tries to verify some of the basic assumption made about the perceived quality in a subjective assessment experiment. Furthermore, the experiment as described in the following is conducted in a controlled environment in order to achieve statistically significant results.

4.3 Test Environment

In this section, at first the layer-encoded video format used for the experiment is presented and afterwards the generation of the test sequences is described, the decision to use stimulus-comparison as the assessment method is motivated, and finally the test application is presented.

4.3.1 Layer-Encoded Video Format - SPEG

SPEG (Scalable MPEG) [80] is a simple modification to MPEG-1 which introduces scalability in the transmission rate of a video stream. In addition to the possibility of dropping complete frames (temporal scalability), which is already supported by MPEG-1 video, SNR scalability is introduced through layered quantization of the Discrete Cosine Transform (DCT) data [80]. The extension to MPEG-1 was made for two reasons. First, there are no freely available implementations of layered extensions for existing video standards (MPEG-2, MPEG-4), second, the granularity of scalability is improved by SPEG combining temporal and SNR scalability. As shown in Figure 4.3, a priority (p_0(highest) - p_{11}(lowest)) can be mapped to each layer. The QoS Mapper (see Figure 4.4, which depicts the SPEG pipeline and its components) uses the priority information to determine which layers are dropped and which are forwarded to the Net Streamer.

	I	B	P
Level 0	p_0	p_1	p_2
Level 1	p_3	p_4	p_5
Level 2	p_6	p_7	p_8
Level 3	p_9	p_{10}	p_{11}

Figure 4.3: SPEG layer model

The decision to use the SPEG as a layer-encoded format for the subjective assessment is based on the following reasons. SPEG is designed for a QoS-adaptive video-on-demand (VoD)

approach, i.e., the data rate streamed to the client should be controlled by feedback from the network (e.g., congestion control information). In addition, the developers of SPEG also implemented a join function that re-converts SPEG into MPEG-1 [124] allowing the use of standard MPEG-1 players, e.g., the Windows Media Player. Scalable video encoders available as products (e.g., [138, 139]) were not an option because videos created by those can only be streamed to the corresponding clients which do neither allow the storage of the received data on a disk nor the creation of scheduled quality variations.

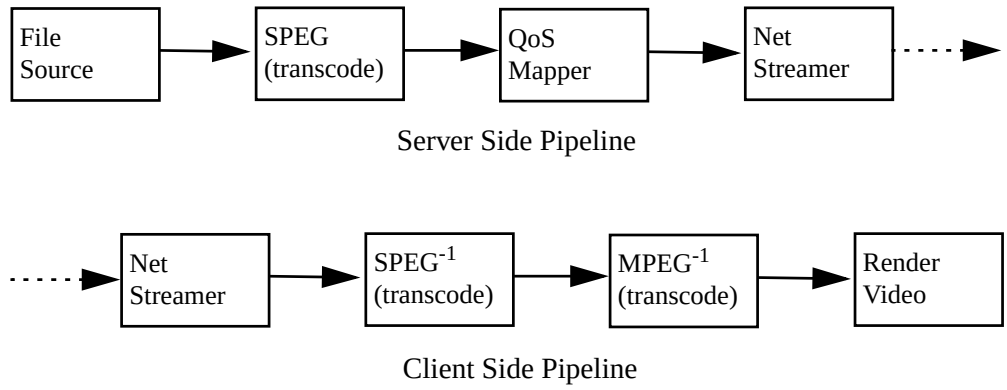


Figure 4.4: Pipeline for SPEG [124]

Figure 4.5 shows one single frame of an SPEG encoded sequence in its four possible steps of quality.

4.3.2 Test Generation - Full Control

Since the test sequences must be created in a deterministic manner, the SPEG pipeline was slightly modified. The most important difference is, that for the subjective assessment data belonging to a certain layer must be dropped intentionally and not by an unpredictable feedback from the network or the client. This modification is necessary, since identical sequences must be presented to the test candidates in the kind of subjective assessment method that is used in the experiment. Therefore, the QoS Mapper is modified in a way that layers are dropped at certain points in time specified by manually created input data. Additionally a second output path to the MPEG^{-1} module is added allowing to write the resulting MPEG-1 data in a file and the NetStreamer modules are completely eliminated.

4.3.3 Measurement Method - Stimulus Comparison

The subjective assessment method is widely accepted for determining perceived image and video quality. Research that was performed under the ITU-R led to the development of a standard for such test methods [129]. The standard defines basically five different test methods which are briefly explained in the following. Figure 4.6 (on page 53) gives an overview of the classification of the different test methods. A detailed description of the single test methods can be found in [140].

Base Layer | Base Layer + Layer 1

Base Layer + Layer 1+ Layer 2 | Base Layer + Layer 1+ Layer 2 + Layer 3

Figure 4.5: One SPEG frame in four different quality levels (sequence Table Tennis)

- **DSCQE:**
  Double-stimulus continuous quality evaluation (DSCQE) is qualified for the assessment of new codecs or streaming video. The test is executed as follows: The original and the impaired video sequence are shown to the client in nondescript order. After the second sequence the test candidate assesses the quality of both sequences. Meanwhile, both sequences will be presented again and afterward the test candidate has to make the final assessment.
- **DSIS and DSBV:**
  Two additional test methods which make use of reference sequences are double-stimulus impairment scale (DSIS) and double-stimulus binary vote (DSBV). The main difference between both test methods is the comparison scale. While for DSIS a scale of five discrete values is used, the scale for DSBV is binary and allows only the assessment if the second sequence was equal to or worse than the first sequence. In both tests the reference sequence is always presented first.
- **SSCQE:**
  Single-stimulus continuous quality evaluation (SSCQE) is the first of two tests that belong to the category of test without a reference sequence. A sequence that can be of

up to 30 minutes in duration is presented to the client. The quality of the sequence changes over time. The test candidate uses a control unit to assess the perceived quality of the sequence concurrently. The control unit is sampled every 2 to 5 seconds resulting in a graph that represents the video quality over time.

- **SC:**
  Stimulus comparison (SC) is another single-stimulus test method. With SC two impaired sequences are directly compared with each other. The test candidates do not assess the quality of a single sequence but the quality difference in between the sequences. The assessment is executed after the second sequence is shown to the test candidate on a discrete scale consisting of seven options (see Table 4.1).

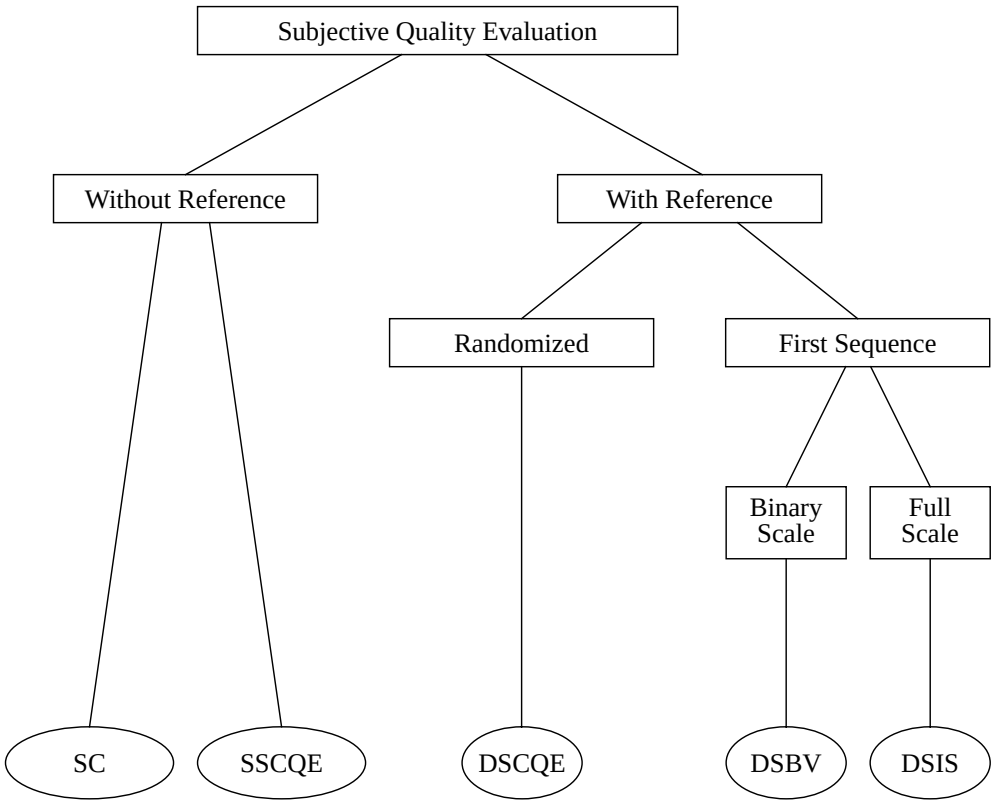


Figure 4.6: Test method classification

Since it is a major goal to investigate the basic assumptions about the quality of layer-encoded video, SSCQE is not the appropriate assessment method because comparisons between two video sequences are only possible on an identical time segment and not between certain intervals of the same video. In addition, SSCQE was designed to assess the quality of an encoder (e.g., MPEG-1) itself.

Two test methods which better suit the kind of investigations performed in this thesis are DSCQS and DSIS. Compared to SSCQE they allow one to asses the quality of a codec in relation

Table 4.1: Comparison scale

Value	Compare
-3	much worse
-2	worse
-1	slightly worse
0	the same
1	slightly better
2	better
3	much better

to data losses [127] and, therefore, are more suitable if the impairment caused by the transmission path is investigated.

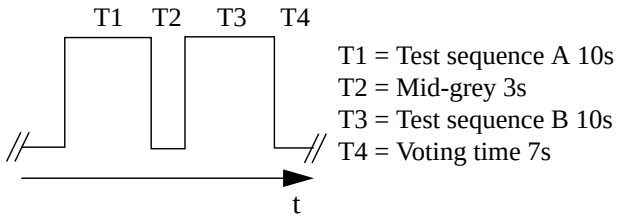


Figure 4.7: Presentation structure of test material

The SC method differs from DSCQS and DSIS in a way that two test sequences with unequal qualities are shown (see Figure 4.7) and the test candidates can vote on a scale as shown in Table 4.1. Comparing two impaired videos directly with each other is the primary goal of this investigation. Since this is represented best by the SC method, the decision was made to use this method in the subjective assessment.

Additionally, preliminary tests (see Appendix B) have shown us that test candidates with experience in watching videos on a computer are less sensitive to impairment. That is, they recognize the impairment but do not judge it as annoying as candidates who are unexperienced. This effect is dampened since only impaired sequences have to be compared with each other in a single test that is based on the SC method. Preliminary tests with the DSIS method, where always the original sequence and an impaired sequence are compared, delivered results with less significance compared to tests performed with the SC method.

4.3.4 Test Application - Enforcing Time Constraints

Automated execution of the assessment was realized by an application [141] that was developed for this specific purpose. Since a computer is used to present the videos anyway, the candidates perform their voting also on the computer. Using this application has the advantage that time constraints demanded by the measurement method can be easily enforced, because voting is only pos-

sible during a certain time interval (exactly 7 second as shown in Figure 4.7). As a convenient side effect, the voting data is available in a machine readable format, thus simplifying the statistical analysis of the gathered data. Figure 4.8 shows a snapshot of the application used to perform the subjective assessment.

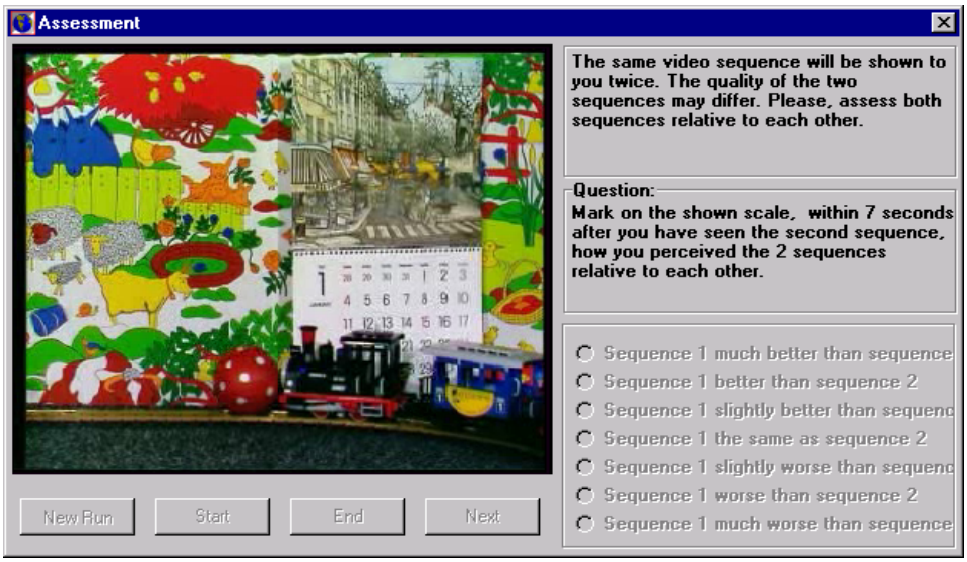


Figure 4.8: Test application

4.4 Experiment

4.4.1 Scenario

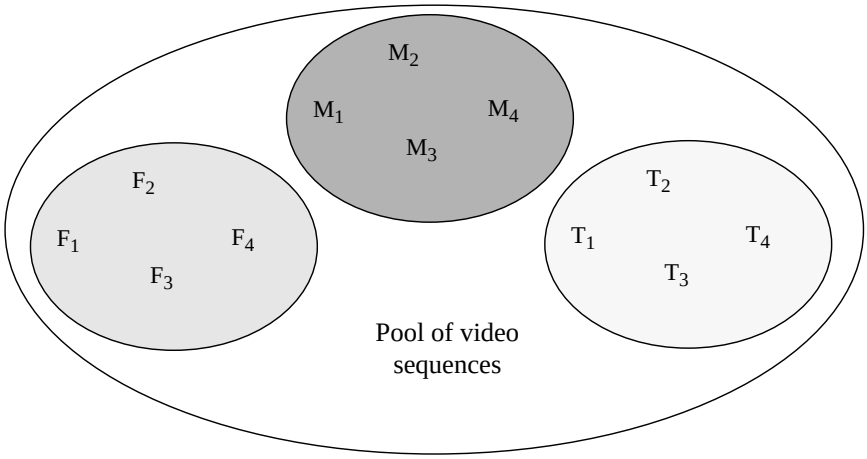
Since quality metrics for layer-encoded video are very general, the focus must be set on some basic test cases in order to keep the number of tests that should be performed in the experiment feasible. Therefore, only isolated effects were investigated, one-by-one at a time, which on one hand keeps the size of a test session reasonable and on the other hand still allows to draw conclusions for the general assumptions, as discussed above. Thus, the investigation rather concentrates on observing the quality ranking for isolated effects like frequency variations (as shown in sequences (b1) and (b2) in Figure 4.2) than on combined effects (as shown in Figure 4.1). This bears also the advantage that standardized test methods [129], which limit the sequence length to several seconds, can be applied. All patterns that were used for the experiment are shown in Figure 4.10.

4.4.2 Candidates

The experiment was performed with 115 test candidates (76 males and 39 females), between the age of 14 and 64. 89 of them had experiences with watching videos on a computer.

4.4.3 Procedure

Each candidate had to perform 15 different assessments and each single test lasted for 33 seconds. All tests were executed according to the SC assessment method. The complete test session per candidate lasted for about 15 minutes[1], on average. Three video sequences for this experiment that have been frequently used for subjective assessment [142] were chosen. The order of the 15 video sequences was changed randomly from candidate to candidate as proposed in the ITU-R B.500-10 standard [129] (see also Figure 4.9). After some initial questions (age, gender, profession) three assessments were executed as a warm-up phase. This should avoid that the test candidates are distracted by the content of the video sequences as reported by Aldridge et al. [130] (see also Appendix B). In order to avoid that two consecutive video sequences (e.g., F_2 is following F_1 immediately) have the same content a pattern for the chronological order of the test sessions, as shown in Figure 4.9, is defined. F_x can be any video sequence from the F pool of sequences that



Pattern	I_1	I_2	I_3	F_x	M_x	T_x	F_x	M_x	T_x	F_x	M_x	T_x	F_x	M_x	T_x
Sequence 1	I_1	I_2	I_3	F_3	M_1	T_4	F_1	M_2	T_1	F_4	M_3	T_2	F_2	M_4	T_3
Sequence 2	I_1	I_2	I_3	F_3	M_1	T_4	F_1	M_2	T_1	F_4	M_3	T_2	F_2	M_4	T_3

F= Farm
M= Mobile & Calendar
T= Table Tennis

Figure 4.9: Random generation of test sequence order

has not been used in this specific test session, so far. Thus, a complete test session for a candidate could have a chronological order as shown in Figure 4.9 (*Sequence1* and *Sequence2*).

[1] Only watching the sequences and voting took less time, but the candidates had as much time as they wanted to read the questions and possible answers for each test ahead of each test.

4.4.4 Layer Patterns

Figure 4.10 shows the layer patterns of each single sequence that was used in the experiment, except for the first three warm-up tests where the comparison is performed between the first sequence that consists of four layers and the second that consists of only one layer. Each of the three groups shows the patterns that were used with one type of content. Comparisons were always performed between patterns that are shown in a row (e.g., (a1) and (a2)). As already mentioned in Section 4.2 the goal of this investigation was to examine fundamental assumptions about the influence of layer changes on perceived quality. This is also reflected by the kind of patterns used in the experiment. It should be mentioned that the single layers are not equal in size (contrary to the presentation in Figure 4.10) as the size of the n^{th} layer is given by the following expression: $s_n = 2s_{n-1}$ and, thus, segments of different layers have different sizes. Preliminary experiments showed that equal layer sizes are not appropriate to make layer changes perceivable; this is regarded as a realistic assumption since layered schemes that produce layers with sizes similar to the ones created by SPEG exist [143, 144].

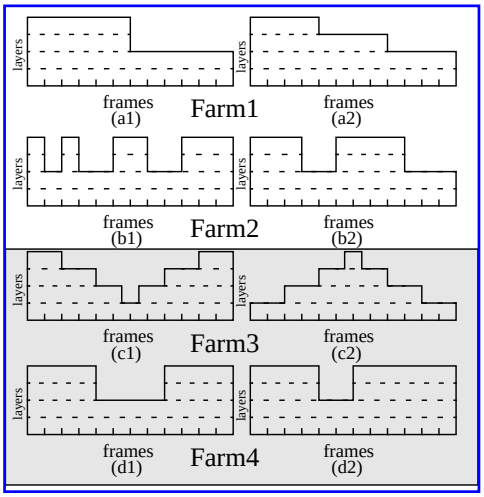
In the experiment, a differentiation between two groups of tests is made, i.e., one group in which the amount of segments used by a pair of sequences is equal and one in which the amount differs (the latter have a shaded background in Figure 4.10).

The main interest of the subjective assessment was in cases where it is necessary to compare the influence of additional segments that are added on different locations in a sequence (as shown in the test *M&C1* in Figure 4.10). Results of those tests could have an implication on how two measure and improve the quality of existing retransmission scheduling techniques (see Section 4.2.1). Yet, some of the effects like the influence of the frequency of layer variations could only be investigated with test sequences consisting of a different number of segments (as shown in the test *T-Tennis2* in *Figure 4.10*).

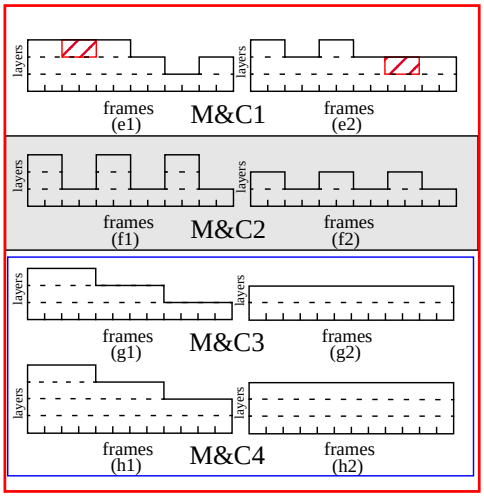
Since segments from different layers are not equal in size, the amount of data for the compared sequences differs. However, somewhat surprisingly and as discussed in Section 4.5.3, a larger amount of data (resulting in a higher PSNR value) does not necessarily lead to a higher perception of quality. Additional tests with different quantities of segments in between a pair were chosen to answer additional questions and make the experiment more consistent as shown in Section 4.5.2.

4.5 Results

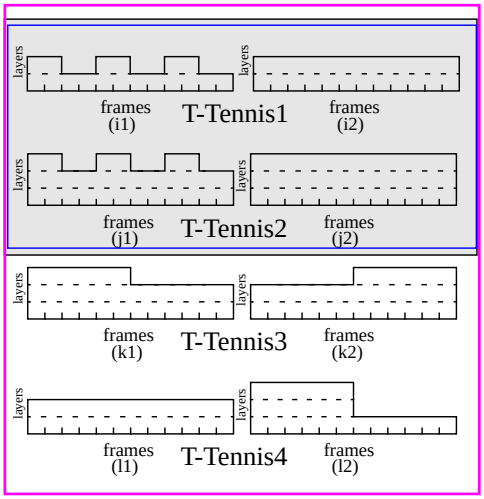
In the following, the results of the experiment described in Section 4.4 are presented. Given the statistical nature of the gathered data it is obvious that the presented results cannot prove an assumption but only make it less or more likely. The overall results of all experiments are summarized in Figure 4.11 and are discussed in the following subsections. Next to the statistical results obtained from the subjective assessment, objective data in terms of the average PSNR per sequence is also provided. The average PSNR was obtained by comparing the original MPEG-1



Patterns for Sequence “Farm” (F1-F4)



Patterns for sequence “Mobile & Calendar” (M1-M4)



Patterns for sequence “Table Tennis” (C1-C4)

Figure 4.10: Segments that were compared in the experiment

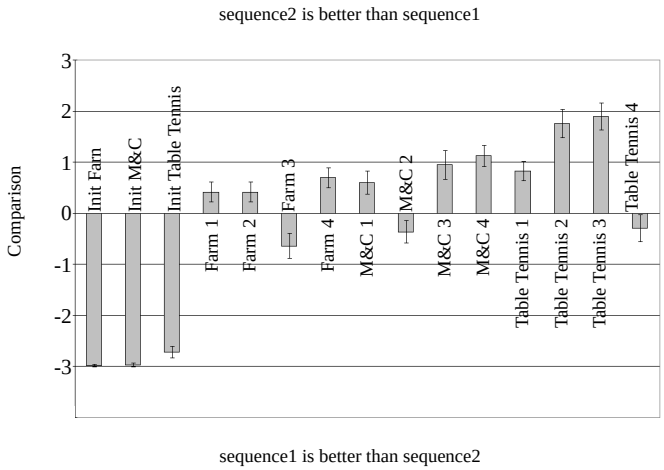


Figure 4.11: Average and 95% confidence interval for the different tests of the experiment

sequence with the impaired sequence on a per frame basis. This results in 250 single PSNR values per sequence which were used to calculate the average PSNR.

4.5.1 Same Amount of Segments

In this section, the results for the assessments of tests in which the total sum of segments is equal are discussed. That means that the space covered by the pattern of both sequences is identical.

A) Farm1: Amplitude

In this assessment the stepwise decrease was rated slightly better than one single but higher decrease. The result tends to justify the assumptions that were made about the amplitude of a layer change (as described in Section 4.2.1).

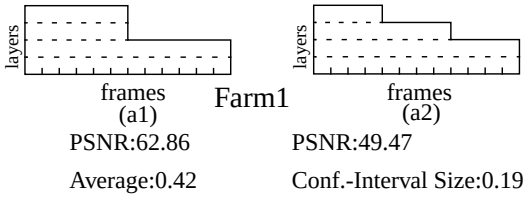
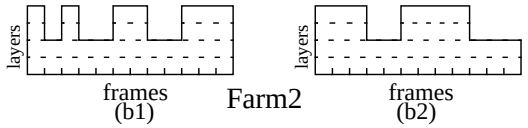


Figure 4.12: Farm1

B) Farm2: Frequency

The result of this test shows an even higher likelihood that the second sequence has a better perceived quality than in the case for *Farm1*. It tends to confirm the assumption that the frequency of



PSNR:61.46 PSNR:73.28

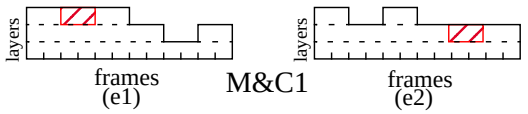
Average:0.42 Conf.-Interval Size:0.19

Figure 4.13: Farm2

layer changes influences the perceived quality, since, on average, test candidates ranked the quality of the sequence with lesser layer changes higher.

C) M&C1: Closing the gap

This test tries to answer the question: would it be better to close a gap in a layer on a higher or lower level? The majority of the test candidates decided that filling the gap on a lower level results in a better quality than otherwise. This result tends to affirm assumptions made for retransmission scheduling.



PSNR:63.15 PSNR:52.38

Average:0.6 Conf.-Interval Size:0.23

Figure 4.14: M&C1

D) M&C3: Constancy

With an even higher significance than in the preceding tests, the candidates considered the sequence with no layer changes as the one with the better quality. One may judge this a trivial and unnecessary test, but the result is not that obvious, since (g1) starts with a higher number of layers. The outcome of this test implies that it might be better, in terms of perceived quality, to transmit less but a constant number of layers.

PSNR:48.01 PSNR:25.08

Average:0.95 Conf.-Interval Size:0.28

Figure 4.15: M&C3

E) M&C4: Constancy at a higher level

This test was to examine if an increase in the overall level (in this case by comparison to M&C3) has an influence on the perceived quality. Comparing the results of both tests (*M&C3* and *M&C4*) shows no significant change in the test candidates' assessment. 66% of the test candidates judge

the second sequences ((g2) and (h2)) of higher quality (values 1-3 in Table 4.1) in both cases which makes it likely that the overall level has no influence on the perceived quality.

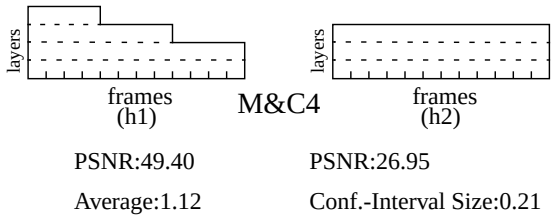


Figure 4.16: M&C4

F) Tennis3: All is well that ends well

The result of this test shows the tendency that increasing the number of layers in the end leads to a higher perceived quality. The result has a remarkably strong statistical significance (the highest bias of all tests).

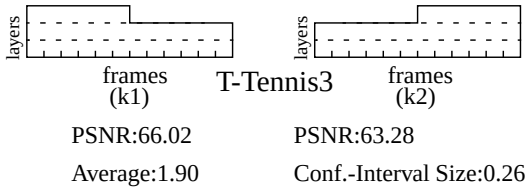


Figure 4.17: Tennis3

G) Tennis4: The exception proves the rule

The result of this test is a little bit surprising since it contradicts the results from Farm1 and M&C3. It can only be assumed that also the content might have an influence on the perceived quality. Yet, to gain more insight in this phenomenon further experiments are necessary.

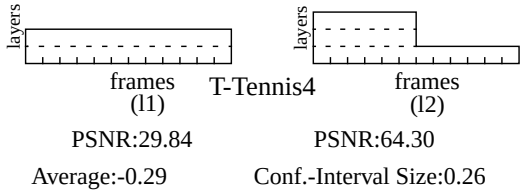


Figure 4.18: Tennis4

4.5.2 Different Amount of Segments

In the following five tests the total amount of segments per sequence differs. All five tests have in common that the perceived quality of the sequence consisting of a pattern that covers a larger number of segments were ranked better. This is obvious, but it makes the overall result more consistent, because test candidates mostly realized this quality difference.

A) Farm3: Decrease vs. increase

Starting with a higher number of layers, decreasing the number of layers, and increasing the number of layers in the end again seems to provide a better perceivable quality than starting with a low

number of layers, increasing this number of layers, and going back to a low number of layers at the end of the sequence. This might be caused by the fact that test candidates are very concentrated in the beginning and the end of the sequence and that, in the first case details become clear right in the beginning of the sequence.

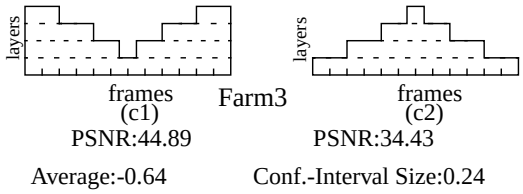


Figure 4.19: Farm3

B) Farm4: Keep the gap small

In this test, the goal was to investigate how the size of a gap may influence the perceived quality. The majority of test candidates (66 out of 115) judged the quality of the sequence with a smaller gap slightly better (Only 6 out of 115 judged the first sequence better). This indicates that filling a gap partly can be beneficial.

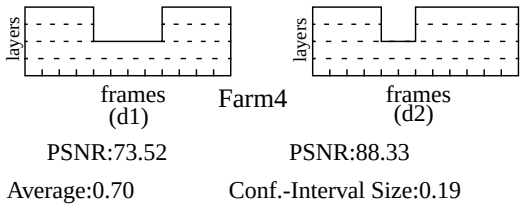


Figure 4.20: Farm4

C) M&C2: Increasing the amplitude

The effect of the amplitude height should be investigated in this test. The result shows that, in contrast to existing assumptions (see Section 4.2.1), an increased amplitude can lead to a better perceived quality.

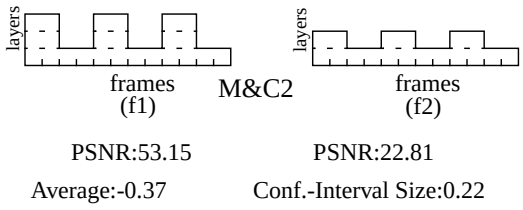


Figure 4.21: M&C2

D) Tennis1: Closing all gaps

This test is contrary to *M&C2* where the additional segments are used to close the existing gaps instead of increasing the amplitude of already better parts of the sequence. This strategy decreases the frequency of layer changes. Test candidates, on average, judged the sequence without layer changes better. The result of this test reaffirms the tendency that was already noticed in M&C1, that the perceived quality is influenced by the frequency of layer changes. The comparison of the results of *M&C2* and *Tennis1* show a tendency towards filling the gaps and, thus, decreasing the

frequency instead of increasing the amount of already increased parts of the sequence is recognizable. Definitely, further investigations are necessary to confirm this tendency, because here the results of tests with different contents are compared and the influence of the content on the perceived quality has not been investigated, so far.

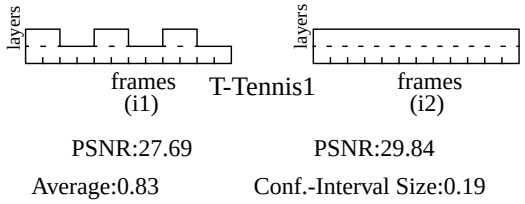


Figure 4.22: Tennis1

E) Tennis2: Closing all gaps at a higher level

In comparison to *Tennis1*, here, the interest is in how an overall increase of the layers (in this case by one layer) would influence the test candidates judgement. Again the sequence with no layer changes is judged better, even with a higher significance than for *Tennis1*. This might be caused by the fact that the number of layer is higher in general in *Tennis2*.

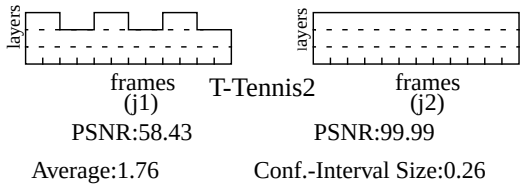


Figure 4.23: Tennis2

4.5.3 Sequence Size and Quality

The PSNR is a popular metric to present the objective quality of video data. Therefore, the average PSNR of each sequence was also computed in order to investigate how subjective and objective quality are related. Since the determination of the objective quality can be performed with much less effort than a subjective assessment, the result of this investigation may provide hints as to whether the determination of the average PSNR is sufficient to define the quality of a video sequence. Note that, since the relation between subjective and objective quality is not the focus of the investigation presented in this section, this can only be seen as a by-product and would certainly need further investigation. (The PSNR values for each sequence are given in Figure 4.12 through Figure 4.23.)

The results of the subjective assessments are contrary to the results of the PSNR in 8 of the 12 test cases. The obtained results for the test in which the sum of segments was equal for each sequence (Section 4.5.1) are even stronger. They do not indicate a positive correlation between both quality metrics (see Table 4.2). From the results of the subjective assessment we see a strong

tendency that, in the case of layer-encoded video, the quality of a sequence is not well represented by the average PSNR.

Table 4.2: Comparison between subjective and objective quality (same amount of segments)

Shape	Farm1	Farm2	M&C1	M&C3	M&C4	T-Tennis3	T-Tennis4
PSNR of shape 1	62.86	61.46	63.15	48.01	49.40	66.02	29.84
PSNR of shape 2	49.47	73.28	52.38	25.08	26.95	63.28	64.30
Average of assessment	0.42	0.42	0.60	0.95	1.12	1.90	-0.29

contrary to subjective assessment in accordance with subjective assessment

4.6 The Spectrum

The investigation presented in Section 4.5.3 reveals that the average PSNR is not well suited to represent the perceived quality of layer-encoded video. This lack of an appropriate objective quality metric for layer-encoded video lead to a new objective metric called the *spectrum.* The goal during the development of the spectrum was to express the factors that influence the perceived quality (as presented in Section 4.5) through a mathematical expression giving similar results as the ones obtained by the subjective assessment.

Therefore, the *spectrum* of a cached layered video v can be introduced:

$$s(v) = \sum_{t=1}^{T} z_t \left(h_t - \frac{1}{\sum_{i=1}^{T} z_i} \left(\sum_{j=1}^{T} z_j h_j \right) \right)^2 \qquad (2)$$

With h_t and z_t defined as:

- h_t - number of layers in time slot t, $t = 1,..., T$
- z_t - indication of a step in time slot t, $z_t \in \{0,1\}$, $t = 1,..., T$

Without loss of generality, a slotted time period with slots corresponding to the transmission time of a single (fixed-size) segment is assumed as well as that all layers are of the same size.

The spectrum captures the frequency as well as the amplitude of quality variations. The amplitude is captured by the differences between quality levels and average quality levels where larger amplitudes are given higher weight due to squaring these differences. The frequency of variations is captured by z_t. Only those differences that correspond to a step in the cached layered video are taken into account. While the spectrum as defined in (2) looks very similar to the usual variance of quality levels for the cached video, it is important to note that the introduction of the z_t takes into account the frequency of changes in the quality levels. A spectrum of the value 0 represents the

best possible quality, while the spectrum increases with a decreasing quality. An example calculation of the spectrum is shown in Figure 4.24.

The decision to name this objective quality metric spectrum is caused by the fact that its is influenced by the amount of layer variations (i.e. the frequency of layer variations). It does not reflect what is meant by the spectrum in a strong mathematical sense (the result of the transformation from the time domain into the frequency domain) and, thus, must not be mistaken with it.

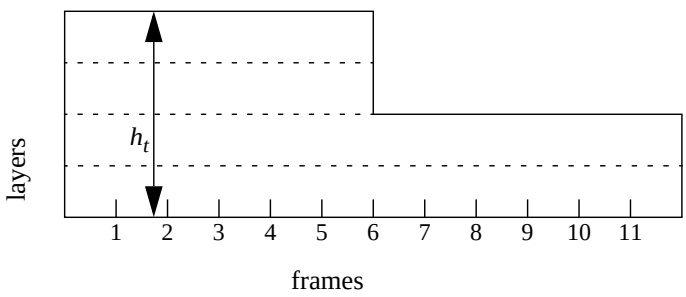


Figure 4.24: Parameters to calculate the spectrum

4.6.1 Comparison of the Spectrum with the Subjective Assessment Results and the PSNR

In Table 4.3, the spectrum of the shapes presented in Section 4.3.1 is compared with the results from the subjective assessment and the PSNR for each sequence. With the exception of the cases for the *Farm1* (Figure 4.12) and *T-Tennis3* (Figure 4.17) tests, there is a consistency between the subjective quality and the spectrum. In contrast to the subjective results, the PSNRs of the sequences is only consistent in one of the six cases (test *Farm2)*. This argues for the hypothesis that the spectrum is more suitable as an objective quality metric than the PSNR because the PSNR of each shape and the results of the subjective assessment are not consistent, as shown in Section 4.5.3. Though, it must be mentioned that the spectrum does not regard time dependencies very well as can be seen in the case of the *T-Tennis3* test (Figure 4.17) where the spectrum is equal for both cases but the second shape was assessed as having a higher quality. The latter is due to the fact that the amount of layers increases towards the end of the sequence. With the knowledge that the spectrum is a suitable metric for the quality of layer-encoded video, it was chosen to rate and compare the retransmission scheduling algorithms that are presented in Chapter 5.

4.7 Summary

A statistical analysis of the experiment mostly validates assumptions that were made in relation to layer variations and the perceived quality of a video:

- The frequency of variations should be kept as small as possible.
- If a variation cannot be avoided the amplitude of the variation should be kept as small as possible.

One basic conclusion from the results in Section 4.5.2 is: adding information to a layered video increases its average quality. Yet, adding information at different locations can have a substantial

Table 4.3: Comparison among spectrum, subjective, and objective quality

Shape	Farm1	Farm2	M&C1	M&C3	M&C4	T-Tennis3
s(v) of shape 1	2	6.86	2	2	2	0.5
s(v) of shape 2	2	4	1	0	0	0.5
PSNR of shape 1	62.86	61.46	63.15	48.01	49.40	66.02
PSNR of shape 2	49.47	73.28	52.38	25.08	26.95	63.28
Average of assessment	0.35	0.55	0.73	1.18	1.02	2.18

contrary to subjective assessment

in accordance with subjective assessment

inconclusive

effect on the perceived quality. Assumptions made for heuristics in retransmission scheduling could be substantiated by this investigation. That means, it is more likely that the perceived quality of a layer-encoded video is improved if

- the lowest quality level is increased, and
- gaps in lower layers are filled.

The results from Section 4.5.3 should be used to refine the retransmission scheduling heuristics in relation to the size of each single layer. Therefore, the metric that represents the quality improvement must also take into account that it might be more expensive to retransmit a segment of layer n+1 than of layer n. Another interesting outcome of the experiment is the fact that a quality improvement may be achieved by retransmitting less data, if a layered encoding scheme is used in which the layers are not of identical size. The obtained results can, in addition, be used to refine caching replacement policies that operate on a layer level [109] as well as layered multicast transmission schemes which try to offer heterogeneous services to different subscribers as, e.g., in the receiver-driven layered multicast RLM [94] scheme and its derivations.

The results of this investigation clearly strengthen the assumption that a differentiation between objective and subjective quality, in the case of variations in layer-encoded video, must be made, if the objective measure is based on the PSNR. A new objective quality measure (the spectrum) that was developed based on the results of the subjective assessment is a more suitable metric for the quality of layer-encoded video compared to the PSNR.

Chapter 5: Retransmission Scheduling

5.1 Motivation

With SAS, it is very likely that videos are not cached in their best quality when they are cached for the first time. However, for subsequent requests which shall be served from the cache it may be unattractive to suffer from the possibly very bad or strongly varying quality experienced by the initial transmission of the video that has been selected for caching. Therefore, missing segments of the cached video should be retransmitted to enable higher quality service from the cache to its clients. The most interesting issue here is how to schedule the retransmissions, i.e., in which order to retransmit missing segments, in order to achieve certain quality goals for the cached video content. A further design issue is, when to schedule retransmissions.

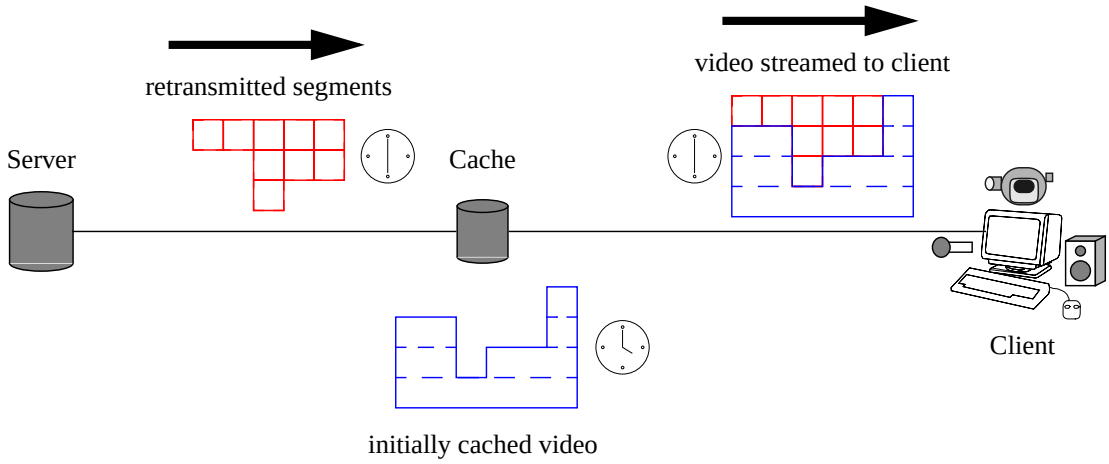


Figure 5.1: Retransmission scheduling

The method of retransmission scheduling can also be used in clients, if they allow buffering and the buffer can store at least as much data as can be sent during an RTT [91]. However, the focus in this thesis is on retransmission into caches, since these retransmissions can be beneficial for more than one client and caches are a necessary element of *Scalable Adaptive Streaming* in any case. This mechanism is called retransmission due to the fact that an entire layer was not transmitted for a certain interval due to congestion on an intermediate link, but would have been transmitted if no congestion had occurred. Figure 5.1 gives a basic overview of the retransmission process. A video object is initially cached as shown by the blue shape. At a later point in time this object is requested by a client which is also served by the cache. The cache now requests missing segments and forwards them in combination with the segments of the cached video object to the client.

Thus, the quality of the video object streamed to the client has a higher quality than the initially cached object. Depending on the decision of the cache, the retransmitted segments might also be stored on the cache in order to increase the quality of the cached object. When and how these retransmissions are performed is discussed in more detail in Section 5.1.1.

In this chapter, an investigation on retransmission scheduling is described. The goal of this investigation is to develop mechanisms for retransmission scheduling to improve the quality of layer-encoded video at client and cache. Since an investigation on optimal retransmission scheduling showed that it cannot be applied in SAS due to the fact that it is computationally infeasible (see Section 5.2), heuristics are developed which have a much lower complexity. The creation of new heuristics is also caused by the fact that an existing heuristic for retransmission scheduling revealed some drawbacks, as is shown in Section 5.4. Knowledge gained by the subjective assessment (see Chapter 4) is used for the creation of the new retransmission scheduling heuristics. A performance analysis of these heuristics is carried out through simulations (see Section 5.5). Next to the retransmission scheduling heuristics four different approaches that focus on the location of the maximum quality improvement are presented. During the investigation on retransmission scheduling it became obvious that retransmissions can be performed to maximize the quality of a layer-encoded video at either the cache (see Section 5.6) or the client (see Section 5.4). Thus simulations for the two approaches are performed which lead to a third hybrid approach which maximizes the quality for the current viewer while increasing the quality on the cache almost to the maximum (see Section 5.7).

5.1.1 Retransmission Time

There are four possible occasions when to perform retransmissions. All four have the goal to improve the quality of the cached content as soon as possible after the decision was made to start the retransmission:

1. Directly after the initial streaming process: the cache starts requesting missing segments without waiting for further requests for a certain video. This allows one to offer the highest possible quality to requests that arrive during the retransmission phase.
2. During the initial streaming process: the cache starts requesting retransmissions after a certain amount of time after the start of the streaming session. Data that is retransmitted does not improve the quality for the actual viewer, but increases the quality of the cached video object and, thus, the quality for viewers that request the object at a later point in time.
3. During subsequent requests: the cache serves subsequent requests but, simultaneously, also orders missing segments from the server.
4. During requests for different content from the server: in [86] and Chapter 7, a technique is presented that allows the transmission of requested segments (for an already cached video) in addition to video data that is streamed from the server to the cache.

The first and second alternative have in common that a cached video's quality is improved as fast as possible with the difference that in version 1 an independent session is created to perform the retransmissions.

The third alternative inherits the advantage of write-through caching that any bandwidth between the cache and server is used only if a client request is directly related to it. This is a major advantage in environments where bandwidth between server and cache is scarce. With the fourth alternative, the cache can decide about which of the cached video retransmissions should be performed when a new video is requested from the server. All four can be supported by retransmission scheduling, since it only determines in which order missing segments should be sent to the cache. The importance of retransmission scheduling is affirmed by the results from Section 5.5.2 ("number of layers") which show that, depending on the number of layers a video consists of, one retransmission phase might not be sufficient to improve the quality of the cached video to its maximum level.

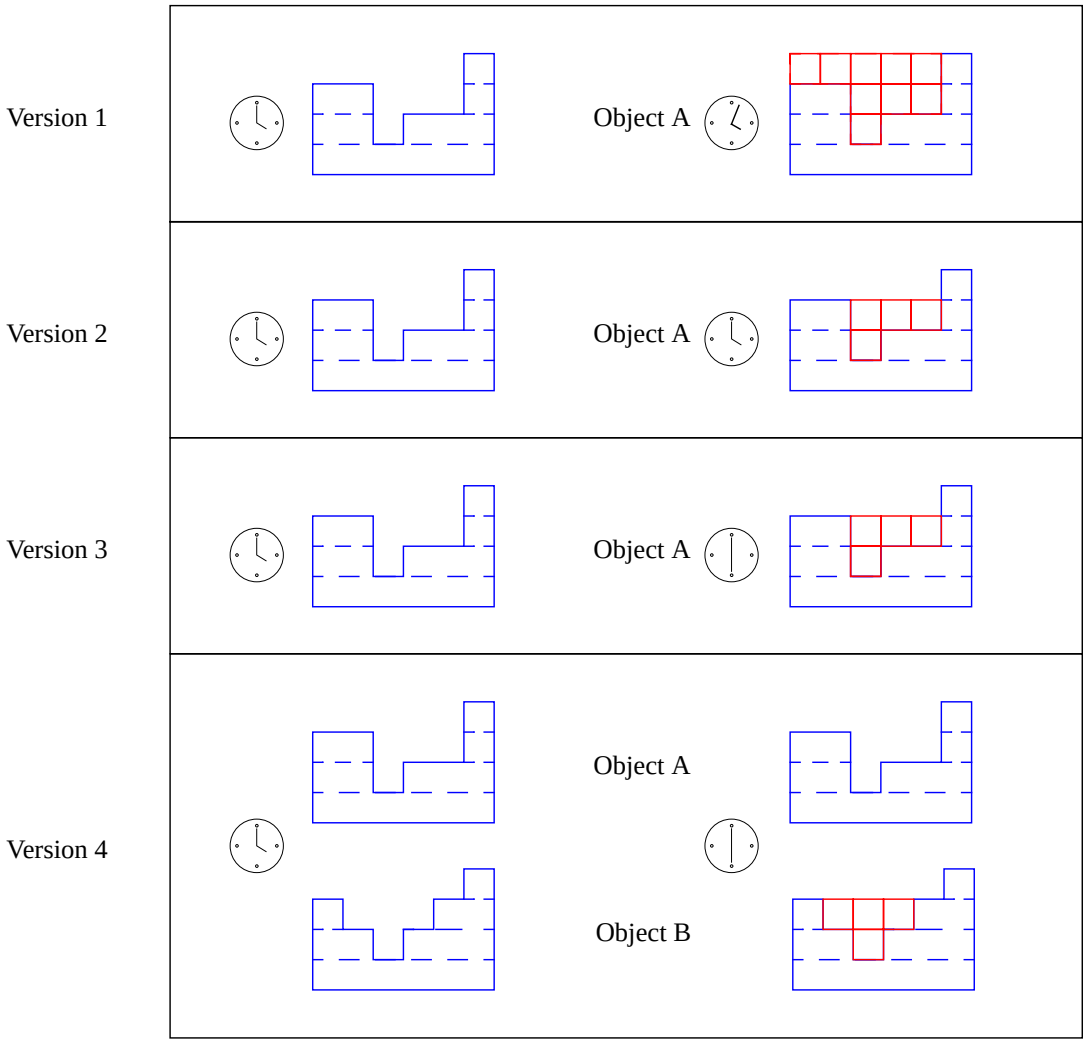


Figure 5.2: Retransmission time

Figure 5.2 gives an example for the four versions when to perform retransmissions. The reason that only four segments are retransmitted in the case of version 3 and 4 is based on the assumption that the additional available bandwidth for retransmissions was equivalent to four segments. This is not the case for version 1 where an independent session is created to transmit missing segments from the server to the cache and the transmission time is not limited.

Let us recall the example from Section 2.6 where students from different universities use the distribution system to receive on-line lectures from remote universities. In this case, the four possibilities to perform retransmissions could look as follows:

- The popularity information of the requested object is known, and based on this information further requests for this specific object are very likely. This might be the case for lectures advanced in time where popularity information from preceding sessions of this lecture reveal that it is very likely further clients served by this cache might also request this object. If the collected popularity data would also allow one to draw conclusions about the time the requests for a single object had been made, and this data shows requests following each other in a time period shorter than the playout duration of the video object, applying the first of the four retransmission options would be a good choice. The quality of the cached video would be increased as soon as possible and clients requesting the object during the retransmission phase would receive the highest possible quality.
- An option to the method above is to perform the retransmission during the actual streaming phase. This can, for example, be performed with the fair share claiming (FSC) method as presented in Chapter 7. The drawback of this method is the fact that probably not all missing segments are retransmitted and the video object is not cached in its highest quality.
- The popularity information about the requested video object might not be known and, therefore, it might not be foreseeable if the object will ever be requested again. In this case, it is not sure that retransmissions will be of any benefit. Thus, retransmissions are only performed if further requests for this object are made. For example, if the first session of a lecture is requested for the first time, no popularity information about this video object belonging to this lecture is available on the cache serving the client posing the request.
- Due to bandwidth limitations not all requested segments may be transmitted in a retransmission session from the server to the cache. Yet, popularity information show a high likeliness for further requests of this object. For example, although retransmission scheduling was performed, a session of a very popular lecture (represented by video object *A*) is not cached in full quality. A client requests a different lecture (video object *B*) with low or unknown popularity. In this case, retransmission scheduling can be performed for video object *A* while the streaming phase of video object *B* is taking place.

5.1.2 Retransmission Focus

Next to the retransmission time, one can also differentiate between a retransmission scheduling approach which has the goal to maximize the quality for the current viewer or an approach that maximizes the quality of the cached video. The first approach is described as *viewer centric*, while the second is described as *cache centric* throughout this thesis. The viewer centric approach will be presented in more detail in Section 5.4 and the cache centric approach in Section 5.6. A modified version of the viewer centric approach called *cache-friendly viewer centric* is presented in *Section 5.7*.

5.1.3 Scheduling Goals

The rationale for making an effort to schedule retransmissions in an intelligent way is that the presentation quality for users that are served from the cache can be enhanced. Therefore, it has to be made explicit what constitutes a quality enhancement, i.e., a goal for retransmission scheduling algorithms to strive for is needed. However, it is commonly assumed and also shown in Chapter 4 that users react very sensitive to quality variations of a video [95]. Hence, a retransmission scheduling algorithm that tries to avoid or even decrease quality variations for a cached video can be considered superior to others which do not take this into account. The results of the subjective assessment performed in Chapter 4 show that the negative effect of quality variations has two dimensions: the frequency of variations and the amplitude of variations. The goal of retransmission scheduling should be to minimize both. In Chapter 4, a new subjective quality metric called the spectrum is presented that can be used to determine the quality of a layer-encoded video. Since the quality of a layer-encoded video is anti-proportional to the resulting value of the spectrum, the retransmission scheduling goal for a video, v, can now be stated as the minimization of the spectrum $s(v)$.

5.2 Optimal Retransmission Scheduling

In this section, the problem complexity of retransmission scheduling by looking at optimal retransmission scheduling is discussed. Since the determination of optimal retransmission schedules is either computationally infeasible or at least intensive, some heuristic schemes are presented. One of them has been proposed in [109], whereas the others are newly developed motivated by the shortcomings of the former.

The goal of retransmission scheduling is to minimize the spectrum of an already cached layered video subject to the constraint that any available bandwidth is used for retransmissions. This constraint ensures that a cached video is further enhanced even if for all time slots the same quality level is reached, i.e., the spectrum equals 0.

A formulation of optimal retransmission scheduling as a mathematical program is given in Figure 5.3. Here, the overall available retransmission capacity is modeled as an estimate. Yet, in this investigations a constantly available bandwidth is assumed, i.e.,

d_t - number of retransmitted layers for time slot t

h_t - number of layers in time slot t

v' - the cached video after retransmissions

H - the maximum number of layers

$\tilde{B}(\hat{t})$ - estimated amount of overall retransmission capacity for all time slots till $\hat{t}$

Minimize $s(v')$ (3)

subject to (4)

$$\sum_{t=1}^{\hat{t}} d_t = \tilde{B}(\hat{t}) \qquad \forall \hat{t} = 1, \ldots, T \tag{5}$$

$$_t + d_t - h_{t-1} - d_{t-1} \leq Hz_t \; \forall t = 1, \ldots, T \tag{6}$$

$$_{t-1} + d_{t-1} - h_t + d_t \leq Hz_t \; \forall t = 1, \ldots, T \tag{7}$$

$$H - h_t \geq d_t \geq 0 \qquad \forall t = 1, \ldots, T \tag{8}$$

Figure 5.3: Optimal retransmission scheduling model.

$$\tilde{B}(\hat{t}) = \frac{B}{T} \times \hat{t}, \tag{9}$$

where B is the overall retransmission capacity for the video. This is certainly a simplifying assumption; yet, the algorithms presented in the following do not depend on it. Simulations with a varying bandwidth presented in Chapter 7 did not show any significant influence on the algorithm's performance.

Optimal retransmission scheduling is a discrete non-linear optimization problem. As such it is, to the best of our knowledge, analytically intractable. It is very similar to the quadratic assignment problem, which is known to be NP-complete [145]. To illustrate the complexity of retransmission scheduling let us consider the search space for an exhaustive search assuming that in each time slot at least one layer is missing, then a *lower* bound for the size of the search space can be obtained:

$$\binom{T}{B},$$

For example, for 100 time slots and a retransmission capacity of 50 this amounts to 1.534×10^{93} possible ways of reordering missing segments (and this is only a very loose lower bound). Thus, given reasonable restrictions on computing power, an exhaustive search for reasonable values of the number of time slots T is computationally infeasible.

Next to the heuristics that are presented in the following the optimal retransmission scheduling algorithm, based on an exhaustive search, was also implemented in the custom simulation envi-

ronment that is described in Section 5.5. To get an impression on how long it would take to determine the optimal set of missing segments that should be retransmitted the execution time for the algorithm was measured for the following example. An initially cached video similar to the one shown in Figure 5.4 is used as the starting point for the algorithm. Table 5.1 shows the duration that it took the algorithm to calculate the optimal set of segments for retransmission starting from 1 up to 7 possible segments. The measurement was executed on a standard PC running Linux RedHat 7.3 with a Pentium III (500 MHz) and 500 MByte of main memory.

Table 5.1: Execution times for optimal retransmission scheduling

Segments to retransmit	1	2	3	4	5	6	7
Opt. Duration (s)	0.0003	0.0054	0.1222	2.8206	56.1878	1168.5	23686.5
Opt. Spectrum	20.73	18.92	16.73	16.4	14.55	14.1	12.55
U-SG-LLF Duration (s)	$5.3x10^{-5}$	$5.4x10^{-5}$	$5.4x10^{-5}$	$5.4x10^{-5}$	$5.5x10^{-5}$	$5.5x10^{-5}$	$5.5x10^{-5}$
U-SG-LLF Spectrum	20.73	22.67	16.73	18.92	18.92	18.92	12.55

The results of this measurement reveal that the optimal algorithm cannot be applied for retransmission scheduling. Especially with the knowledge that the retransmission schedule, in some cases, must be calculated on the order of seconds. In the case of seven segments that can be retransmitted the resulting spectrum that is obtained by the optimal algorithm is identical to the one that is obtained by the application of a user centric unrestricted heuristic (see Figure 5.5, Figure 5.7, and Figure 5.8). The amount of time needed to determine the schedule with the algorithm that is based on the unrestricted heuristic is negligible compared to the optimal algorithm. The resulting values for the spectrum shown in Table 5.1 demonstrate the drawback of the heuristic. The optimal spectrum will not be determined in all cases, but considering the performance gain the heuristics are an applicable alternative.

5.3 Heuristics for Retransmission Scheduling

There are two basic approaches in retransmission scheduling that influence the design of the heuristics developed for this purpose. The first approach is to maximize the quality for the current viewer, while the second is to improve the overall quality of the cached video object. In Section 5.4, heuristics for the first approach are presented and in Section 5.6 it is shown how a simple modification allows the application of these heuristics for the second approach. Section 5.7 presents a combination of the two aforementioned approaches.

5.4 Viewer Centric Retransmission scheduling

As mentioned above, retransmission algorithms are intended to improve the received quality of the actual viewer or the quality of the stored version on the cache. Here, the focus is on improving the quality for the current viewer at the client. Thus, it must be assured that segments requested for retransmission arrive at the client at the actual playout deadline. All of the presented heuristics take this requirement into account, but it is also shown that this is the limiting factor in terms of obtaining an identical copy of the video object on the cache.

5.4.1 Window-based Lowest Layer First (W-LLF)

The first heuristic that is investigated is proposed in [109]. It is fairly simple and called Window-based Lowest Layer First (*W*-LLF), because the cache always looks a certain number of time slots ahead of the current playout time and requests retransmissions of missing segments from the server in ascending order of their layer levels. To ensure that the retransmitted segments do not arrive after their playout time (t_p) to the current client, a prefetching offset O_p for the examined time window is introduced. O_p should be chosen sufficiently large such that O_p > RTT for the transmission path between server and cache at all times. Overall, the time window [$t_p + O_p$, $t_p + O_p + W$] slides over the video in discrete steps of length W until it is finished (t_{end}). The operation of the algorithm is further illustrated in Figure 5.4.

W-LLF bears some obvious disadvantages:

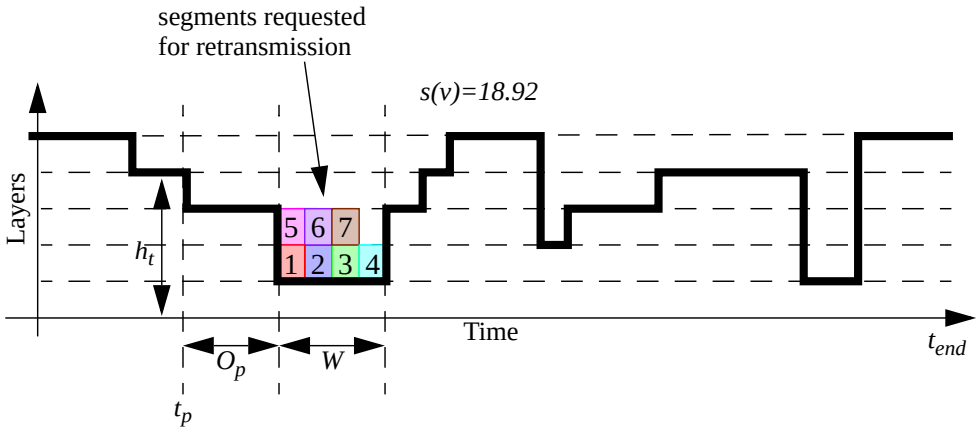


Figure 5.4: W-LLF operation.

- If, e.g., an already complete area (all layers are entirely cached) is scanned, no retransmissions are scheduled for this prefetching window, although there might very well be later parts of the video which could benefit from retransmissions.
- It may be possible that the currently available bandwidth between server and cache would allow the transmission of more segments than those that are missing in the current prefetching window and again additional segments could be requested from the server to allow for a faster quality upgrade of the cached video.

Although, these obvious drawbacks might be eliminated by extensions of the *W*-LLF algorithm, they exhibit a fundamental weakness of *W*-LLF: the restriction of scheduling missing segments for retransmission only for a certain number of time slots ahead. Therefore, *W*-LLF is likely to be rather shortsighted with respect to the scheduling goal of minimizing the spectrum of videos stored on the cache. In the following, a new kind of retransmission scheduling algorithms that eliminates this restricted view is introduced.

One rare case in which *W*-LLF can be superior compared to the heuristics presented in the following is if the viewer decides to not watch the video completely. Then, segments requested for retransmission that are temporally located after the moment the viewer decides to stop watching the video do not improve its perceived quality. This situation can occur with the unrestricted heuristics presented in the following but has no influence on *W*-LLF if the window size is kept relatively small.

5.4.2 Unrestricted Priority-based Heuristics

The problems with *W*-LLF as described above lead to the idea to avoid the use of a prefetching window *W* for retransmission scheduling. That means an unrestricted look at all missing segments ahead of the current playout time t_p (plus the prefetching offset O_p) is taken when making requests for retransmissions from the server. Note that the unrestricted algorithms still send periodic retransmission requests to the server (every *W* time slots) to ensure on the one hand that retransmissions and playout to the client are kept synchronized and on the other hand that the modified shape of the cached video due to retransmitted segments can be taken into account by the scheduling algorithms.

A further goal was to adapt the scheduling decisions to the results obtained from the subjective assessment (see Chapter 4), instead of rigidly assigning the highest priority to the first segments with the lowest layer level, as performed by previous approaches [109].

In the following, three heuristics of the more general class of unrestricted priority-based retransmission scheduling algorithms are described.

A) Unrestricted Lowest Layer First (U-LLF)

This algorithm is very similar to *W*-LLF because it uses as priority solely the layer level. In contrast to *W*-LLF, however, it always scans the interval $[t_p + O_p, t_{end}]$ in order to request missing segments from the server (every *W* time slots). Figure 5.5 gives an example how the missing segments would be scheduled for retransmission. The numbers for each segment define the order in which the segments should be sent from the server.

B) Unrestricted Shortest Gap Lowest Layer First (U-SG-LLF)

Considering the definition of the spectrum in Section 5.1.3 and taking into account the scheduling goal of minimizing the spectrum, one can observe that there are, in principle, two ways to decrease the spectrum of a video: to increase the lowest quality levels (which is taken care of by choosing the lowest layer levels first), *or* to close gaps in the video, i.e., reduce the number of

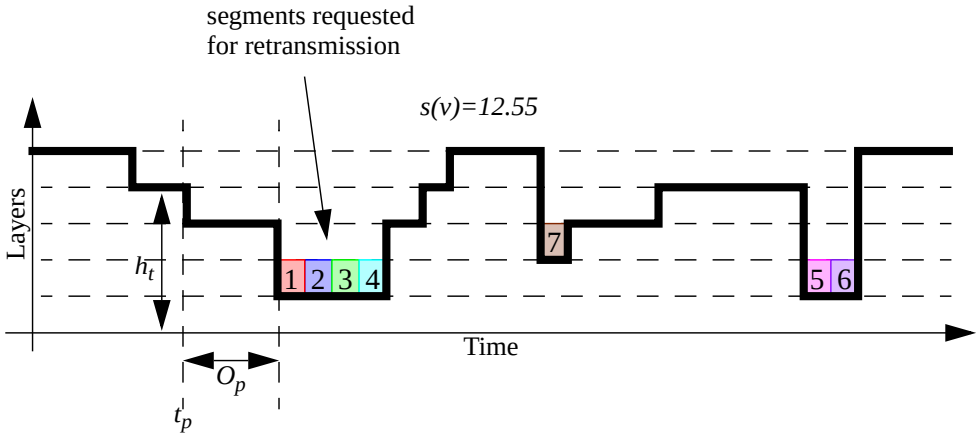


Figure 5.5: U-LLF operation.

$z_t \neq 0$. The latter is not captured by simply using layer levels as priorities. Figure 5.6 gives an illustrative example of how the spectrum is affected by the closing of gaps.

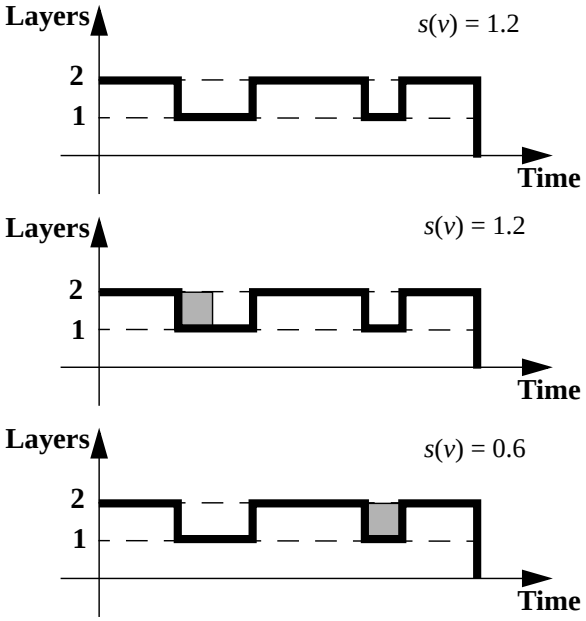


Figure 5.6: Influence of closing gaps on spectrum.

The influence of closing gaps on the spectrum can potentially be quite high. Therefore, in contrast to *W*-LLF and U-LLF, a prioritization of the missing segments is performed which also takes the closing of gaps into account. This prioritization is achieved by first sorting the segments according to the length of the gap they belong to and then use their layer levels for further sorting. The resulting heuristic is called Unrestricted Shortest Gap Lowest Layer First (U-SG-LLF). An example of the scheduling of missing segments is given in Figure 5.7.

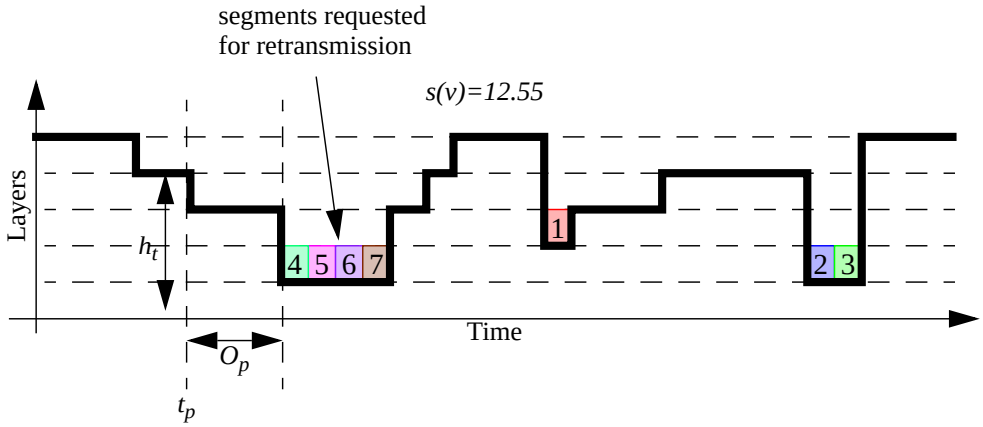


Figure 5.7: U-SG-LLF operation.

C) Unrestricted Lowest Layer Shortest Gap First (U-LL-SGF)

Since it is by no means clear which sorting criterion (i.e., gap length or layer level) should be used first, a further heuristic where missing segments are first sorted by their layer level and then sorted further by gap lengths was investigated. This heuristic is called Unrestricted Lowest Layer Shortest Gap First (U-LL-SGF). Comparing Figure 5.7 and Figure 5.8 shows that this heuristic can result in a different retransmission schedule.

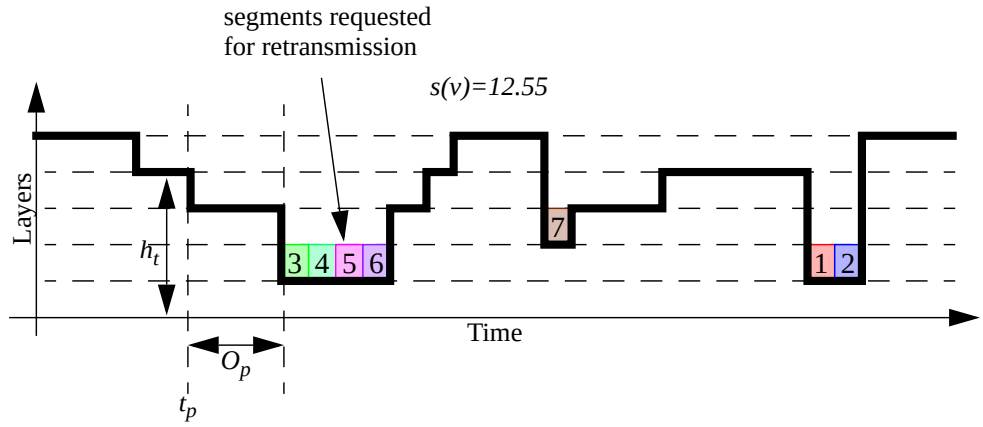


Figure 5.8: U-LL-SGF operation.

5.5 Simulations

In order to compare the different retransmission scheduling algorithms from the previous section and investigate their dependency upon different parameters, a number of experiments based on a custom simulation environment (implemented in C++) were performed.

The simulations were performed in the following manner. For each simulation an instance of a layered video on the cache is randomly generated. Here, such a layered video instance is modeled as a simple finite birth-death process since it is the result of the congestion-controlled video transmission which restricts state transitions to direct neighbor states. $\{0,..., H\}$ is the state space and

birth and death rate are chosen to be equal to $1 - 1/\sqrt{3}$ (for all states) which results in a mean length of 3 time units for periods with stable quality level[1]. A discrete simulation time is used where one unit of time corresponds to the transmission time of a single segment. In Figure 5.9, an example video instance generated in this way is given.

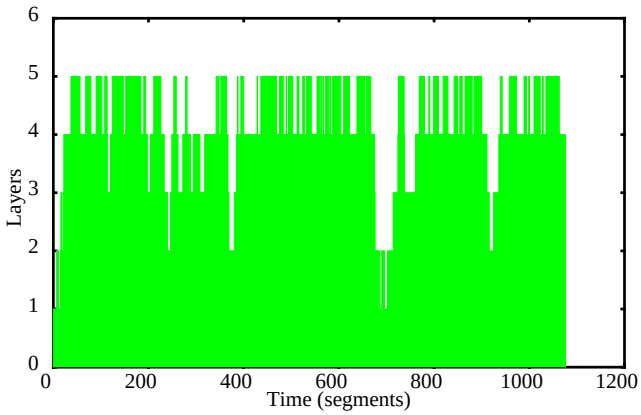


Figure 5.9: Randomly generated layered video on the cache.

The simulation environment allows the application of the different algorithms described in Section 5.3 and to vary parameters such as the bandwidth available for retransmissions between server and cache. During the simulations, the spectrum (as defined in Chapter 4) of the cached video instances is continuously calculated, and the different algorithms' performance is compared, given parameters such as the available bandwidth. In all simulations a prefetching offset of $O_p = 5$ segments is assumed.

5.5.1 Comparison of the Heuristics

At first, a series of 1000 simulations with all retransmission scheduling algorithms from Section 5.3 where all parameters were chosen identical (except the windows sizes for *W*-LLF) was performed. This large sample ensured that the 95% confidence interval lengths for the spectrum values were less than 0.5% of the absolute spectrum values for all heuristics. The results for the evolution of the spectrum values for the different algorithms are shown in Figure 5.10.

These results indicate that there is a significant gain with respect to the spectrum of the cached video for the unrestricted retransmission algorithms in comparison to *W*-LLF. Of course, if window sizes are chosen large enough for *W*-LLF it improves and finally approaches U-LLF.

By taking a closer look at Figure 5.10 it becomes clear that the final spectrum of the U-SG-LLF heuristic is lower by a value of 100 compared to U-LLF and U-LL-SGF. Taking into account the results presented in Table 4.3 where a spectrum improvement by the value of 2 already leads to an increase in the subjective quality, U-SG-LLF can be seen as significantly better than the other heuristics.

[1] The parameter choice is rather arbitrary. However, simulations with other values showed no significant impact on our results.

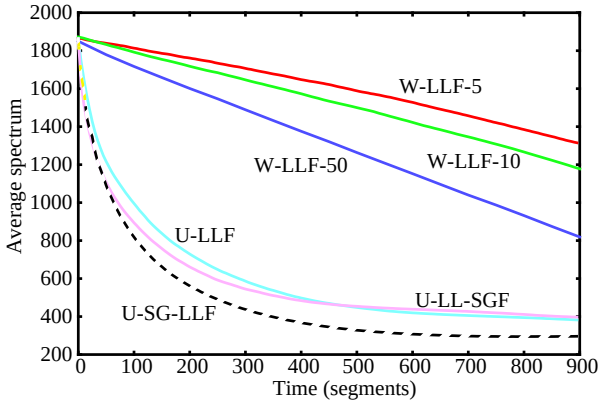


Figure 5.10: Average spectrum of 1000 simulation runs for each heuristic. (10 layers, retransmission bandwidth = 2, window size = 5 (*W*-LLF-5), window size = 10 (*W*-LLF-10), window size = 50 (*W*-LLF-50)).

In Figure 5.11 the results of one simulation for all four heuristics are shown. The single figures show the differences between the heuristics in more detail. For the case of W-LLF-5 the amount

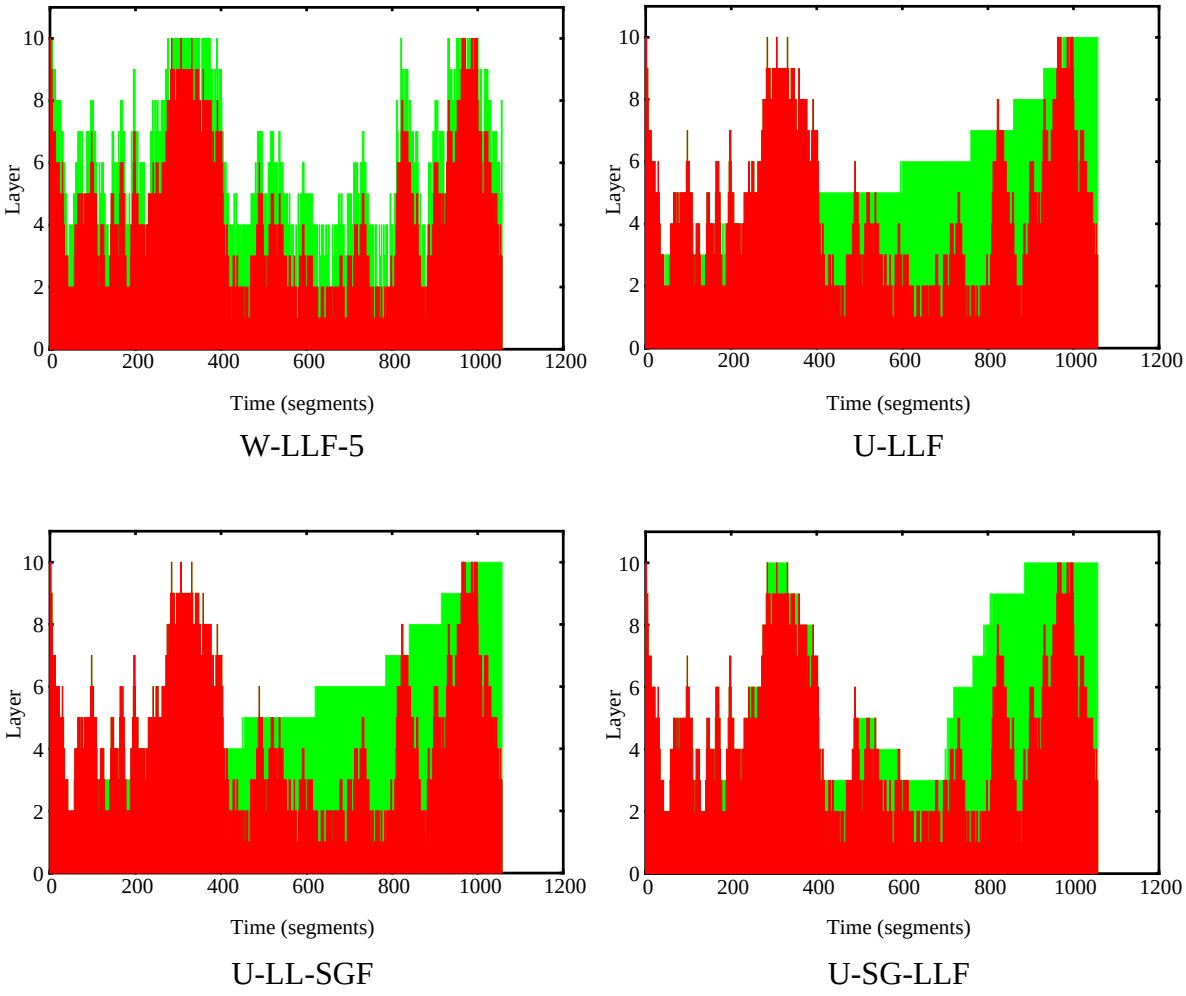


Figure 5.11: Single simulation of all four heuristics

of layer variations is not significantly reduced which, in contrast, is the case for the unrestricted heuristics. Comparing the unrestricted heuristics with each other shows that U-LLF and U-LL-SGF behave almost identical while the result of U-SG-LLF is quite different and better with respect to the spectrum. All three have in common that a large amount of retransmitted segments are located in the second half of the sequence (time segments 500 - 1000). This effect will be discussed in more detail in Section 5.5.3. Overall these results reflect what is also shown in Figure 5.10: the spectrum is reduced only by a small amount in the case of W-LLF, U-LLF and U-LL-SGF behave almost similar, and U-SG-LLF provides the largest spectrum reduction.

5.5.2 Parameter Dependency Analysis

In the following, the heuristics' dependencies on certain parameters are investigated. For all of these simulations, the U-SG-LLF heuristic is used only, since it showed the best performance of all heuristics in the experiment of the preceding section.

A) Number of Layers (Increasing Bandwidth)

For this simulation, the number of layers per cached video were set to be either 5, 10, or 20 layers. It is assumed that the single layer size is constant and, thus, the bandwidth needed to stream a 20 layer video is twice as much as the one needed for a 10 layer video. To isolate the effect of encoding videos with different number of layers, the available retransmission bandwidth was scaled in proportion to the number of layers, i.e., for 5 layers 2 segments of retransmission bandwidth per time unit were assumed, for 10 layers 4 segments, and for 20 layers 8 segments. For each of these three alternatives 1000 simulations were performed and again the average of the spectra over time was calculated.

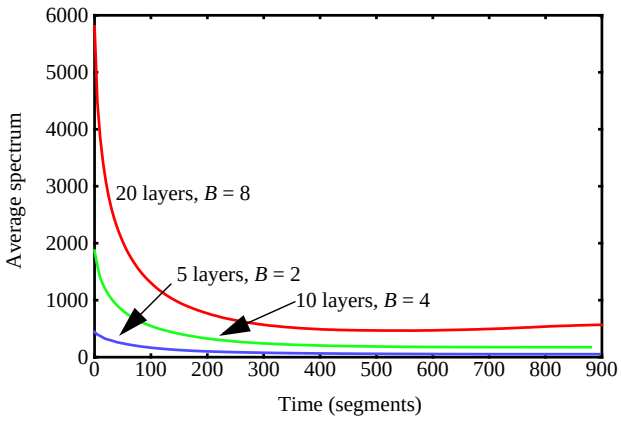


Figure 5.12: Different number of layers.

As Figure 5.12 shows, the spectrum converges for each of the 5 and 10 layer video. For the 20 layer video the spectrum increases slightly in the end. This effect is explained in Section 5.5.3. Yet, the higher the number of layers the higher the average spectrum. This is intuitive because the more fine-grained the layered encoding the more variations may be introduced during a conges-

tion-controlled transmission and the harder it is for the retransmission scheduling to smooth these variations.

B) Number of Layers (Constant Bandwidth)

For this simulation, the number of layers per cached video are determined to be either 5, 10, or 20 layers. Increasing the number of layers in this case does not increase the maximum bandwidth of the layer-encoded video. Instead, the bandwidth of each single layer is decreased. In our work on *Fair Share Claiming* ([86] and Chapter 7), it is investigated how a TCP-friendly transmission exploits its fair share of network resources taking into account that the constrained granularity of layer-encoded video inhibits an exact adaptation to actual transmission rates.

The result of this investigation shows that available bandwidth for retransmission is reduced with an increasing number of overall layers. Thus the bandwidth available for retransmission was not increased as in the simulation presented above. The retransmission bandwidth is set to one segment per time unit for all three types of video objects. Also here 1000 simulations were run for each of the three heuristics. As Figure 5.13 shows, the spectrum converges for each of the three alternatives. A comparison to the results presented in Figure 5.12 shows that the final average spectrum for the 10 and 20 layer cases are higher because the amount of segments that could be retransmitted is lower due to a reduced relative bandwidth. This simulation shows that one retransmission phase might not always be sufficient to obtain full quality for the cached content, since the spectrum is not reduced to 0.

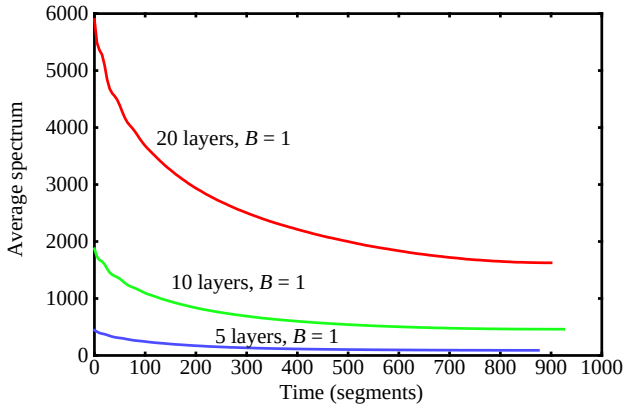


Figure 5.13: Different number of layers (const. bandwidth)

C) Available Retransmission Bandwidth

In the next set of simulations, the effect of different amounts of available retransmission bandwidth on the performance of U-SG-LLF was investigated.

Not surprisingly, the spectrum converges faster with a higher available retransmission bandwidth, as shown in Figure 5.14. The reason for the very similar spectrum curves for $B = 6$ and 10 is due to sufficient retransmission bandwidth for both cases which allows to retransmit all missing segments that have a playout time larger than 500. Due to the prefetching offset, missing segments from the beginning cannot be retransmitted and, therefore, a spectrum of 0 is not achieved.

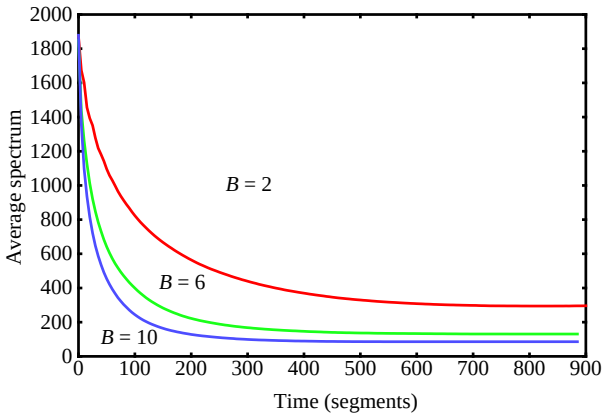
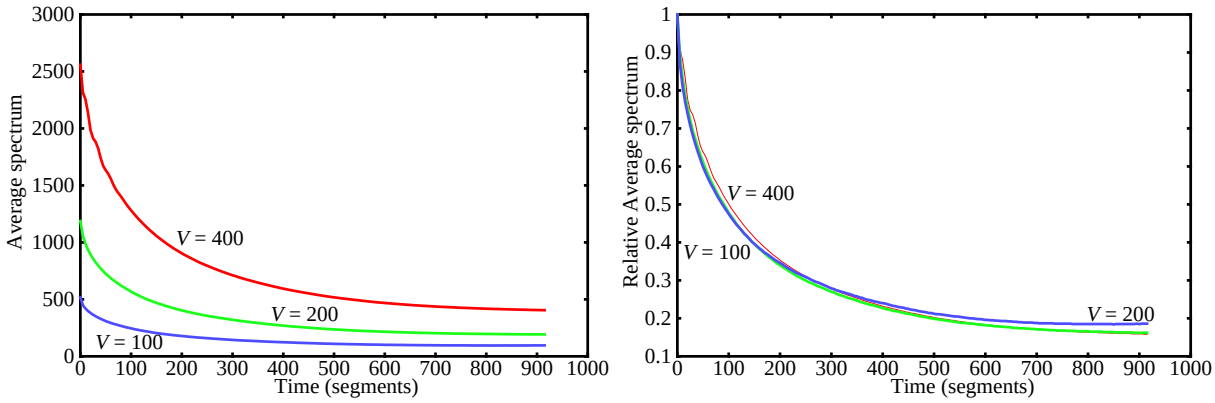


Figure 5.14: Different amounts of available retransmission bandwidth.

D) Amount of Layer Variations

Depending on the situation on the network the amount of layer variations that are introduced by the adaptive transmission can vary. To investigate the behavior of the amount of layer variations on the retransmission scheduling heuristics an additional investigation was performed in which the amount of those variations was different. Three different simulations were performed assuming that the number of layer variations is 100, 200, and 400, respectively. As shown in Figure 5.15 (a), the value of the spectrum is proportional to the amount of layer variations. Observing the relative reduction of the spectrum through the algorithm shows (see Figure 5.15 (b)) that it performs almost identical independent from the initial amount of layer variations. The relative spectrum is obtained as the ratio between the actual value of the spectrum and the maximum value for each time step.



Different amount of layer variations (Spectrum)
(a)

Different amount of layer variations (Relative Spectrum)
(b)

Figure 5.15: Amount of layer variations and relative spectrum

E) Initial Transmission Quality

Finally, a series of simulations were performed in which different initial transmission qualities were assumed resulting in cached videos where the maximum number of cached layers is lower than the maximum number of layers for the original video. In contrast to the preceding experiments, the spectrum values are not sampled but a single simulation is used since the relevant effects can be shown in more detail. For each simulation, the ratio between the maximum of cached layers (MCL) and the maximum of original layers (MOL) is modified.

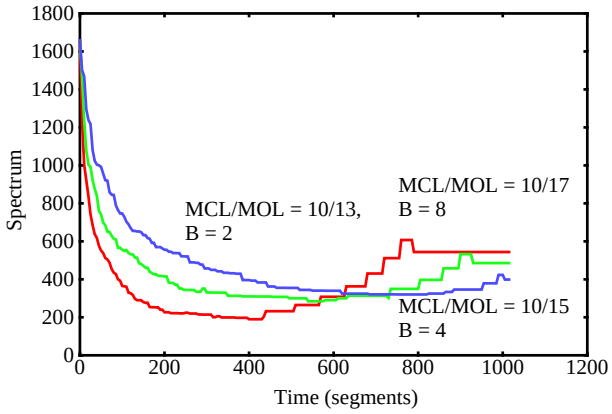


Figure 5.16: Influence of initial transmission quality.

As Figure 5.16 shows, spectrum values start to rise again for the last third of the video. This effect is especially pronounced for worse initial transmission qualities (MCL/MOL=10/17). To intensify the illustration of this effect the bandwidth available for retransmissions is proportionally increased according to the MOL and is set to 2, 4, and 8, respectively.

Looking at the cached video that results from the retransmission scheduling heuristic (U-SG-LLF) in Figure 5.17 sheds light on the reason for this effect. It can be observed that the retransmission scheduling "builds a staircase" at the end of the cached video which is not beneficial with respect to the minimization of the spectrum. The reason for this behavior is that the algorithm only considers missing segments ahead of the playout time ($t_p + O_p$). Thus, if all gaps are closed the algorithm starts to request segments from the next layer starting from $t_p + O_p$. This happens several times leading to the staircase shape exhibited in Figure 5.17.

5.5.3 Totally Unrestricted Heuristics

As a consequence of the results from Section 5.5.2, a further investigation should show how the modification of the heuristics in a way that retransmissions are not limited to segments that are located after $t_p + O_p$, could cure the problem the heuristics had with bad initial transmission qualities. However, it has to be observed that such a totally unrestricted retransmission scheduling algorithm bears the possibility that retransmitted segments may arrive too late for the current client and might thus be retransmitted in vain if no other client ever requests that video. Thus, some of the attractiveness of write-through caching is lost. On the other hand, it also offers the chance to obtain a complete copy of the video on the cache.

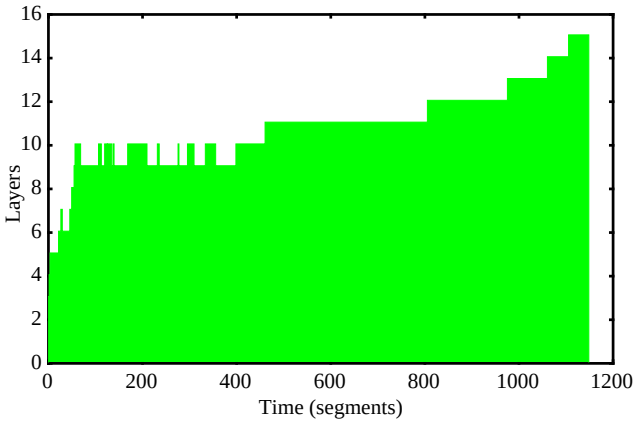


Figure 5.17: Cached video after retransmission phase.

The simulations from Section 5.5.2 were repeated with the now totally unrestricted version of U-SG-LLF. The results are shown in Figure 5.18.

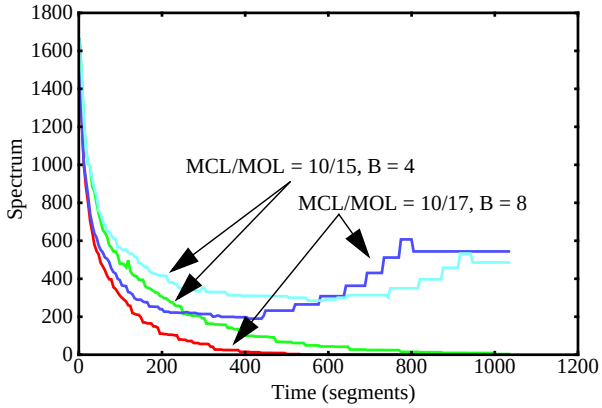


Figure 5.18: Comparison of restricted and totally unrestricted heuristic.

Obviously, the problem of rising spectrum values is solved. This observation is reinforced when the cached video as it results from the totally unrestricted heuristic in Figure 5.19 is compared to its counterpart in Figure 5.17.

However, in order to assess how many segments would be scheduled for retransmission which could not be viewed any more by the current viewer, also these “late” segments were recorded: with MCL/MOL = 10/15 38% and with MCL/MOL = 10/17 37% of the retransmitted segments arrived too late. This is certainly a substantial number of late segments and, thus, one has to make a decision between generating a fairly smooth cached video and using all available retransmission bandwidth to benefit the current client.

5.6 Cache Centric Retransmission Scheduling

The heuristics presented in Section 5.4 are designed to maximize the perceived quality for the current viewer of the video. Since this can have a negative effect on the cached video object as shown in Figure 5.18 and previous results from Section 5.5.3 imply a possible solution for this problem,

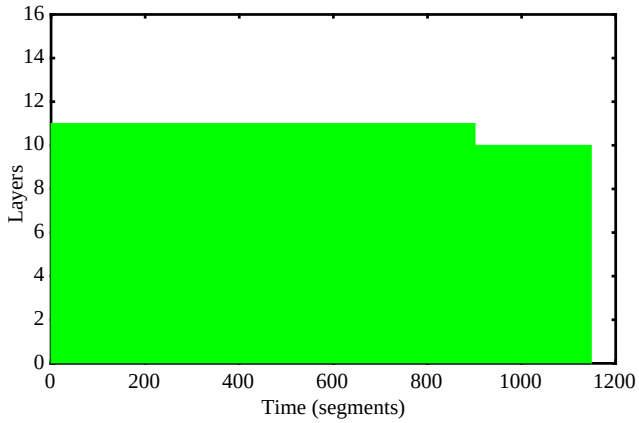


Figure 5.19: Cached video after retransmission phase for totally unrestricted heuristic.

the unrestricted heuristics presented in Section 5.4 are modified. With this modification segments for which the playout time t_p has already passed, can also be scheduled. Thus, the overall quality of the cached video is improved for all potential viewers and not only the current one. This rather *social* approach is called *cache centric* retransmission scheduling in contrast to *viewer centric* retransmission scheduling. Since it is debatable which of the strategies (client or cache centric) should be applied for retransmission scheduling, a possible cache operator should have the chance to choose between both. To investigate the behavior of the heuristics for cache centric retransmission scheduling an additional set of simulations was performed. As a representative for all unrestricted heuristics Figure 5.20 depicts the behavior of the cache centric U-LLF heuristic.

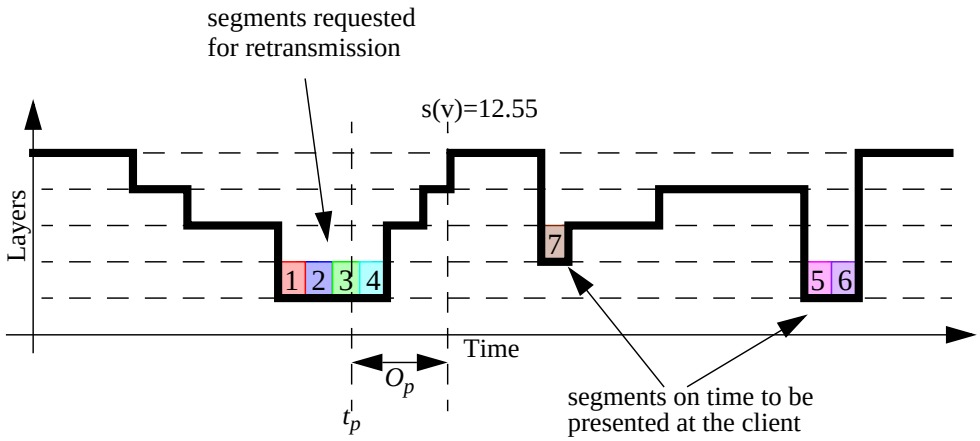


Figure 5.20: Cache centric U-LLF operation.

The simulation environment is identical to the one presented in Section 5.3 with the exception that the implementation for the unrestricted heuristics was slightly modified as described above.

5.6.1 Comparison of the Heuristics

As in the simulation for the viewer centric retransmission scheduling heuristics, a series of 1000 simulations with all three modified retransmission scheduling algorithms from Section 5.6 where

all parameters were chosen identical was performed. This investigation was performed in order to see if the cache centric retransmission scheduling heuristics would behave in a different way. As can be seen in Figure 5.22 the results of the simulations meet the expectations. The final spectrum of the cache centric heuristics is significantly lower than the spectrum of the viewer centric ones. (To show this difference in more detail the result for U-SG-LLF from Section 5.5.1 is shown in Figure 5.22 too.) The final spectrum of the cache centric U-SG-LLF heuristics is lower by a value of 217 compared to the viewer centric U-SG-LLF heuristic.

As with the simulations for the viewer centric heuristics, of all the algorithms, U-SG-LLF performed best. Also here the results of a single simulation for all three heuristics are shown. As can be seen from Figure 5.21 the staircase-effect introduced by the heuristics from Section 5.4 does not occur with the totally unrestricted heuristics. Similar to the results presented in Figure 5.11 U-LLF and U-LL-SGF perform almost identical while the outcome of U-SG-LLF is different.

5.6.2 Spectrum for the Current Viewer

To assure that the cache centric approach is also beneficial for the current viewer the spectrum of the received video at the client is calculated additionally. As can be seen in Figure 5.23 (a) the

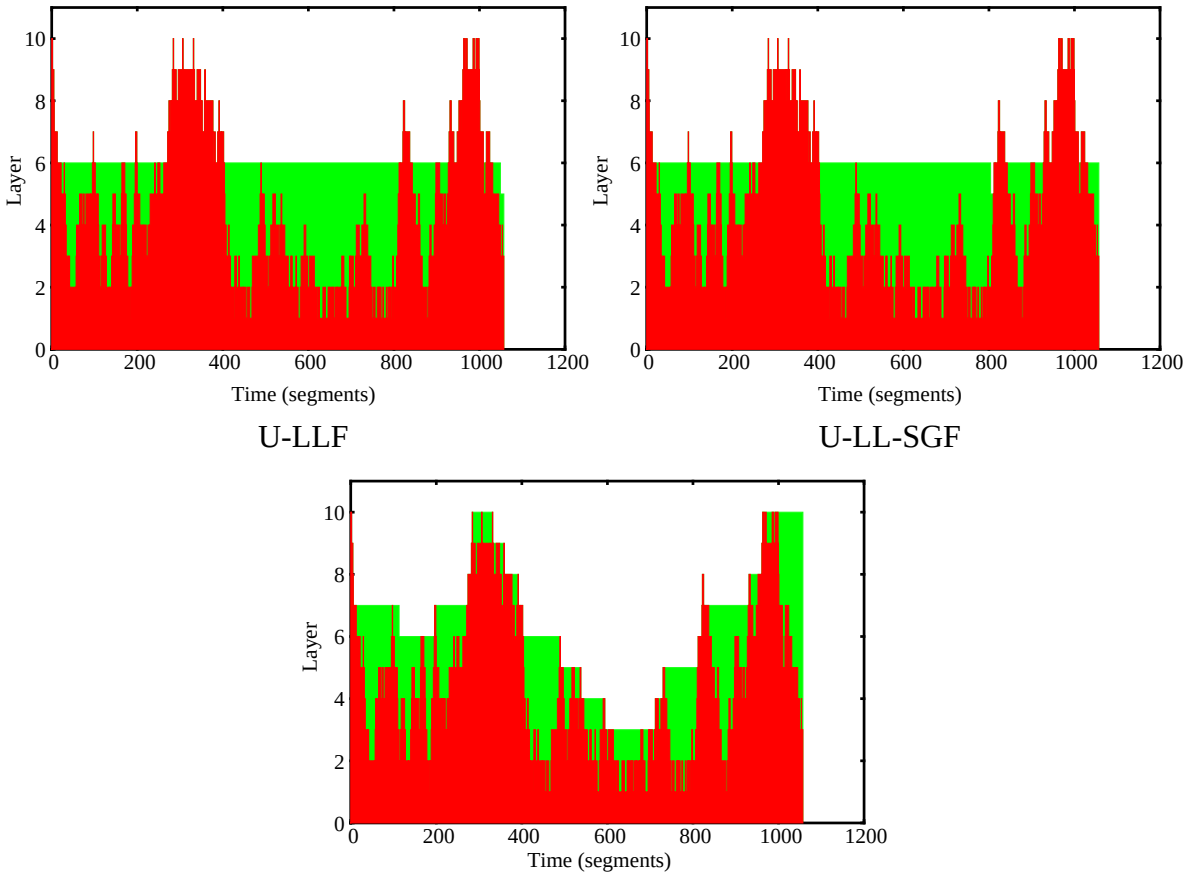


Figure 5.21: Single simulation of totally unrestricted heuristics

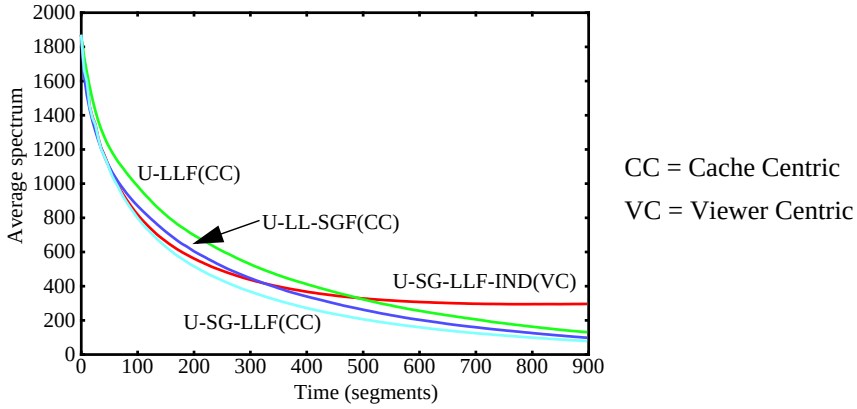
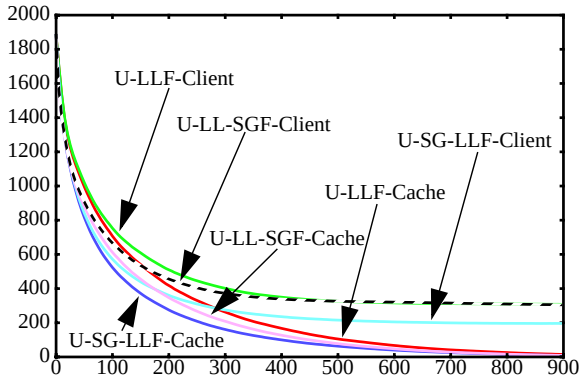


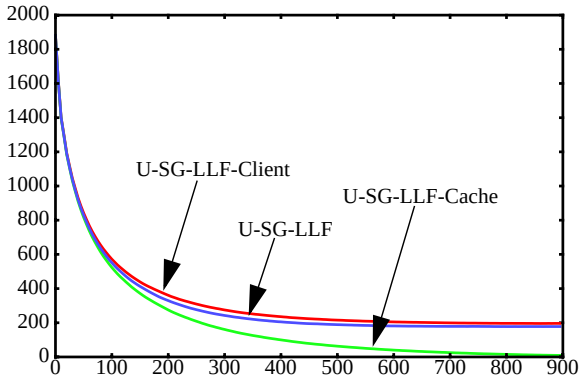
Figure 5.22: Average spectrum of 1000 simulation runs for each heuristic (10 layers, retransmission bandwidth = 2).

spectrum at the client is significantly higher than the one at the cache. The resulting spectrum at the cache is almost reduced to a value of 0, while the resulting spectrum at the client is a least 200 (for the case of U-SG-LLF). The cause for the increased spectrum at the client is based on the timing constraints for the playout of the single segments of the video (see Section 5.5.3). A comparison between the spectrum of the viewer centric approach and the spectrum at the client for the cache centric approach (Figure 5.23 (b)) shows that the latter is only slightly worse. The spectrum at the cache is equal to the spectrum at the client in the case of the viewer centric approach and, thus, worse than the one for the cache centric approach. Therefore, the cache centric approach has the advantage of achieving a better quality at the cache, while the resulting quality at the client is only slightly worse compared to the viewer centric approach. In Section 5.7, an approach is presented which combines the benefits of the client- and cache centric approaches.



Cache and client spectrum for the cache centric approach

(a)



Cache and client spectrum for the cache centric approach and client spectrum for the viewer centric approach

(b)

Figure 5.23: Cache vs. client spectrum

5.6.3 Parameter Dependency Analysis

A parameter dependency analysis is also performed for the cache centric heuristics. Similar to the analysis for the viewer centric heuristics only the U-SG-LLF heuristic is regarded for this analysis, since it showed the best performance of all heuristics in the experiment of the preceding section.

A) Number of Layers (Increasing Bandwidth)

As in the simulation in Section 5.5.2 the number of layers per cached video were set to be either 5, 10, or 20 layers. In this simulation it is assumed that the single layer size is constant and, thus, the bandwidth needed to stream a 20 layer video is twice as much as the one needed for a 10 layer video. The results of the simulation (shown in Figure 5.24) are similar to the ones for the viewer centric heuristic with the difference that the final spectrum reaches a lower value.

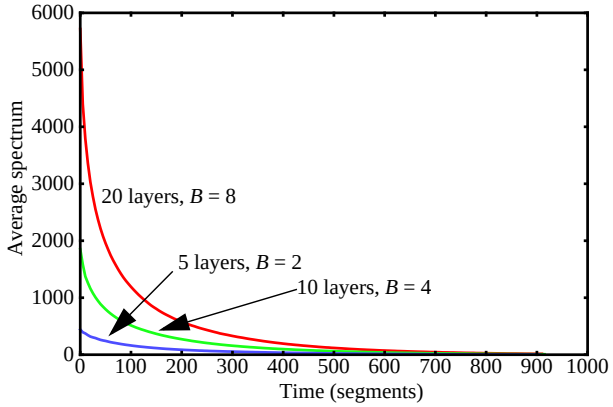


Figure 5.24: Different number of layers (increasing bandwidth)

B) Number of Layers (Constant Bandwidth)

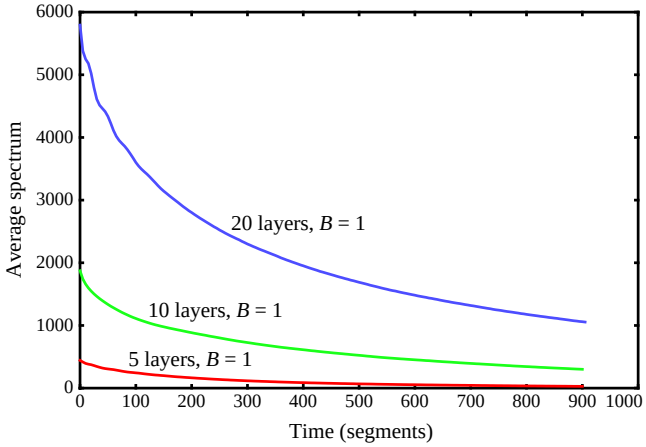


Figure 5.25: Different number of layers.

Also here the results of the simulation are similar to the corresponding ones in Section 5.5.2. Note that the values for the final spectra (Figure 5.25) are lower than in the case of the viewer centric heuristic (Figure 5.14). As in Table 5.5.2, one retransmission run might not be sufficient to

retransmit all missing segments to the cache. (Even for the object that consists of up to 5 layers the final spectrum is larger than zero.)

C) Available Retransmission Bandwidth

In the next set of simulations, the effect of different amounts of available retransmission bandwidth on the performance of cache centric U-SG-LLF was investigated. Not surprisingly, the

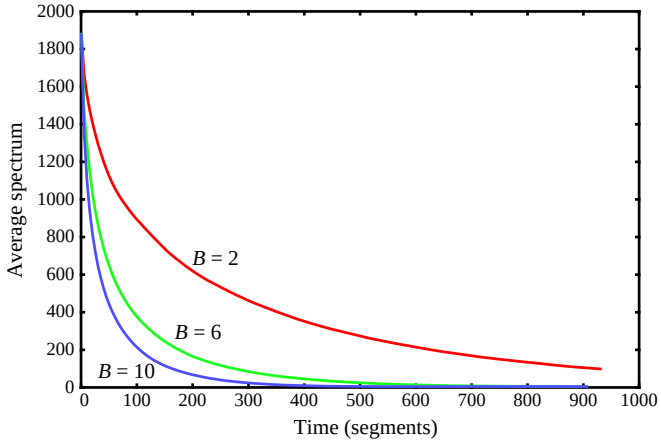
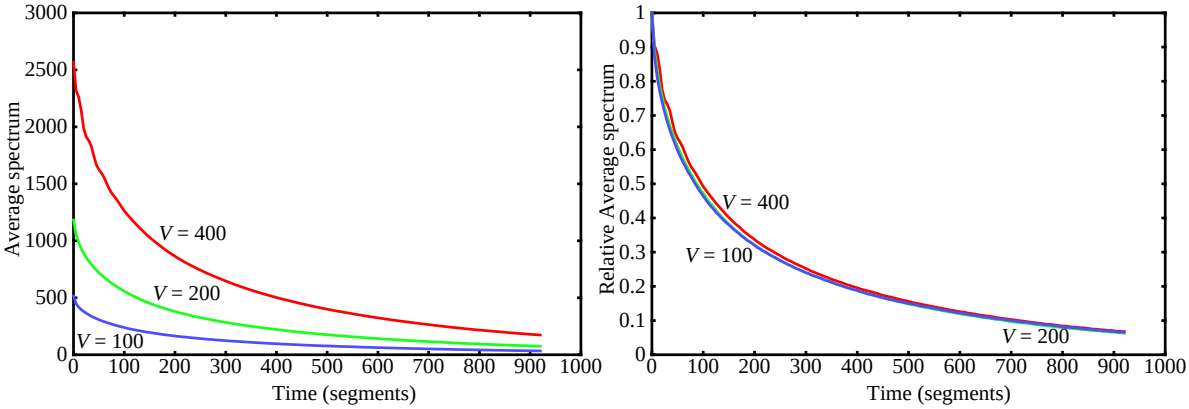


Figure 5.26: Different amounts of available retransmission bandwidth (10 layers).

spectrum converges faster with a higher available retransmission bandwidth, as shown in Figure 5.26. The reason for the very similar spectrum curves for B = 6 and B =10 is due to sufficient retransmission bandwidth for both cases which allows one to retransmit all missing parts of the cached video. Thus, in contrast to the simulation in Section 5.5.2 the video is stored on the cache in full quality.

D) Amount of Layer Variations



Different amount of layer variations (Spectrum)
(a)

Different amount of layer variations (Relative Spectrum)
(b)

Figure 5.27: Amount of layer variations

The result of the simulation on the amount of layer variation on the cache centric U-SG-LLF heuristic is shown in Figure 5.27. As expected, also here the spectrum is proportional to layer varia-

tions but the convergence is stronger for the cache centric heuristic as can be seen from the relative average spectrum of this simulation.

5.7 Cache-friendly Viewer Centric Retransmission Scheduling

One special case that can occur with viewer centric retransmission scheduling is the one in which no more segments are retransmitted, although there are still missing segments. This is the case when all missing segments with a playout time larger than t_p are retransmitted as shown in Figure 5.28 for the U-LLF heuristic. At this point in time there exist no further segments which have a playout time larger than t_p but there is still the possibility to transmit missing segments with a playout time that has already passed. Those segments are useless at the client and can only be used to improve the quality of the cached content. Thus, they should not be forwarded from the cache to the client. This approach is called *cache-friendly viewer centric* retransmission scheduling.

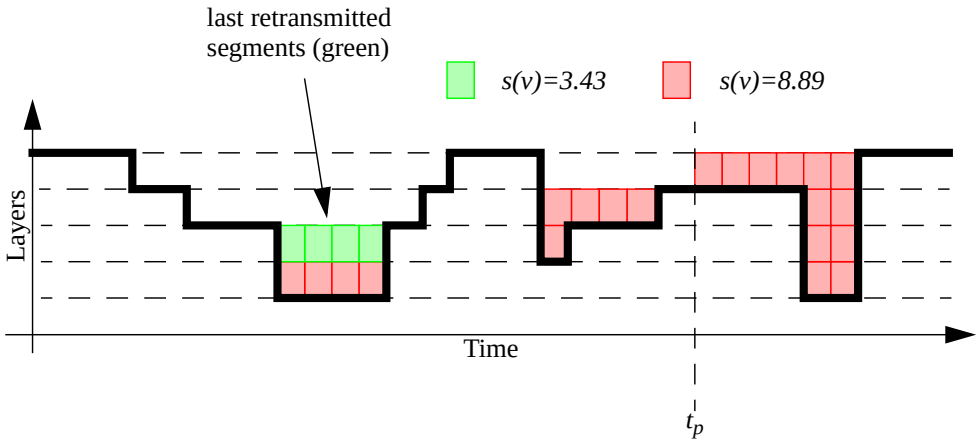
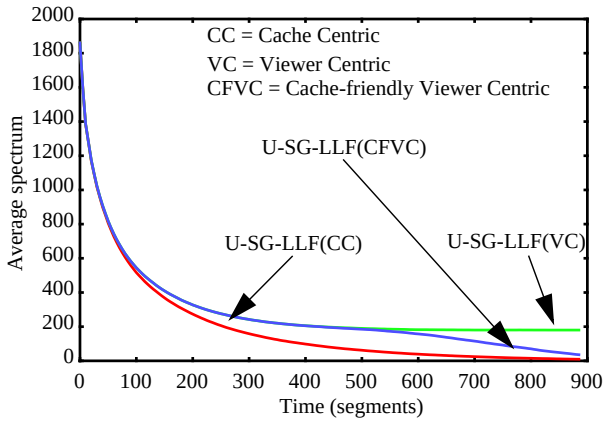


Figure 5.28: Cache-friendly viewer centric U-LLF operation.

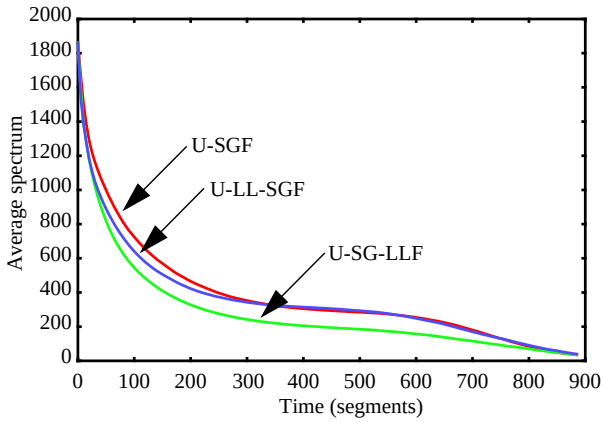
In Figure 5.28, only the red segments are forwarded to the client since they arrived at the cache early enough to allow a timely playout at the client. The green segments are useless at the client but improve the quality of the cached video which can be beneficial for other clients requesting that video at a later point in time. The perceived quality for the viewer is exactly the same as in the case of plain viewer centric retransmission scheduling while the resulting quality of the cached video is always better in the case of cache-friendly viewer centric retransmission scheduling. As it is shown in Chapter 7, the additional bandwidth that is needed to transmit those cache-friendly segments does not influence the quality of the stream that is forwarded to the client, if the fair share claiming transport mechanism is used. In this case, missing segments are only retransmitted if enough additional bandwidth on the link between server and cache is available.

5.7.1 Simulations



Comparison of the simulation results between viewer centric, cache centric, and cache-friendly viewer centric U-SG-LLF heuristic.

(a)



Comparison of the simulation results between the three heuristic for the cache-friendly viewer centric approach.

(b)

Figure 5.29: Average spectrum of 1000 simulation runs for each heuristic (10 layers, retransmission bandwidth = 4)

Using the same simulation environment as described in Section 5.5, a series of simulations were performed with the cache-friendly viewer centric heuristic. In Figure 5.29 (a) the different retransmission scheduling approaches (cache centric, viewer centric, cache-friendly viewer centric) for the U-SG-LLF heuristic are compared with each other. The comparison of the three different heuristics for the cache-friendly viewer centric approach (Figure 5.29 (b)) affirms the results of the simulations presented in Section 5.5.1 and Section 5.6. Also in the simulation for the approach presented here U-SG-LLF is the best performing heuristic, while U-LLF and U-LL-SGF perform almost similar. Since U-SG-LLF again is the best performing of the three heuristics, it was chosen for the comparison between the different approaches depending on the retransmission focus. As shown in Figure 5.29, the cache friendly viewer centric approach results in a better spectrum on the cache than the plain viewer centric approach. On the other hand, the resulting spectrum for the cache-friendly viewer centric approach is slightly higher than the cache centric approach. Note that the resulting spectrum for the cache-friendly approach at the client is identical to the spectrum of the plain viewer centric approach.

5.8 Summary

The work presented in this chapter focused on the problem of how to deal with retransmissions of missing segments for a cached layered video in order to meet users' demands to watch high quality video with relatively little quality variations. After the introduction of the basic retransmission scheduling approaches and the motivation of the scheduling goal, the complexity of the optimal retransmission scheduling algorithm and the drawbacks of this algorithm are shown. Those results lead to the development of different retransmission scheduling heuristics. The development of

these heuristics was influenced by the results of the subjective assessment on variations in layer-encoded video (see Chapter 4) and preceding work by [109]. Based on the retransmission focus, that means, should the current viewer or the cached version of the video be maximized, three different approaches viewer centric, client centric, and cache-friendly client centric are proposed.

A custom simulation environment was built to compare the proposed heuristics with each other. This simulation does not only allow a comparison of the single heuristics between each other but additionally the performance of the heuristic if used in the scope of a certain retransmission approach (viewer centric, cache centric, cache-friendly viewer centric) can be compared with each other. In addition, simulations on the dependency of the heuristics on system parameters are performed.

The results of the simulations show that the newly created heuristics can outperform an existing one. They also show that one of the three new heuristics (U-SG-LLF) outperforms the remaining two. The comparison of the results for the different retransmission approaches shows that the cache-friendly viewer centric approach can be seen as a good compromise, since the quality for the current viewer is maximized, while the quality of the cached object can be increased close to the maximum.

Nevertheless, administrators of a VoD service that makes use of retransmission have the freedom to choose one of the three approaches based on their preferences without requiring significant changes on the overall SAS architecture.

Chapter 6: Polishing

6.1 Motivation

The combination of caching and adaptive streaming bears the disadvantage that a layer-encoded video can only be cached according to the available bandwidth of the path between server and cache, if a congestion controlled transport mechanism is applied. The created copy of the video stored on the cache might, depending on the path's condition, contain a large amount of layer variations. Thus, forwarding such a video object towards a client might be annoying for a watching user.

In this chapter, a new technique, called polishing, is presented. With polishing a cache considers to send only a subset of the segments of a locally stored object in order to reduce layer variations to the client. Investigations by performing a subjective assessment of layer variations in videos (see Chapter 4) have shown that it can be beneficial to omit the transmission of certain segments, especially if the amount of layer variations is reduced.

At first, this might sound rather strange since some information is not transmitted at all and, thus, the PSNR of the video is reduced. Yet, investigations in Chapter 4 have shown that, despite reducing the PSNR, reducing layer variations can increase the perceived quality of a video, if done carefully. That is, in the previously performed subjective assessment, a layered video with a lower PSNR but less layer variations was given a higher perceptual quality than the same video with a higher PSNR and a larger amount of layer variations.

Reconsidering the example from Section 2.6 polishing can be applied in two cases. In the first case, a connection to the server that stores the original version of the video object cannot be established but a cached version of the video object already exists on the local cache. Assuming that not all missing segments are retransmitted so far, polishing can be applied to reduce quality variations. In the second case, the storage space at the local cache is exhausted and new video objects should be stored on the cache. Instead of removing complete, less popular objects from the cache polishing can be applied to remove a certain amount of segments. Thus, new objects can be added to the cache's storage while only segments of certain objects are removed instead of complete objects.

In the remainder of this chapter an optimal polishing technique is specified and compared with a less complex heuristic by performing a series of simulations. In additional simulations the effect of using polishing for fine-grained cache replacement strategies is investigated.

6.2 Polishing and its Applications

In this section, the fundamental ideas behind polishing are presented. It is shown how polishing can be applied to increase the quality when an incomplete video object is transmitted from the cache to the client. In addition, a cache replacement method based on polishing is presented.

6.2.1 Transport

As mentioned above, the amount of variations in a layer-encoded video that is stored on the cache can be reduced by omitting the transmission of certain segments from the cache to the client.

The challenge of polishing is to identify segments that should not be transmitted in order to increase the perceived quality of a video object at a client. Polishing is a new technique that determines those segments that should not be transmitted from a cache to a client with the goal to increase the perceived quality of a video at a client. After the caching process of a certain video object is finished, the polishing algorithm (as described in Section 6.4) is executed. This algorithm identifies segments that should not be played out in subsequent streaming sessions from cache to clients. Note, that the identified segments are not removed from the cache. If this video object is requested, the transport mechanism of the cache will take into account the information gained by polishing and decide which data will be sent to the client and which not.

The information gained by polishing can, e.g., be used to stream a polished version of a video object if retransmission scheduling cannot be performed. This might be the case if the server or the link between the server and the cache is down or the server does not have additional capacity to allow retransmission in addition to already active streams. The general concept of polishing in such a case is shown in Figure 6.1.

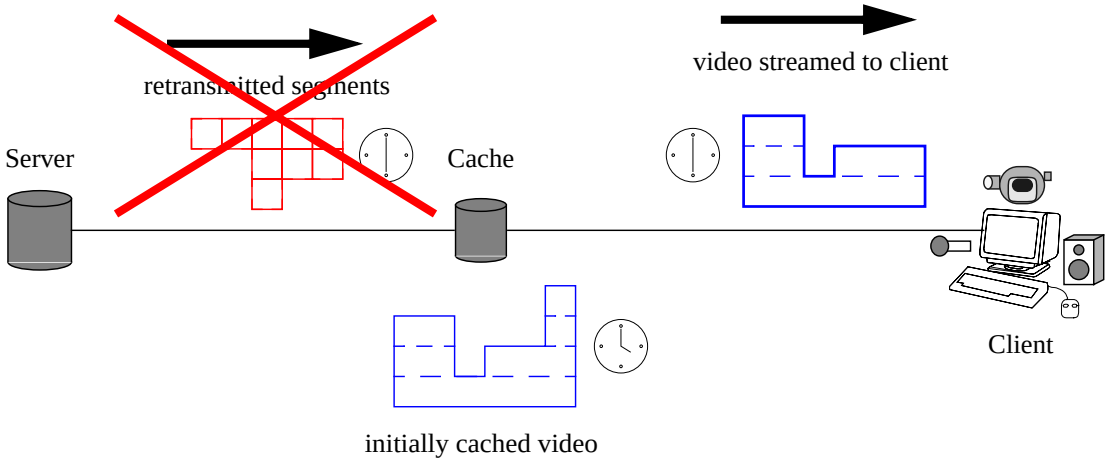


Figure 6.1: Polishing in the case of impossible retransmissions

6.2.2 Fine-grained Cache Replacement

Since polishing identifies segments that are of less importance in relation to quality, the information gained by applying this technique can also be used for cache replacement strategies. Assum-

ing that the storage space on a cache is exhausted and data has to be removed from the cache in order to allow the caching of a new object, segments identified by polishing can be deleted. A fine-grained replacement scheme based on segments can increase the efficiency of the cache as shown by [112]. If, in addition, popularity information is taken into account, e.g., segments of the least popular object are deleted first, the quality of the cached content is also based on its popularity. Figure 6.2 shows the basic principle of cache replacement with the aid of polishing. In this example, four objects are stored on the cache. After performing polishing, 15 segments are identified by the algorithm that can be removed from the cache and, thus, new storage space is created to store an additional object on the cache. The second alternative in Figure 6.2 shows the case in which popularity based caching is applied. The popularity of the cached object decreases from left to right influencing the number of segments that are identified for removal by polishing, leading to the fact that objects with a higher popularity are cached in a better quality.

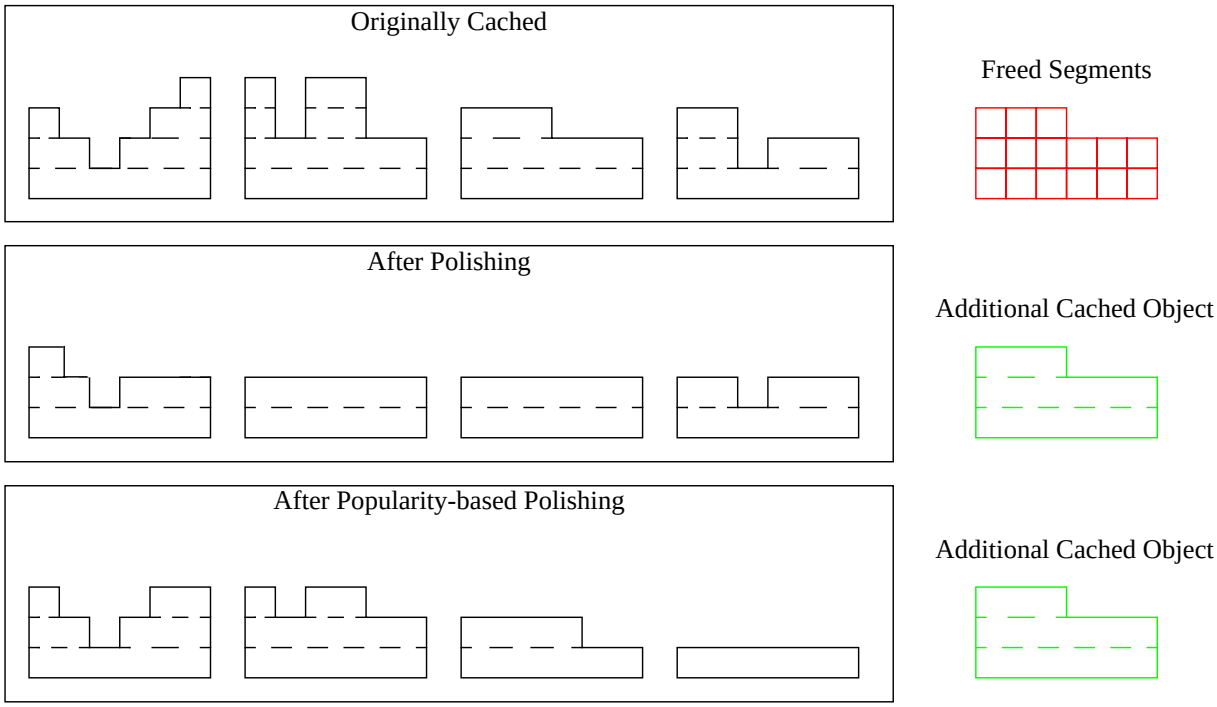
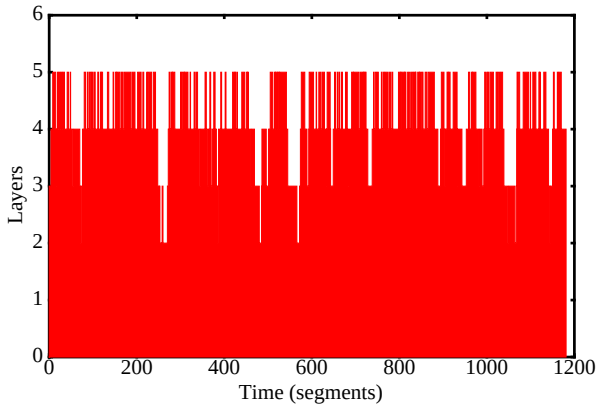


Figure 6.2: Cache replacement with the aid of polishing

6.2.3 The Spectrum in Combination with Polishing

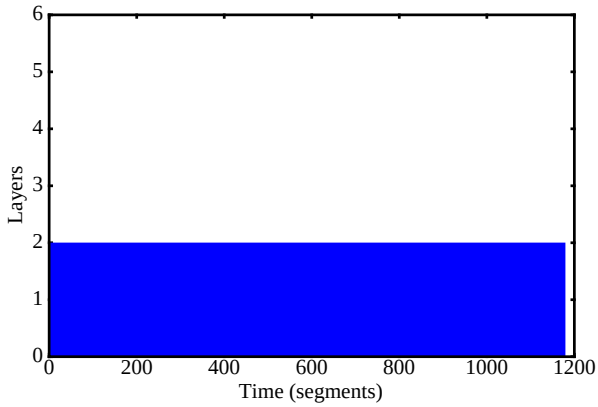
Although, the spectrum is a good metric for the quality of layer-encoded video (see Section 4.6) and, thus, in the case of retransmission scheduling the minimization of the spectrum is the goal to strive for (see Section 5.1.3) this goal cannot directly be applied for polishing. The spectrum becomes zero for the case that no layer changes occur, irrespective of how many layers the video object consists of. In the case of retransmission scheduling where new segments are added to the video due to the fact that any additionally available bandwidth is used for retransmissions, achieving a spectrum of zero always leads to a better quality than the one of the originally cached object.

In the case of polishing this assumption is not valid since segments are discarded from instead of added to the layer-encoded video. If the decision to drop certain segments would be solely driven by the spectrum, polishing could lead to the fact that all segments of incomplete layers are discarded. This effect, which is denoted as *over-polishing* (see Figure 6.3), is undesirable because it decreases the quality in a drastic manner. In Section 6.4, a new algorithm for polishing that avoids the problem of *over-polishing* is presented.



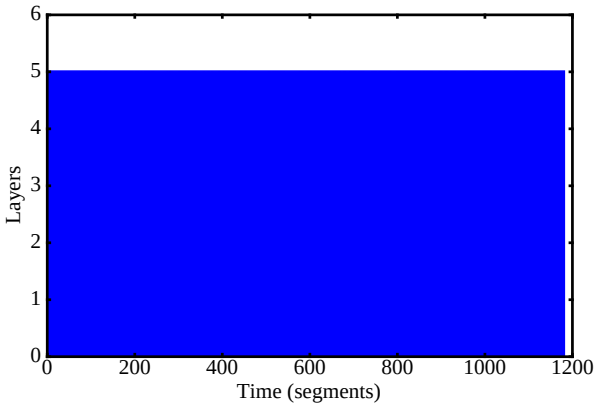
Originally cached layer-encoded video

(a)



Over-polished layer-encoded video (s(v) = 0)

(b)



Layer-encoded video after retransmission scheduling (s(v) = 0)

(c)

Figure 6.3: Polishing vs. retransmission scheduling

6.2.4 Example

In this section, a simple example to demonstrate the effect of polishing is given. It is assumed that a layer-encoded video is stored in the cache as shown in Figure 6.4. The variations in the number of layers are caused by a congestion controlled transmission between server and cache which results from the network conditions on the path between both (see Section 6.5 for details on how the congestion controlled transmission was simulated). Figure 6.4 also shows the layer-encoded

video as it would be transmitted to the client after the polishing algorithm has been performed. In addition, Figure 6.4 shows the result of a simple heuristic where only the highest (5th) layer is dropped. Further details about the optimal polishing algorithm are given in Section 6.4. Figure 6.4 shows a significant reduction in layer variations due to polishing. A reduced spectrum (28 compared to 193 of the originally cached object) indicates a better perceptual quality compared to the one of the originally cached object. The effect of *over-polishing* has been avoided. The example also shows the storage space that could be freed on a cache based on the information that is gained by polishing. In comparison, for the simple heuristic, that simply drops the top layer, the amount of storage space that can be freed due to optimal polishing is higher. A simulative investigation on polishing as a mechanism for cache replacement is given in Section 6.5.3.

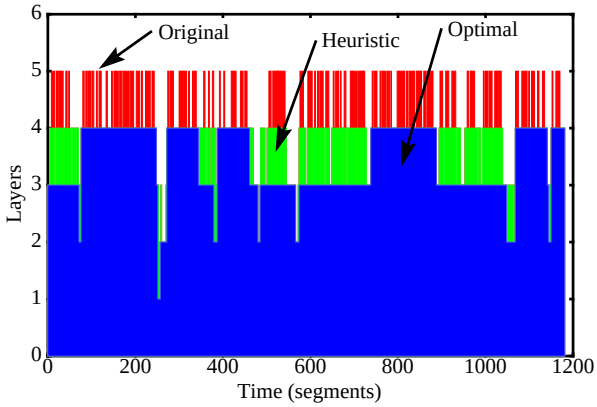


Figure 6.4: Comparison of originally cached and polished (heuristic and optimal) video object

6.3 Existing Work on Polishing

The work presented in this chapter is, to the best of our knowledge, the first investigation on this topic.

Rejaie et al. [109] introduce a fine-grained cache replacement mechanism that allows the deletion of single segments. Each layer of a video is regarded separately. Beginning at the top layer of a video, for each single layer segments are removed from end to beginning, while in our case the whole video is regarded for the removal of segments. Thus, in the case of Rejaie's approach segments of the top layer of the cached video are removed until none of the segments of this layer is left. If more space on the cache is needed, this process will be continued on the next lower layer. In the case of polishing segments from all available layers can be removed independently. The approach presented by Paknikar et al. [110] allows only the removal of complete layers which is somewhat similar to the heuristic presented in Section 6.5. Also in the analytical investigation performed by Kangasharju et al. [111] only complete layers can be dropped.

Quality based caching [146] is an additional approach for partial caching which assumes that metadata information about the quality of a scalable video is available. For example, the metadata would provide that removing the top layer of a 5 layer video would reduce the quality of the video by 20%. The authors leave open how this necessary metadata can be obtained.

6.4 Optimal Polishing

To polish a layer-encoded video means to minimize the spectrum while at the same time maximizing the number of segments played out to the client and could thus be regarded as a multi-objective optimization problem. Two characteristics of that optimization problem make it hard to be treated directly: on the one hand, the two competing optimization goals and, on the other hand, the non-linear quadratic form of the spectrum. Therefore, the spectrum is substituted by a new metric, namely layer variations, and a utility based approach where parameters for the relative weighting between the two competing goals of polishing are introduced.

Polishing - which under these prerequisites means maximizing the playback utility of a video - can be formulated as the mixed integer programming problem [147] given in Figure 6.6.

The two parameters u_l and p describe the utility of the video playout. u_l is the utility for receiving layer l (and all lower layers) in one period t. Obviously, the more layers are played back the higher the utility. p describes the utility loss for a layer change. By including u_l into the optimization process the *over-polishing* effect described in Section 6.2.3 is avoided. p prohibits quality loss by changing the playback layer too often.

The variable, h_t, contains the highest layer of the polished video at time, t, it can never be higher than the highest cached layer (see constraint (13) and Figure 6.5). The binary variable, z_t, is needed to account for layer changes in the target function. z_t is forced to one by constraints (11) and (12)when the highest layer of the polished video changes. The binary variable, b_{tl}, stores whether a layer, l, is included in the polished video in period, t, or not, constraint (14) expresses its relationship with the highest layer, h_t.

This problem can be solved with standard techniques like Branch and Bound and the Simplex algorithm [147]. We used the commercial mathematical programming solver Ilog CPLEX [148] to solve the problem.

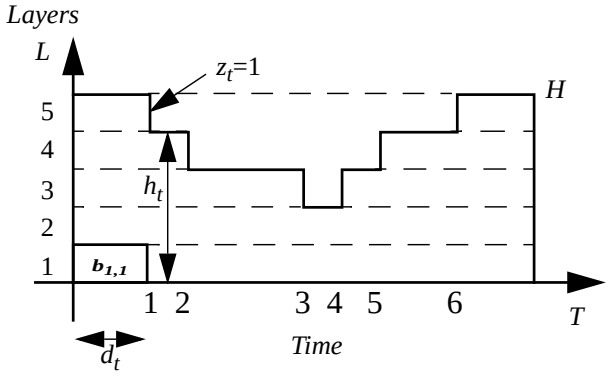


Figure 6.5: Cached layer-encoded video

6.5 Simulations

To verify if polishing is a valid approach and to obtain further information on the influence of the utility factors, u_l and p, a series of simulations were performed. An additional goal was also to

Indices:

$l = 1, \ldots, L$ -layer of the video

$t = 1, \ldots, T$ -period t

Parameters:

h_t^{cached} - number of the highest layer that is cached for period, t, all lower layers are cached in period, t, too

H - sufficiently large number ($H \geq L$)

d_t - length of period (in seconds)

u_l - utility of receiving layer, l, for one second (if video is played back for one second on layer 3 it generates a utility of $u_1 + u_2 + u_3$)

p - utility loss for a change in the number of layers that are played back

Variables:

h_t - the layer the video is played back in period, t

z_t - binary variable, one if a layer change occurs at the beginning of period, t, zero otherwise

b_{tl} - binary variable, one if video is played back in period, t,at layer, l, or higher, zero otherwise

Optimization problem:

$$Max\left(\sum_{l,t} u_l d_t b_{tl} - \sum_t p z_t\right) \quad (10)$$

subject to

$$h_t - h_{t-1} \leq H z_t \qquad \forall t = 2, \ldots, T \quad (11)$$

$$h_{t-1} - h_t \leq H z_t \qquad \forall t = 2, \ldots, T \quad (12)$$

$$0 \leq h_t \leq h_t^{cached} \qquad \forall t = 1, \ldots, T \quad (13)$$

$$l b_{tl} \leq h_t \qquad \forall t = 1, \ldots, T \quad \forall l = 1, \ldots, L \quad (14)$$

$$b_{tl} \in \{0, 1\} \qquad \forall t = 1, \ldots, T \quad \forall l = 1, \ldots, L \quad (15)$$

$$z_t \in \{0, 1\} \qquad \forall t = 2, \ldots, T \quad (16)$$

Figure 6.6: Optimization model

investigate how a simple heuristic performs in comparison to the optimal polishing algorithm. This heuristic simply drops one or more adjacent layers completely, beginning from the top.

The simulations are performed in the following manner. For each simulation an instance of a layered video on the proxy cache is randomly generated as described in Section 5.5. We use a discrete simulation time where one unit of time corresponds to the transmission time of a single segment. In Figure 6.4 (Original), an example video instance generated in this way is given. On each instance of a layer-encoded video created as described above our polishing algorithm is performed. The polishing algorithm was implemented using the mathematical programming solver Ilog CPLEX [148]. Before and after polishing the spectrum of the video is calculated in order to obtain information about the quality change. An example of such a simulation step is shown in

Figure 6.4 (before and after polishing) with the following set of parameters: $u_1 = 1$, $u_2 = 1$, $u_3 = 1$, $u_4 = 1$, $u_5 = 1$ and $p = 8$.

6.5.1 Utility Parameters

To obtain better insights in the influence of the parameters, u_l and p, a series of simulations with varying values for those parameters were performed. The results of this simulation are presented in Figure 6.7 and Figure 6.8. For each parameter set 100 video objects were randomly created and polished as described above. The average spectrum and the average total amount of segments were calculated before and after running the polishing algorithm. Figure 6.7 shows the results for

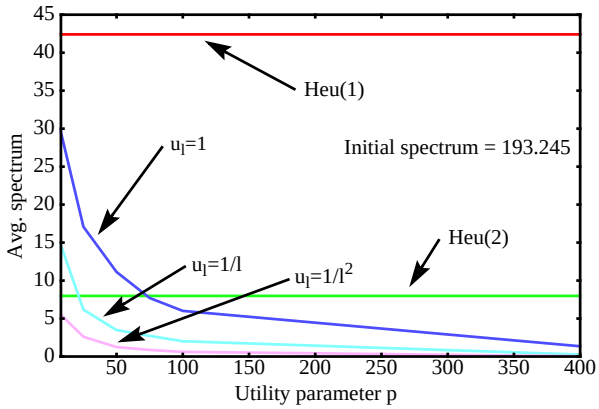


Figure 6.7: Average Spectrum

three different simulations and the spectrum for two versions of the heuristic. In the first version the top (Heu(1)) layer is dropped and in the second the two top (Heu(2)) layers are dropped. Then the average spectrum and average amount of segments of all 100 resulting objects are calculated. As can be derived from both figures, the heuristic has the disadvantage that it is static while, in the case of optimal polishing, the selection of the parameters, u_l and p, influences spectrum and amount of segments of the polished video object. On the other hand, the heuristic is simple and can be applied with little computational effort and the obtained results are fairly close to the ones of the optimal polishing.

In the case of polishing, the different simulations were performed with $u_l = 1$, $u_l = 1/l$, and $u_l = 1/l^2$ respectively.

In this specific simulation the heuristic turns out to be an alternative compared to the optimal polishing. A closer look at Heu(2) shows that the spectrum is significantly reduced while the amount of segments is reduced by a third. This result can also be achieved by applying the optimal polishing algorithm but at the price of a higher computational effort.

An additional series of simulations was performed with the difference that the initial videos used as input for the polishing simulation were generated in a different way. In this case TFRC traces generated by a ns2 simulation were used to generate the initial video. The bandwidth information generated by the ns2 simulation is used to determine how many layers of a video can be transmitted during a certain period. This complies with a cached layer-encoded video that has

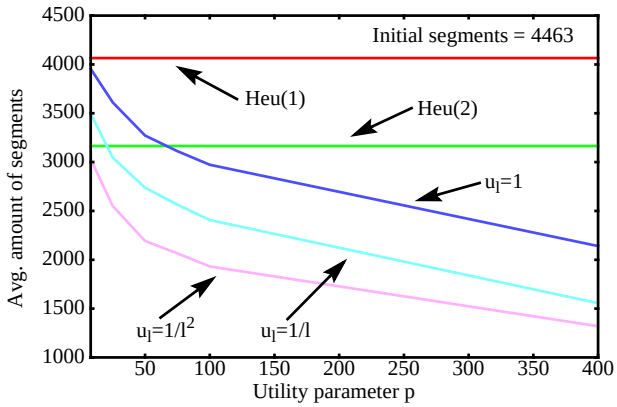


Figure 6.8: Average amount of segments per video object

been transmitted via TFRC. An example transmission is shown in Figure 7.1. This procedure is described in more detail in Section 7.2.3.

Figure 6.9 depicts the resulting average spectrum obtained by the application of the heuristic and the optimal polishing algorithm. Compared to the results of the simulation where the videos are generated randomly, the results of optimal polishing are similar. Comparing the results for Heu(2) (Figure 6.7 and Figure 6.9) with each other shows that the reduction in the spectrum is not as high with the TFRC based simulation. This difference is caused by the fact that TFRC also starts with a slow-start (identical to TCP) and, for a short while until the occurrence of a loss event, the sending rate can become quite high. Thus, reducing the top or the two top layers might affect only a small portion of the video. In simulations presented in the following section it is shown that this static behavior of the heuristic can reduce the efficiency of the heuristic.

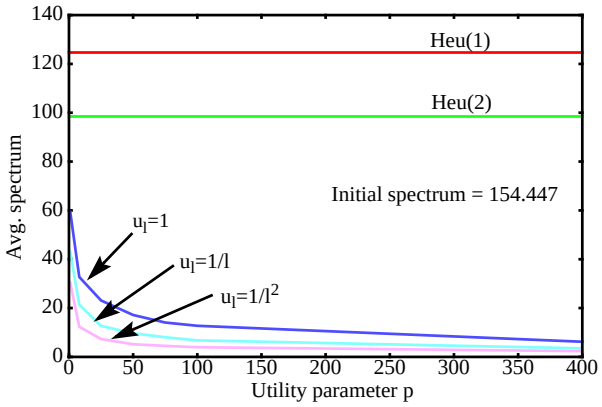


Figure 6.9: Average spectrum (TFRC based)

The difference in the average amount of segments for the random (Figure 6.8) and TFRC-based (Figure 6.10) simulation is caused by the fact that the TFRC traces are shorter in duration. The length of such a trace is equivalent to 400 time units while in the case of random based simulation the layer-encoded video has a length of approximately 1200 time units. Nevertheless, the behavior

of the optimal polishing against u_l and p is almost similar for both simulation types. The average amount of segments decreases with an increasing p and a decreasing u_l.

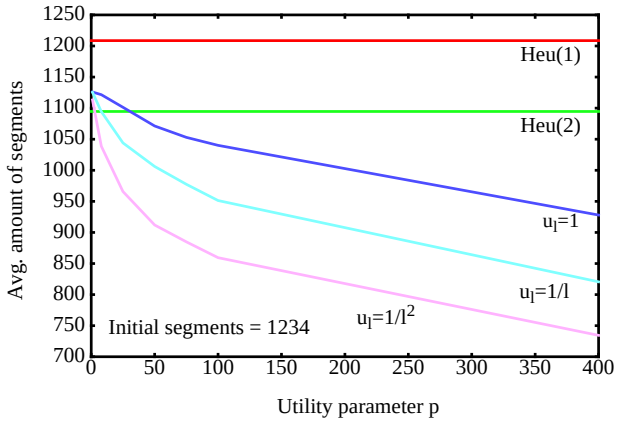
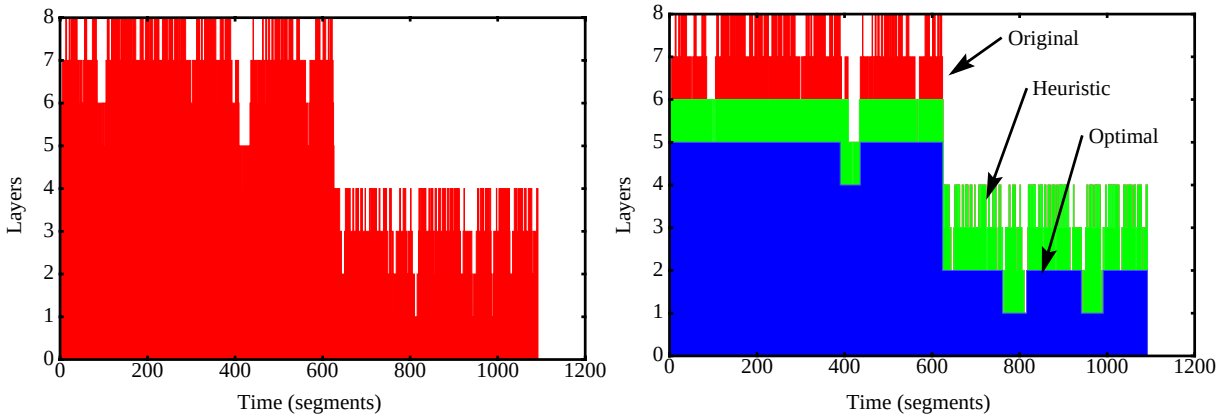


Figure 6.10: Average amount of segments per video object (TFRC based)

6.5.2 Polishing Layer-Encoded Video with Different Quality Regions



Originally cached layer-encoded video (a)

Originally cached and polished layer-encoded video (b)

Figure 6.11: Polishing for two-staged layer-encoded video

An example for a video object, which consists of two different quality regions, is shown in Figure 6.11. There can be several reasons that can cause the creation of such a video object on the cache. One possibility is a limited bandwidth between server and cache caused by competing traffic. In the given example the transmission of the competing traffic started after half of the video is already streamed from the server to the cache. The occurrence of two major quality regions in a cached video is rather arbitrary, since several of such *regions* with different quality levels can occur depending on the situation on the path between server and cache. Nevertheless, the chosen example is sufficient to demonstrate the drawback of the heuristic presented above. As can be

seen in Figure 6.11 (b), the disadvantage of the heuristic is the fact that only the region with the better quality (higher number of layers) is polished. In this example the two top layers are dropped. This effect does not occur with optimal polishing where both regions are polished. Applying the heuristic might be annoying for the viewer. The already existing quality decrease between the two regions is even intensified by the high number of quality changes in the second half of the video. With optimal polishing, quality variations are reduced in both regions and, thus, the quality decrease between the two regions is not that intense. Comparing the two resulting spectra (see Table 6.1) of the second region demonstrates the effect mentioned above. The spectrum in the second region is significantly higher for the heuristic than for optimal polishing, while for the first region both spectra are identical.

Table 6.1: Spectrum per region

	Region 1	Region 2
Heuristic	6	86.81
Optimal Polishing	6	1.2

Also for the case of the layer-encoded video consisting of two quality regions a series of simulations was performed. The simulation environment is identical to the one described in Section 6.5 with a slightly modified creation process for the initially cached video that results in quality variations as shown in Figure 6.11 (a). Similar to the simulations described in Section 6.5.1, 100 video objects are randomly created and the average spectrum and total amount of segments are calculated. Figure 6.12 shows the average spectrum for the optimal polishing depending on the parameters, u_l and p. The resulting spectra for the three variations of the heuristic are always larger than the ones for optimal polishing.

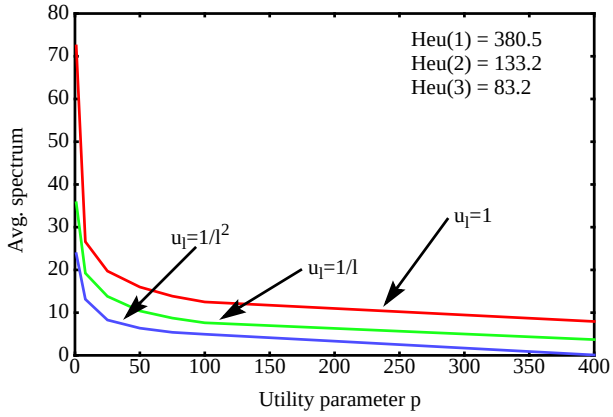


Figure 6.12: Average Spectrum

The comparison of the average amount of segments per video object (see Figure 6.13) shows that the three variations of the heuristic result in higher values than the optimal polishing. An interesting case is the one for $u_l = 1$ and $p < 50$ where the spectrum for the optimal polishing is

lower than the one of Heu3 but the amount of segments is larger. This is a different result to the one presented in Section 6.5.1 where the spectrum for the optimal polishing is always higher than the one for Heu2, if the amount of segments is larger for optimal polishing than for Heu2. The result of this simulation shows the advantage of the optimal polishing compared to the heuristic in the case that the cached video consists of regions with varying quality levels. This advantage becomes even more obvious if one imagines cached video objects which consist of more than two regions with different quality levels. The polishing effect (i.e. reducing the amount of layer variations) would probably only occur in the region with the highest quality level, while the remaining regions would remain unpolished.

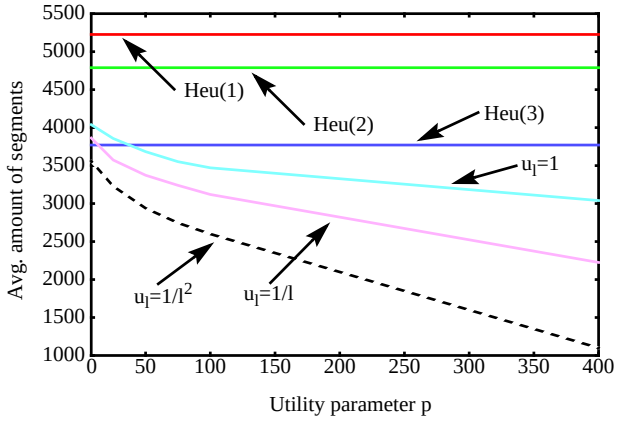


Figure 6.13: Average amount of segments per video object

6.5.3 Replacement

The goal of an additional set of simulations was to investigate how polishing can be used for cache replacement. This simulation is thought to demonstrate how useful polishing can be for cache replacement. In Section 6.5.4 an additional simulation is presented that takes also popularity information of the single video objects into account for the cache replacement.

At the beginning of the simulation it was assumed that the cache was initially filled with 50 unpolished video objects, consuming all of the cache's storage space. Then, 50 additional objects should be incrementally, i.e., one per time slot, stored on the cache. To generate the additional storage space for these objects in the first step the already cached objects are polished. If the space gained by polishing is not sufficient or in the case that all objects are already polished, a cached object will be removed in FIFO manner. Figure 6.14 gives an overview of how this simulation was performed.

This simulation was performed for the parameter set $p = 50$ and $u_l = 1$. In an additional simu-

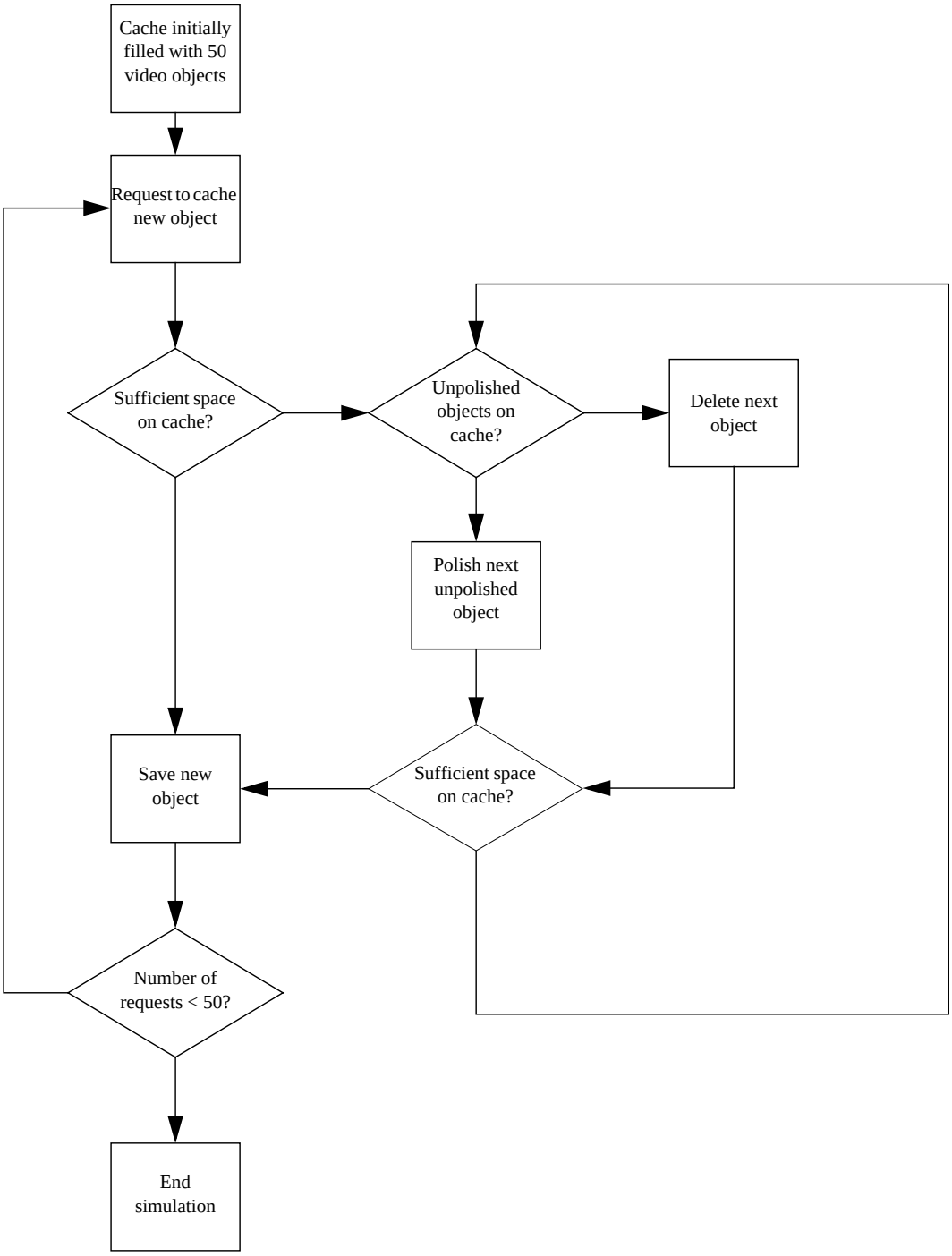


Figure 6.14: Simulation procedure

lation (no-polishing), objects are not polished but simply removed. The cache replacement simulation was also performed for the two versions of the heuristic. Figure 6.15 depicts the spectrum for each of the simulations, while the average amount of segments per object is shown in Figure

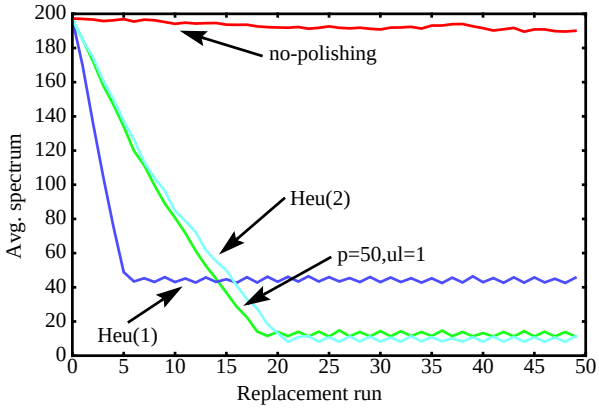


Figure 6.15: Spectrum for cache replacement

6.16. An interesting result is revealed by the comparison of both graphs for the case $p = 50$ and

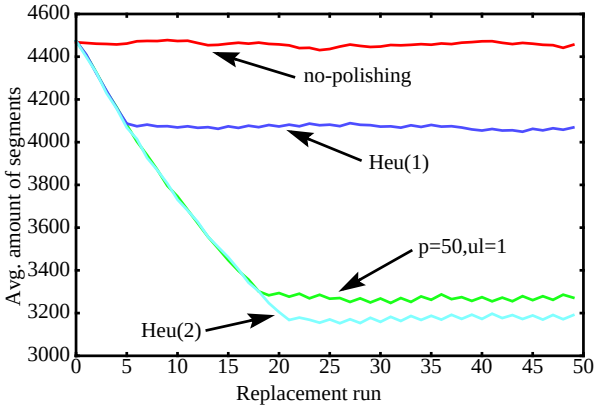


Figure 6.16: Average amount of segments per object for cache replacement

$u_l = 1$, where a high reduction of the spectrum results in a moderate reduction of the average amount of segments per cached object. Compared to a cache replacement which does not incorporate polishing the total amount of video objects stored on the cache is higher in the polishing simulations (see Table 6.2). Thus, integrating polishing into the cache replacement method is

Table 6.2: Total amount of objects

Simulation	no-polish	p=50,u_l=1	Heur(1)	Heur(2)
Objects	50	68	55	70

beneficial since a higher amount of video objects can be cached and layer changes are reduced (smaller spectrum). Based on the parameters that can be chosen for the polishing method, the behavior of the cache replacement method can be influenced. For example, a cache operator can control, with the aid of this parameters, if more objects in a lower quality or vice versa should be cached. Also in this simulation, the results of the heuristic are close to the optimal polishing result.

6.5.4 Popularity-based Cache Replacement

The goal of this simulation is to investigate whether optimal polishing can be applied to polish cached videos based on their popularity. That means, less segments are deleted from popular videos while the amount of deleted segments increases for less popular objects. Thus, the quality of the cached object is directly related to its popularity. In contrast to the aforementioned approaches polishing is not performed individually, but the complete content of the cache is regarded and polished according to the popularity of each single object and the amount of space that should be freed. This problem can be, similar to the problem presented in Section 6.4, formulated as mixed integer problem as shown in Figure 6.18.

Compared to the model presented in Section 6.4, two additional parameters are introduced: the popularity of each video object, w_v, and the total capacity of the cache's storage that the already cached objects can consume, K^{max}. The latter allows one to determine how much storage space should be freed on the cache to allow the caching of new video objects.

The simulation for this investigation was changed in comparison to the one presented in Section 6.5.3. Here, one can specify the amount of cache space that should become available for the caching of new data, $K^{total} - K^{max}$. In addition, each video object is assigned a certain popularity w_v. Figure 6.17 shows the originally cached and the resulting polished video object for the video with the highest (a) and the lowest (b) popularity on the cache. In this case, ten video objects are stored on the cache and 25% of the total cache space is freed by polishing the cached videos according to their popularity. For the case of the most popular video object 20% of the original segments is deleted, while for the least popular video object 31% of the original segments is removed from the cache.

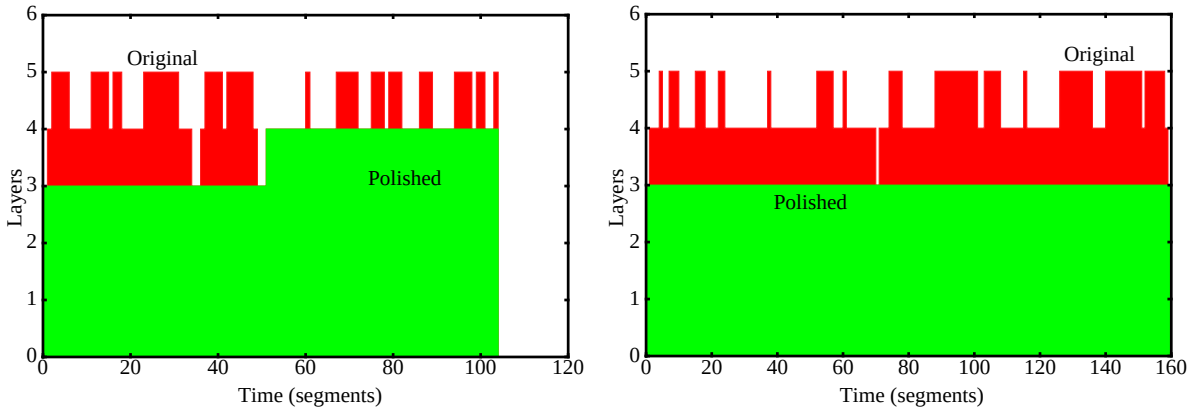


Figure 6.17: Polishing for most and least popular video object

This simulation was performed 20 times. For each single simulation the initial cache state was randomly generated. Table 6.3 shows the amount of segments (in percent) that was removed from each of the ten cached video objects. Objects which are shaded equally were assigned the same

Indices:

$v = 1, \ldots, V$ -number of video object

$t = 1, \ldots, T$ -period t

Parameters:

h_t^{cached} - number of the highest layer that is cached for period, t, all lower layers are cached in period, t, too.

H - sufficiently large number ($H \geq L$)

K^{max} maximum capacity available for already cached video objects

w_v - popularity of the video object

u_l - utility of receiving layer, l, for one second (if video is played back for one second on layer 3 it generates a utility of $u_1 + u_2 + u_3$)

p - utility loss for a change in the number of layers that are played back

Variables:

h_t - the layer the video is played back in period, t

z_t - binary variable, one if a layer change occurs at the beginning of period, t, zero otherwise

b_{tl} - binary variable, one if video is played back in period, t,at layer, l, or higher, zero otherwise

Optimization problem:

$$Max \sum_v \left(\sum_l \sum_t w_l u_{lt} b_{vtl} - \sum_t p z_{vt} \right) \quad (17)$$

subject to

$$h_{vt} - h_{vt-1} \leq H z_{vt} \quad \forall t = 2, \ldots, T \; \forall v = 1, \ldots, V \quad (18)$$

$$h_{vt-1} - h_{vt} \leq H z_{vt} \quad \forall t = 2, \ldots, T \; \forall v = 1, \ldots, V \quad (19)$$

$$0 \leq h_{vt} \leq h_{vt}^{cached} \quad \forall t = 1, \ldots, T \; \forall v = 1, \ldots, V \quad (20)$$

$$l b_{vtl} \leq h_{vt} \quad \forall t = 1, \ldots, T \; \forall v = 1, \ldots, V \; \forall l = 1, \ldots, L \quad (21)$$

$$b_{vtl} \in \{0, 1\} \quad \forall t = 1, \ldots, T \; \forall v = 1, \ldots, V \; \forall l = 1, \ldots, L \quad (22)$$

$$z_{vt} \in \{0, 1\} \quad \forall t = 2, \ldots, T \; \forall v = 1, \ldots, V \quad (23)$$

$$\sum_v \sum_t b_t h_{vt} \leq K^{max} \quad \forall t = 1, \ldots, T \; \forall v = 1, \ldots, V \quad (24)$$

Figure 6.18: Cache replacement optimization model

popularity value. The popularity is the highest for objects 1, 2, and 3 while it is the lowest for objects 7, 8, 9, and 10. The popularity for 4, 5, and 6 lies in between the other two groups. The results of this simulation show that with the extended polishing algorithm a very fine granular cache replacement can be achieved. With this algorithm it is possible to free cache space for new content while data from already cached content is removed according to the popularity of the con-

tent. In this specific example 25% of the caches' storage space is available for the caching of new content while none of the cached objects had to be removed completely.

Table 6.3: Average amount of removed segments

Video Object	1	2	3	4	5	6	7	8	9	10
Average amount of removed segments in %	18	14	18	26	26	25	32	31	32	31
Avg. spectrum of polished object	5.6	4.3	5.3	3.1	4.0	5.4	2.2	2.0	2.8	2.4
Avg. spectrum of unpolished object	18.1	19.3	19.9	23.3	32.2	26.7	23.8	21.3	23.9	25.8

6.6 Summary

In this chapter, a new technique, called polishing, is presented which can be either applied for the streaming of data from the cache to the client or as a cache replacement mechanism. Polishing makes use of the fact that a reduction in layer variations can also be achieved by not transmitting certain segments, although this means that some of the data available at the cache is omitted from the client. Optimal polishing, which means to maximize the playback utility of a video, is formulated as a mixed integer programming problem. Since optimal polishing is dependent from several parameters, a simulative investigation is performed to gain better insights into the influence of these parameters. In this simulation, the results of optimal polishing are compared with a simple heuristic that drops certain layers completely. The results show that the heuristic can, in specific cases, achieves similar results as optimal polishing with less computational effort. Yet, simulations with layer-encoded video that consists of different quality regions demonstrate the drawback of the heuristic. In addition, to the parameter analysis, two supplementary simulations on cache replacement were performed. The first one is rather simple and does not take the popularity of single video objects into account. With this simple cache replacement method it can be achieved that a higher overall amount of objects can be stored on the cache while, in parallel, layer variations are reduced. For the second kind of simulation on cache replacement the optimal polishing mixed integer problem is extended to vary the intensity of polishing based on the popularity of the video object. The results of this simulation show that by applying the extended optimal polishing storage space on the cache can be freed while the amount of segments is reduced according to the popularity of the single objects. To abstract, polishing is a valid means to reduce variations in layer-encoded video either, if retransmission from the cache cannot be performed or as a cache replacement mechanism.

Chapter 7: Fair Share Claiming

7.1 Motivation

In Chapter 5, different retransmission scheduling algorithms that meet users' demands to watch high quality video with relatively little quality variations were developed and compared with each other. In this chapter, the focus is on how these scheduled retransmissions can be combined with a TCP-friendly transmission method by claiming the fair share for the TCP-friendly session. Transmitting a layer-encoded video in a TCP-friendly manner does not always result in the case that the session claims its fair share of network resources as will be shown in Section 7.2. Therefore, a mechanism is proposed, called fair share claiming (FSC), which combines the transmission of a layer-encoded video and some additional data, resulting in the utilization of the fair share a session is entitled to. The applicability of FSC is investigated based on a simulation and compared with the results from Chapter 5, while also the scheduling heuristics from Chapter 5 were applied. Additionally, an implementation design for FSC, that makes use of already existing protocols for video streaming, is given.

7.2 Performing TCP-friendly Streaming in Combination with Retransmissions

In this section, the fair share claiming mechanism which makes use of the additional bandwidth that is not claimed by the layer-encoded video without breaking the cooperative rules implied by TCP's resource allocation model is presented. After a detailed description of the FSC mechanism in Section 7.2.1 existing work on FSC is presented in Section 7.2.2. The simulation environment for FSC is presented in Section 7.2.3 and finally the simulation results are shown in Section 7.2.4.

7.2.1 TCP-friendly Streaming

First of all it should be mentioned that is was not the goal to develop new TCP-friendly mechanisms for streaming. In recent years several protocols for the transport of non-TCP traffic with TCP-friendly congestion control were developed. An overview about these protocols is given in Section 3.3.2. For the work on FSC it was decided to use TFRC [48] for several reasons.

It is a rate based congestion control protocol with good TCP-friendliness. The main advantage in combination with A/V streaming is the fairly smooth rate in the steady-state case and, therefore, applications that rely on a fairly constant sending rate are supported. In addition, the protocol is end-to-end which does not require any modifications to the network infrastructure. Transmitting a layer-encoded video with the maximum rate that TFRC would allow does not always make sense. If, e.g., the possible transmission rate would be much higher than the actual rate needed for

the video, the receiver might need a large buffer to store segments until their playout time is reached. Especially in the case of mobile receivers like, e.g., handhelds buffer size might be a scarce resource. Since one of the major goals of the SAS architecture is to support the heterogeneity of the Internet, FSC is a well suited mechanism to fit that requirement as is shown in the following. In Chapter 8, it is shown how FSC can be applied in the case of TCP-friendly transmission between server and cache and an uncontrolled transmission (pure UDP-based) between cache and client.

Rate changes in TFRC will not always result in a rate change for the layer-encoded video because the encoding format provides only a certain number of different layers resulting in a finite amount of possible transmission rates. This can result in a situation where the actual possible transmission rate (determined by the TFRC algorithm) and the rate constituted by the sum of several layers might differ. The following example was chosen to illustrate the problem in more detail: for example, a layer-encoded video that consists of up to three layers, each requiring a constant transmission rate of 0.5 Mbit/s that should be transmitted in a TCP-friendly manner via TFRC. At a certain point in time during the transmission the TFRC algorithm determines a maximum possible transmission rate of 1.3 Mbit/s. This would allow a transmission of two layers of the layer-encoded video whereas 0.3 Mbit/s would be wasted if the video would not be transmitted faster than necessary and an additional third layer cannot be transmitted. The additional bandwidth is the fair share that may be claimed by a corresponding TCP session, yet due to the inelastic and discrete nature of layer-encoded video it cannot be claimed. Nevertheless, finding some data to fill this gap would allow the stream to claim its fair share without breaking the cooperative rules implied by TCP's resource allocation model. Figure 7.1 depicts an example TCP-friendly layered video transmission.

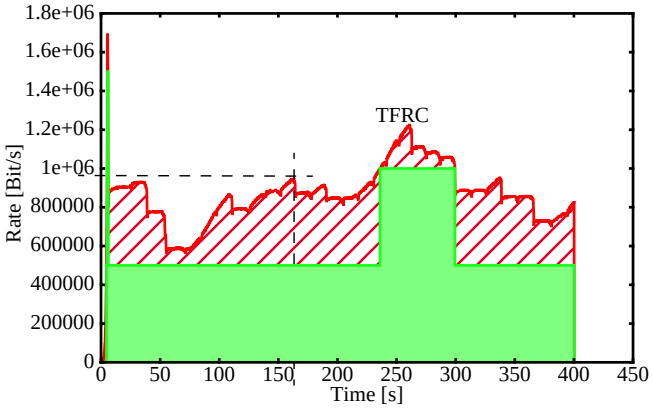


Figure 7.1: Example layer encoded video transmission via TFRC

The creation of the TFRC trace is explained in more detail in Section 7.2.3. In this example scenario the transmission rate for the layer-encoded video is only increased in case that the rate determined by the TFRC mechanism would allow the transmission of an additional layer. If this is not the case the additional bandwidth (marked hatched in Figure 7.1) could be used for the transmission of additional data. In the example shown in Figure 7.1 that would be an additional 142 MByte. Here, the focus is especially on the retransmission of missing segments of the video that

is currently streamed or videos that are already (but not completely) stored on the cache to claim the fair share for this TCP-friendly session (see Figure 7.2). These techniques are referred to as *in-band* FSC for the former case and *out-of-band* FSC for the latter. In Chapter 5, different retransmission scheduling algorithms that could be used to determine which of the missing segments should be transmitted were already devised and analyzed. The following simulation shall shed some light on whether the combination of both techniques (retransmission scheduling and FSC) is an appropriate method to improve the quality of a cached layer-encoded video.

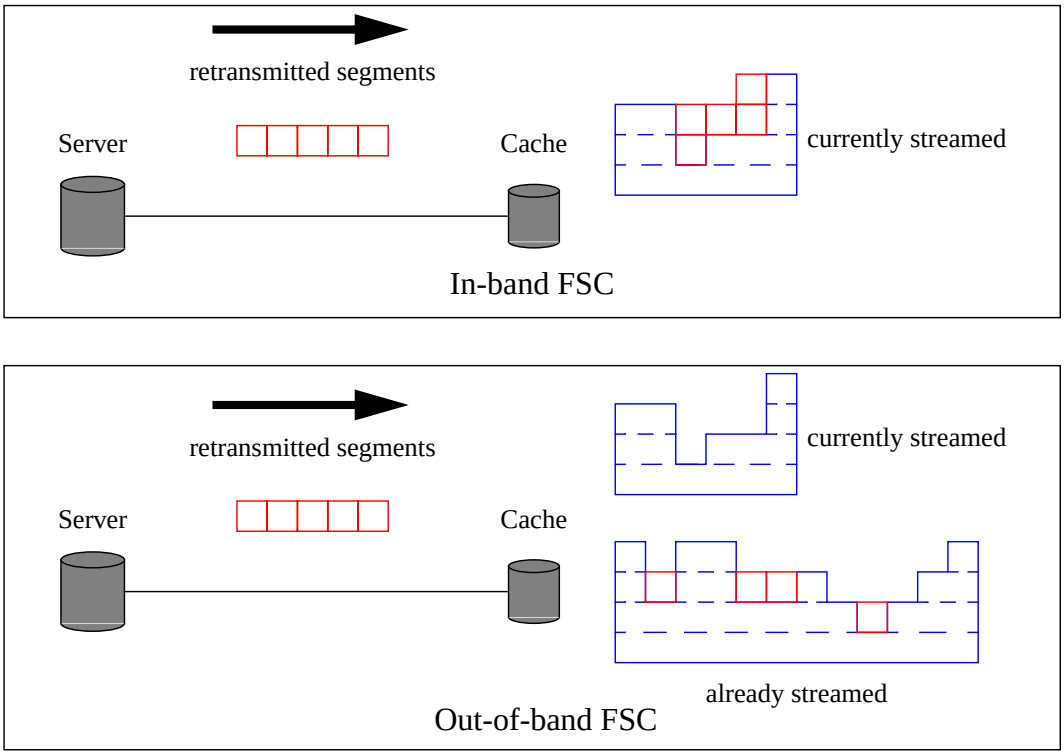


Figure 7.2: In-band and out-of-band FSC

7.2.2 Existing Work on FSC

Rejaie et al. [90] and Saprilla et al. [93] also present mechanisms that claim their fair share and support the transport of layer-encoded video. Both assume that the client has sufficient buffer to allow a transmission rate higher than the receivers consumption rate. While the first approach is limited to Constant Bit Rate (CBR) encoding the second also supports Variable Bit Rate (VBR) layered-encoding. In contrast to the FSC mechanism presented here video transmission into caches is not considered. Both mechanisms do not support the transmission of data that has already missed its deadline for the timely consumption at the client and therefore do not offer any functionality to improve the quality of a video that is being cached or already stored on a cache.

Another approach that supports scalable streams is presented by Law et al. [149]. In their work the focus is mainly on server efficiency and scalability. In comparison to the FSC approach the

quality is adapted due to the capabilities of the receiving client rather than to network conditions. Their architecture does neither envision caches nor incorporate TCP-friendly streaming.

[150] considers the combination of caching and layered video, yet, the latter only for the support of heterogeneous clients but not for congestion control purposes. Furthermore, the emphasis of their work is on optimal cache replacement decisions viewed over *all* videos stored in a cache. For FSC, however, a two-stage decision process is assumed where in the first stage a video is selected for storage in a cache and then the retransmissions of missing segments are scheduled independent from the cache status of other videos. While this represents a restricted problem it ensures that the overall problem still remains manageable. Another difference in their work is the fact that missing segments of a certain layer are only streamed directly to the client in contrast to the FSC approach where the segments are transmitted to the cache to achieve a quality improvement for more than one client.

7.2.3 Simulation

The simulation is split into three single steps in order to create a scenario that represents the mechanisms presented in Section 7.2.1 and to keep the simulation environment more generic:

A) Creation of a TFRC trace.

B) A possible layer-encoded video transmission that is derived from the TFRC trace.

C) Determination of segments that can be retransmitted due to spare bandwidth.

Each single part is explained in more detail in the following sections.

A) Creation of TFRC Traces

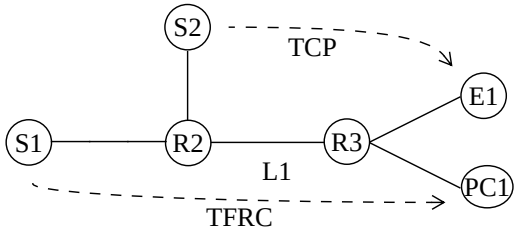


Figure 7.3: Simulation

The TFRC traces are created with the aid of the network simulator ns2 [151] because a TFRC model is already included and ns-2 allows us to create traces that can be used as a basis for the second simulation step. With the simulation configuration shown in Figure 7.3 a scenario for the distribution techniques described in Section 7.2.1 is modeled. The simulation consists of 2 routers, 2 senders and 2 receivers. The routers, R1 and R2, are connected via a duplex link (L1) having a bandwidth capacity of 15 Mbit/s and a delay of 100 ms. This represents a scenario that consists of a server S1 and a cache PC1 that caches video streams and forwards them to requesting clients. To model competing web-like traffic between S2 and E1 we use ON/OFF sources as proposed by Floyd et al. [48]. In addition, there is a TCP session between S2 and E1 which serves as a reference in order to observe the TCP-friendliness of TFRC. It is active throughout the entire

simulation. The ON/OFF sources are also enabled during the whole simulation. One long-lasting TFRC stream, representing the layer-encoded video transmission, is initiated at simulation start. A single simulation lasts for 400 seconds. The trace shown in Figure 7.1 was generated with the method described here.

B) Layer-encoded Video Transmission

In the simulations, layer-encoded video that can consist of up to three layers is assumed. In addition, all layers are of equal size, CBR encoded and, therefore, require an identical transmission rate, which is 0.5 Mbit/s for these simulations. To create a layer-encoded video transmission a TFRC trace created by the methods described above is used as the starting point. A small C++ program was implemented that scans the bandwidth for each entry of the TRFC trace and determines the number of layers that can be transmitted based on the TFRC bandwidth. The rate for the layer-encoded video transmission in Figure 7.1 was generated in this way. It must be stated here that the strategy for increasing or decreasing one of the layers is very simple. In particular, a more intelligent strategy might also contribute to a smoother transmission of the video. During the execution of this program, an additional list is built to store information about the spare bandwidth that is available for the transmission of additional data. For the example shown in Figure 7.1 (on page 112) at 163.2 seconds a spare bandwidth of 452000 Bit/s would be determined. The simulation is discrete since the TFRC trace has a resolution of 0.2 seconds. This restriction had to be made to keep the overall simulation effort in reasonable limits. The error that is introduced by this simplification is negligible since the delay for the link between R1 and R2 is 100 ms and therefore the RTT is at least 200 ms thus leading to the fact that two consecutive rate changes are never less than 200 ms apart.

C) Retransmission

To investigate retransmission scheduling in layered video caches in more detail a simulation environment was built and presented in Section 5.5. In contrast to the simulations presented here, an instance of layer-encoded video was created randomly. The available bandwidth for retransmissions was constant for a single simulation and was only modified to compare the behavior of the retransmission scheduling algorithms in relation to different amounts of available bandwidth. For the simulations presented in this section the bandwidth for retransmissions can change in each step of the simulation. For this reason the simulation environment had to be changed in two respects:

- For each step in the simulation the available bandwidth for the retransmission of missing segments must be calculated. This is performed with the aid of the list generated by the simulation tool described above that contains information about the available bandwidth for retransmissions at a certain point in time.
- If retransmissions are performed for the simultaneously streamed video the retransmission scheduling algorithm can only consider the already transmitted part of the video. This is in contrast to our earlier work where we assumed that the complete instance of a cached layer-encoded video is known.

7.2.4 Simulative Experiments

Two different kinds of simulations as described in Section 7.2.3 were generated, one for in-band and one for out-of-band FSC. The latter were performed to compare the unrestricted retransmission scheduling algorithm (see Section 5.3) against the window based algorithm presented in [109] (For easier identification of both algorithms we refer to them as *unrestricted* and *restricted*, respectively). Since the latter looks always a certain amount of time ahead of the current playout to determine segments for retransmission, it is not applicable for in-band FSC.

A) In-band FSC

The result of a single in-band FSC simulation is depicted in Figure 7.4. It shows how the quality of a layer-encoded video on a cache can be improved with the aid of the FSC technique. In this specific scenario it was possible to add an additional layer for more than half of the length of the complete video. Thus the next client requesting this video from the cache will have the chance to receive it in a significantly better quality than the first client. Unfortunately, there is a small gap for layer 2 between the 200th and 250th second that decreases the quality of the cached content. One possible solution to close this gap would be the usage of the out-of-band FSC technique (during the transmission of some other video). To reduce the amount of layer changes the caching strategy on the cache might decide to delete the short amount of the third layer that was cached due to the peak of the TFRC around 5 seconds after the transmission started, e.g., by applying polishing as presented in Chapter 6.

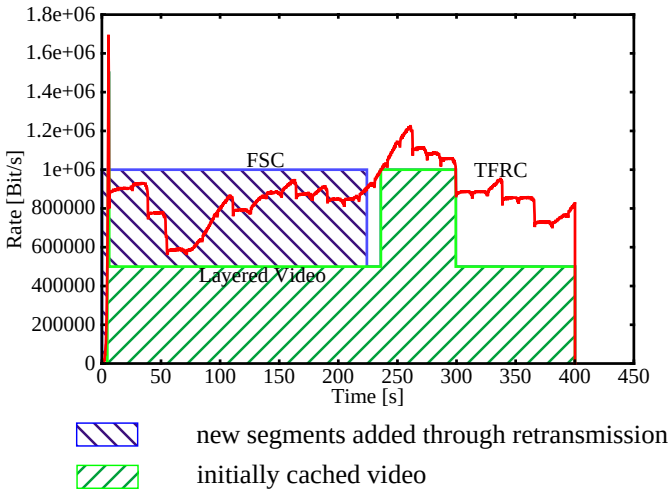


Figure 7.4: Result of an in-band FSC simulation

For the complete in-band FSC simulation 100 different TFRC traces are created as described in Section 7.2.3. For each one of this traces the additional available capacity for retransmissions is calculated. Thus, an average capacity of 5.15×10^8 Bit is available which is equivalent to one single layer of 370 seconds in length.

Figure 7.5 shows the average spectrum for the three cache centric heuristics. The outcome of

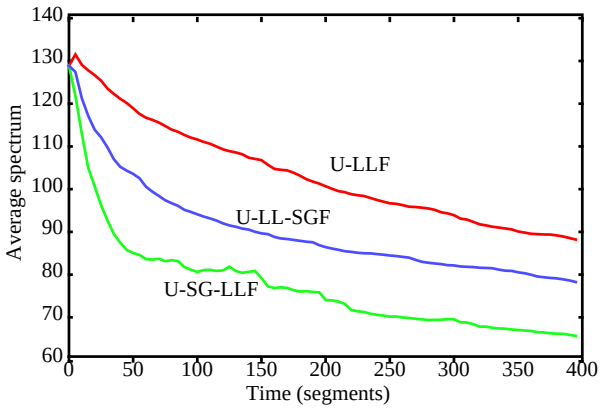


Figure 7.5: Average spectrum for 100 simulations

this simulation confirms the results of the simulation presented in Section 5.6 where a constant available bandwidth for the whole simulation was assumed. Again, U-SG-LLF is the best performing followed by U-LL-SGF and U-LLF. An interesting detail is that all three spectra are monotonic decreasing. This is caused by the fact that in some cases only a small amount of bandwidth is available for retransmissions and, thus, gaps will not be closed completely or, even worse, segments of layers that were not cached at all are retransmitted. The latter increases the amount of layer changes which leads to an increased spectrum. This short term increase should be accepted to allow a quality improvement of the cached video in the long run.

B) Out-of-band FSC

The out-of-band FSC simulation was performed in a slightly different way than the in-band simulation. This is due to the fact that segments for an already cached video are retransmitted. To be able to compare the results of this simulation with the results of the in-band simulation it is assumed that the cached video has the same layout as the layered video trace in Figure 7.4. That is, the initial transmission is identical to the one of the in-band simulation but without any retransmissions for this specific video. This allows the comparison of the quality improvement between the in-band and out-of-band technique in the unrestricted case. Since the “layout” of the video is now completely known both algorithms, restricted and unrestricted, can be applied. A second TFRC trace is generated which determines how much additional bandwidth is available for the retransmission of missing segments. The result of this simulation which is depicted in Figure 7.6 clearly shows the disadvantages of the restricted algorithm. Caused by the fact that only missing segments ahead of time from the actual playout point are regarded for retransmission only small chunks of the missing segments can be retransmitted (the black boxes in Figure 7.6 only appear as boxes due to the low resolution of the plot). The problem of the restricted algorithm is shown in more detail in the magnified part of Figure 7.6 that represents an enlarged part of the out-of-band FSC simulation for this algorithm. The high frequency of layer changes is very annoying for the client currently watching the video (see the results presented in Chapter 4). In contrast to the restricted algorithm the result of the unrestricted algorithm is that the quality of the video is

enhanced by one layer in one contiguous segment which in this specific case does not lead to additional layer changes compared to the initially cached video. The difference in the amount of retransmitted segments in comparison to the in-band simulation (see Figure 7.4) is caused by the fact that the amount of spare bandwidth that is available for retransmission is higher in the in-band case. That is, the rates of the second TFRC trace allow the retransmission of a larger amount of segments as it was the case for in-band FSC.

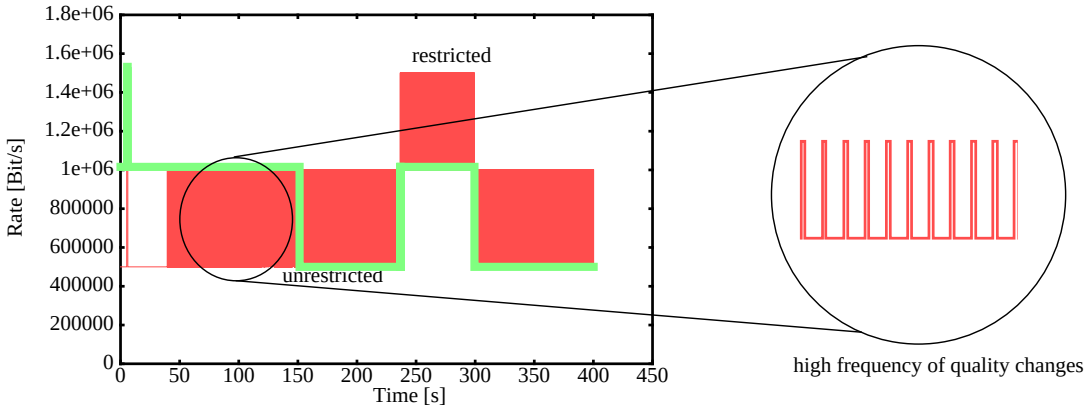


Figure 7.6: Out-of-band FSC simulation

To exclude isolated effects that could occur by only performing a single simulation, 100 simulations were performed based on the TFRC traces already created for the simulation described above. In Figure 7.7, the results of the average viewer-centric unrestricted and the restricted heuristics (see Section 5.4) are shown. Also here the drawback of the restricted heuristic (W-LLF) becomes quite clear; the spectrum is constantly increased caused by the effects shown in Figure 7.6. This is the only case where the results from the simulations presented in Section 5.5.1 cannot be affirmed. Thus, the amount of additional bandwidth that is available for retransmissions influences the performance of the W-LLF heuristic. The results for the viewer-centric unrestricted heuristics confirm the results of the simulation presented in Section 5.5.1.

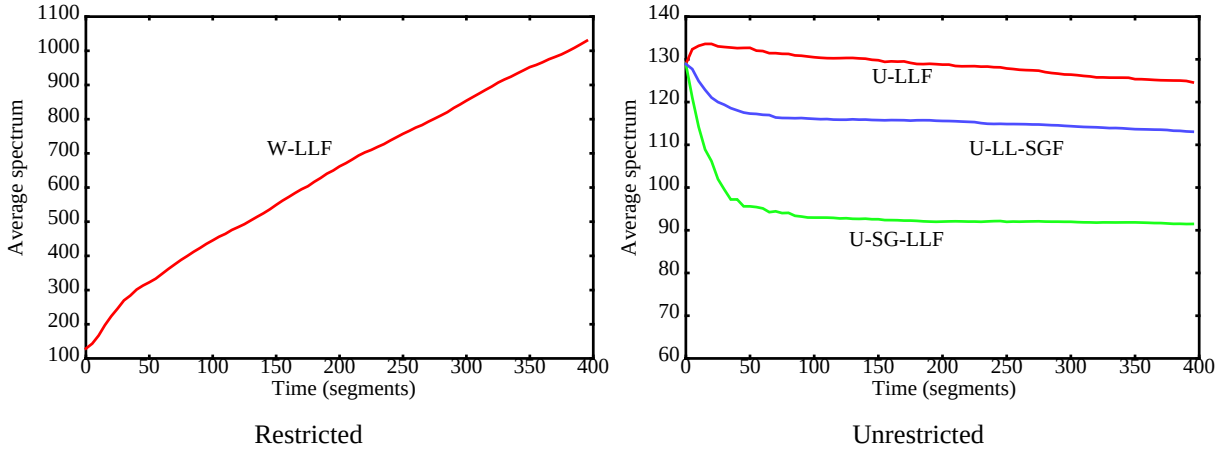


Figure 7.7: Average spectrum for 100 simulations (restricted and unrestricted viewer-centric)

C) Influence of the Total Number of Layers

It is obvious that the amount of bandwidth that is available for retransmission decreases with an increasing number of layers caused by a higher adaptiveness to the bandwidth determined by TFRC. An additional simulation should investigate the dependency between the number of layers of a layer-encoded video and the resulting amount of bandwidth for retransmissions. In the simulation the same TFRC trace is used for each single step and the number of layers is varied between 2 and 20. Increasing the number of layers does not increase the maximum bandwidth of the layer-encoded video, rather the bandwidth of each single layer is decreased. This had to be done to be able to compare the results of each single simulation. Figure 7.8 presents the result of the simulations which shows the percentage of the overall capacity of the TFRC trace that can be used for retransmissions. The result states our assumption that the additional capacity will decrease with an increasing layer granularity. Yet, even with a high number of layers there is still capacity for retransmissions available.

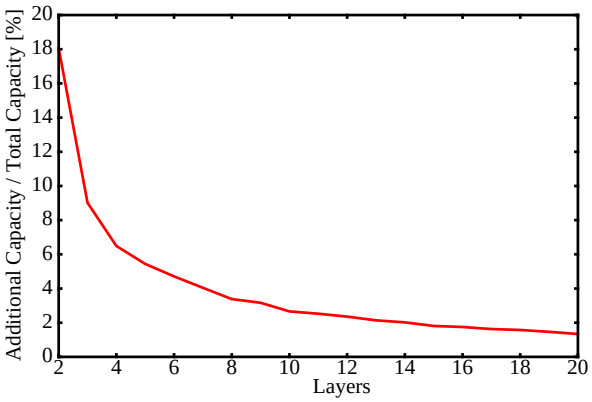


Figure 7.8: Relative additional capacity for retransmissions

The correlation between the number of layers and the additional capacity for retransmissions is shown in Figure 7.9 for a single TFRC transmission.

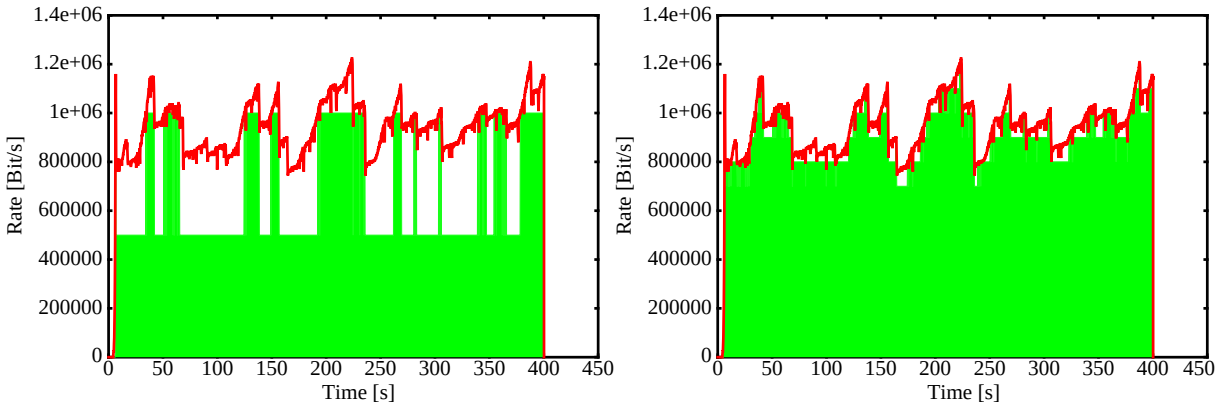


Figure 7.9: Comparison between 2 and 20 layer video

7.3 Implementation Design for FSC

In this section, the design for an implementation of the FSC technique which should be based on already existing, standardized and if possible well established protocols and techniques is presented. However, that has not always been possible which is mainly due to the fact that the proposed TCP-friendly mechanisms require significant changes of the protocol design. Nevertheless, the design should only require modifications at server and cache to allow standard clients in this architecture. In general, a distinction between the transmission of time critical data (the actual stream that is transmitted) and time uncritical data (segments for retransmission) is made. It is the overall goal of this section to show that FSC can be reasonably integrated in streaming applications.

7.3.1 Protocol Suite

The most common approach for audio and video streaming is the usage of RTP[1] over UDP as transport protocols. It is well known that this approach lacks an appropriate congestion control mechanism and might cause problems like congestion collapse if the amount of audio and video streams further increases. That is exactly why different variations of TCP-friendly protocols have been developed that should avoid the occurrence of such problems in the best-effort Internet. As mentioned above, TFRC is one of these protocols and in Section 7.2.1 it is already stated why it is favored as a TCP-friendly protocol for streaming environments. Another advantage is that the TFRC mechanisms can be integrated into the RTP protocol and, thus, the introduction of a completely new protocol in the streaming protocol suite is not necessary. This integration has the additional benefit that no modifications to UDP must be made and therefore possible kernel modifications can be avoided. To enable TFRC functionality in RTP some new header information is needed (see Figure 7.10) and part of the overall protocol behavior must be changed. Fortunately, two of the additionally needed header fields are already contained in the RTP header, *sequence number* and *time stamp*. The additional fields shown in Figure 7.10 must be placed in the RTP extension header. The receiver reports needed by TFRC can be transported by the *application specific information* in the RTCP receiver reports. The frequency of RTCP receiver reports must be highly increased since TFRC requires these reports to be sent every RTT. Since only unicast transmission is envisioned so far in the SAS architecture, the higher amount of reports should neither restrict the raw data transmission nor cause an ACK implosion.

Rather complicated is the identification of missing segments of a layer-encoded video that should be retransmitted. One could imagine that missing segments could be easily identified by the RTP sequence number but this is not true in every case. The sequence number of an RTP packet would only be helpful if the data would be stored as RTP packets on the server's disk, because the simple information of a sequence number would not be sufficient to identify the related part of, e.g., a file that contains an MPEG-1 video where the packet length can vary (wire format and storage format must not necessarily be identical). In the case of LC-RTP the loss rec-

[1] For reasons of simplicity only RTP is mentioned but always the combination of RTP and RTCP is meant.

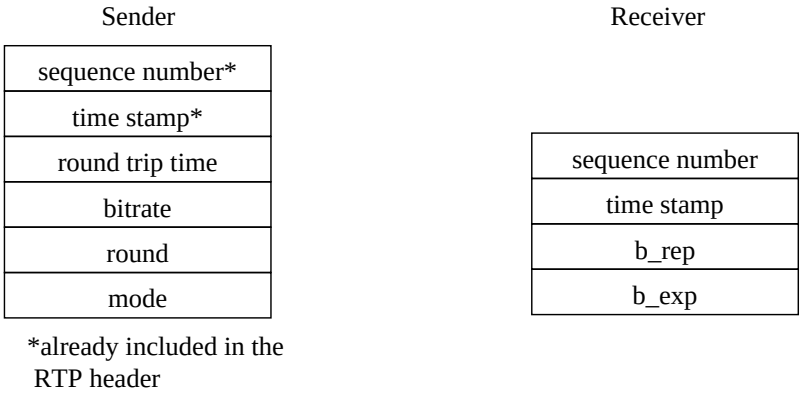


Figure 7.10: Additional TFRC header fields

ognition is realized by a *byte count* which is included in each RTP header. The byte count represents the actual byte position of the data that is included in the RTP packet. Each server implementation has to transform the byte count value into its own file indexing information. As a consequence it is possible to have different file layouts on sender- and receiver-side. For example one server implementation stores the file as raw data and another stores some header information with it. A possible way of inserting the byte count into the RTP header and not into the payload is the use of the extension header of RTP. With the aid of the byte count losses can be exactly identified: the receiver can maintain a list of losses; and the lost segments can be requested from the sender at another point in time. In Section 8.4.1, an RTP payload format is presented that also allows the receiver to detect which segments of a layer have been lost or were not transmitted at all due to network congestion. If lost segments should be retransmitted during the streaming session the RTCP application specific header can be used to send the loss lists from receiver to sender. Should the retransmission be performed out of band, a TCP connection would be sufficient to transmit the loss lists to the sender. Since there exist now two cases that require an RTP extension header we propose that in the case of FSC the RTP protocol should be used with a modified extension header as shown in Figure 7.11.

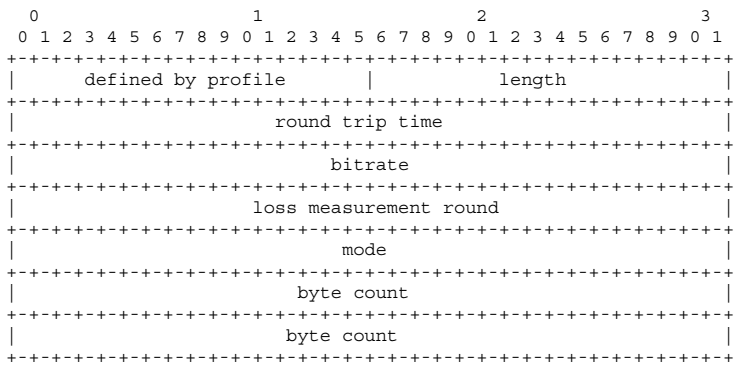


Figure 7.11: RTP header extension for congestion controlled streaming

An additional issue is the multiplexing of the initial video stream and retransmitted segments over one RTP session. In this case we can make use of RTP's mixing functionality. Originally this functionality was thought to combine RTP streams from different senders at a router into one RTP

stream. Here, this functionality is used in a slightly different way: in this scenario no physically separated senders exist but the layer-encoded video and the packets that should be retransmitted can be regarded as two logical sources. Thus both streams[1] can be transmitted via one RTP stream whereas each stream is assigned a different *synchronization source identifier* (SSRC). This technique allows the RTP receiver to correctly identify each of the two streams and forward the packets to their correct destination. It might also be possible to mix more than two streams with this mechanism but this is out of scope for this work. To identify each SSRC correctly the receiver needs additional information about the mapping between streams and SSRCs. The mapping information can be signaled to the receiver with the aid of the *private extension source description* (SDES) item of an RTCP source description packet. This type of RTCP packet contains a list of SSRCs and according SDES items. The *private extension* item is meant for experimental or application specific use. The SDES *private extension* consists of an *item identifier, length information, prefix length, prefix* and *value string* (see Figure 7.12).

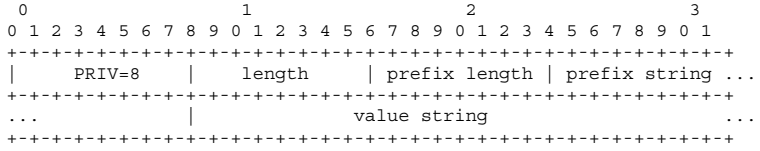


Figure 7.12: Private extension SDES item

The *prefix string* for this specific SDES item will be set to **FSC** to indicate that this information is related to the fair share claiming technique. For the *value string* three different strings are defined so far:

Table 7.1: Value string parameters

Value string	Description
STREAM	SSRC represents layer-encoded video stream
INBAND	SSRC represents a stream for retransmitted segments that belong to the in parallel streamed layer-encoded video
OUTBAND	SSRC represents a stream for retransmitted segments that belong to an already cached video

This additional information allows the demultiplexing of the single sessions of an RTP stream and their correct assignments to instances for further data processing. In Section 7.3.3, it is demonstrated how a correct data path could be established with the aid of the *Stream Handler* (SH) [152] architecture.

[1] To simplify description the retransmission of segments is also described as a stream, although this is not technically correct.

7.3.2 Retransmission Signaling

As mentioned above it might be possible to perform in-band and out-of-band retransmission with FSC. With out-of-band retransmission the respective video is already stored on the cache and one run of the retransmission scheduling algorithm should be sufficient to generate a retransmission list. A simple TCP transmission from receiver to sender to send the list of ordered missing segments should be sufficient. The sender stores this list and in the case of a retransmission request uses this list to obtain information which segments of the original video should be retransmitted.

In the case of in-band retransmission the retransmission signaling must be handled in a different way. First of all the video is not entirely transmitted to the cache. The retransmission scheduling algorithm can only make decisions based on the already received part of the video. Thus the generated list of segments that should be retransmitted might change over time and updates of the list that exists at the sender must be performed. To be able to perform this update the initial list that is created by the retransmission scheduling algorithm should also be stored on the receiver. Each time the algorithm is performed again, the newly generated list and the stored list should be compared. If the differences reach a certain threshold value (for a suitable metric that measures similarity between loss lists) a new list must be transmitted to the sender.

7.3.3 Stream Handler Extension

Our experience with the implementation of streaming applications showed us the need for a generic architecture to handle continuous media streams. This became specifically clear during the development of our experimental KOMSSYS [152]. The platform is used for investigations on A/V distribution systems and, therefore, has to offer support for different encoding formats and transport protocols, but also distribution mechanisms under investigation. Such distribution mechanisms may combine unicast and multicast distribution or may apply segmentation and reordering for efficient delivery. During the initial implementation phase we quickly realized that a monolithic approach would not allow a simple integration of these new distribution mechanisms. This led to our decision to build an environment that is based on a stream handler (SH) architecture (see Appendix C).

In this section, the extensions are defined that must be made in the stream handler architecture to support the FSC technique. In the experimental VoD platform server, cache and client make use of the stream handler architecture. Therefore, a great deal of stream handler modules have already been designed, implemented and tested. To support FSC in the streaming platform it should be tried to reuse these elements and if necessary extend them or create new stream handlers. First of all the modifications that have to be made to the already existing SHs are shown and then the design of the new SHs is presented.

A) RTP/RTCP and LC-RTP/LC-RTCP

The RTP and RTCP functionality is combined in the RTP SH. The FSC technique requires extensions and modifications to both RTP and RTCP as described in Section 7.3.1. For the multiplexing and demultiplexing functionality the RTP SH must be able to receive data from more than one

upstream SH (sender) and forward it to more than one downstream SH (sender) in the case of out-of-band retransmission. This is not the case for in-band retransmission, since the byte count information of LC-RTP defines the position of the retransmitted segment explicitly. Although only LC-RTP is used in the FSC case, the extension should be made for both protocols since RTP is a basis for LC-RTP and the new functionality might also be needed by RTP only. This allows a separated development of RTP with TFRC mechanism and LC-RTP.

On both, the sending and receiving parts, extensions to provide TFRC functionality must be made, too. New fields must be added to the extension header as depicted in Figure 7.11 (on page 121). The changes for RTCP are extensive since on one hand the format of the RTCP receiver reports and on the other hand the timing for the transmission of these must be changed. The TFRC specific information should be transported in application-specific information of an RTCP receiver report and should contain the four fields shown in Figure 7.10 (on page 121). The timing for the receiver reports must be changed in a way that it is based on the RTT information instead of the algorithm proposed in [61].

B) Packetizer and Depacketizer

Several profiles for the transport of standardized audio and video formats in RTP exist [62]. So far no profile for the transport of layer encoded video is defined, caused by the lack of a standard for this technique. Experiences with the development of (de-)packetizers for several audio and video formats [152], have shown that building new SH for this purpose is a rather straightforward task. Depending on what layer encoded video techniques should be supported, it might be necessary to build more than one SH. Due to the lack of a standardized layer-encoded video format a solution that makes use of a pseudo-layered format is presented in Chapter 8.

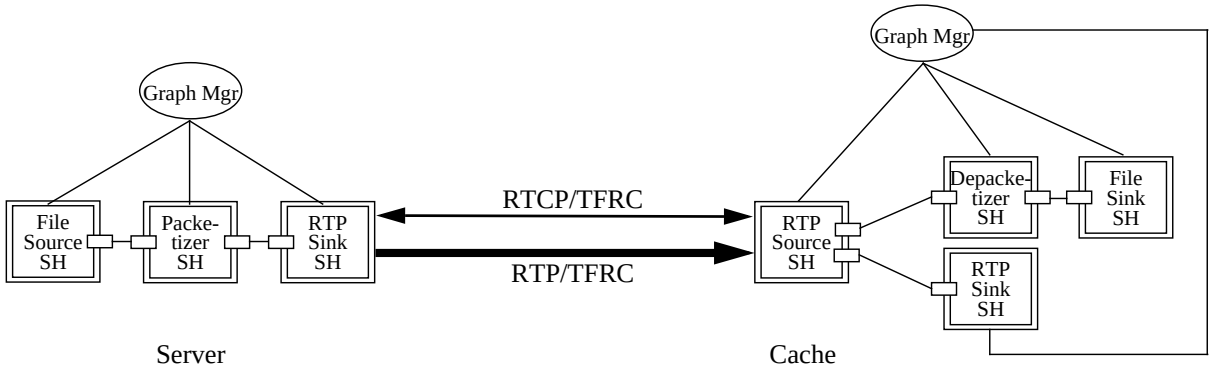


Figure 7.13: Stream handler scenario for in-band FSC

To describe the interaction between the SHs in more detail an example scenario for in-band retransmission is shown in Figure 7.13. Whenever an RTP packet should be sent out on the network the payload will be pulled from the Packetizer SH. With the report functionality of the SH architecture the RTPSink SH can inform the Packetizer SH about the actual transmission rate. With the aid of this information the packetizer can build the two different types of payloads. For the layer encoded video the rate information also determines the number of layers that should be

transmitted. If the number of layers is known the packetizer can determine the resulting capacity that is available for the retransmission of segments and determine which type of packet (layer encoded video or retransmitted segment) will be handed to the RTP Sink SH in the case of a pull request from the latter. With each payload the packetizer must also provide the appropriate SSRC information which allows the correct demultiplexing in the RTP Source SH at the receiver. Further information about the stream handler architecture is given in Appendix C.

7.4 Summary

In this chapter, a new technique called FSC that allows one to claim the fair share for TCP-friendly sessions for the transmission of layer-encoded video in the case that caches are involved is presented. This technique bears the advantage that on the one hand these sessions actually will get their fair share of the link and on the other hand the quality of already cached video can be improved. In order to prove the applicability of FSC a simulation environment that consists of three single simulation steps is created. The single steps include: creation of TFRC traces, layer-encoded video transmission, and retransmission scheduling. A series of simulations based on this simulation environment is performed. The results of the simulations state the applicability of FSC, especially, that in combination with the retransmission scheduling algorithms developed in Chapter 5 a reasonable quality improvement for already cached video can be achieved. It is also shown that another already existing retransmission scheduling algorithm is not well suited for the proposed FSC technique. The simulation results also state that the results of the initial simulations made in Chapter 5 where in contrast to this chapter the initial video objects were created randomly and not based on TFRC traces.

Since the results of the simulations met the expectations, the design for the integration of FSC in an already existing streaming platform is given. An extension to RTP/RTCP is proposed which allows the usage of the TFRC mechanism and, thus, a TCP-friendly streaming between sender and receiver. Additionally, signaling extensions for RTCP are made which allow the demultiplexing of the data that belongs to the actual stream and the data that is retransmitted. For the integration of the FSC mechanism only existing protocols have to be extended, it is not necessary to create new protocols.

Chapter 8: Scalable TCP-friendly Video Distribution for Heterogeneous Clients

8.1 Motivation

In recent years the variety of different types of clients with access to the Internet increased. Only a few years ago, the typical Internet client was a standard PC connected via LAN or modem, but today Internet clients are also set-top boxes, handhelds, mobile phones, or even game consoles and the number of wireless clients is increasing rapidly. Their characteristics in terms of computing power, memory space, and access bandwidth vary greatly, thus, leading to new challenges for a video streaming architecture.

Layer-encoded video ideally supports such an architecture since it allows an adaptation to link bandwidth, client processing power and buffer size, which is not possible with non-adaptive formats for example, MPEG-1. A cache can be introduced as a node in a distribution system to address several of the problems associated with this heterogeneity. It separates the long-distance network from the access network and their distinct characteristics concerning throughput, jitter, and loss. In this chapter, the focus is on the ability to apply, conditionally, congestion control mechanisms separately for the server-cache and the cache-client link. For example, it is possible to stream from server to cache with higher bandwidth than from cache to client.

Basically, the goal of the work described in this chapter is to experimentally investigate the applicability of different transport mechanisms in combination with caches, heterogeneous clients, and layer-encoded video. The setup for the experiment reflects a typical scenario for a SAS architecture.

In Chapter 7, it was the goal to integrate an optional congestion control mechanism into the existing streaming system without loosing efficiency and compatibility. In the following, an extension that allows the streaming of a pseudo-layered format and supports standard clients which have no extended signaling capabilities is presented. Based on this implementation, several experiments to prove the applicability of the proposed SAS architecture (see Chapter 2) were conducted. First, it is shown how controlled packet dropping can result in a smooth transport of the most relevant layers of a layer-encoded video to a client. Second, the effects of an uncontrolled access network are shown and compared with the controlled approach. Both experiments are performed in a testbed, as well as in the Internet. The experiments also show that a sustainable number of layers can be determined easily by applying the congestion control mechanism of TFRC [48] on the access link.

8.2 Transport Scenarios

In this section, the different transport scenarios supported by the SAS architecture are introduced. Caused by the ability of server, cache, and client to perform congestion control there are four possible cases how different transport mechanisms may be used on the link between server and cache and cache and client. Those four possibilities are illustrated in Figure 8.1.

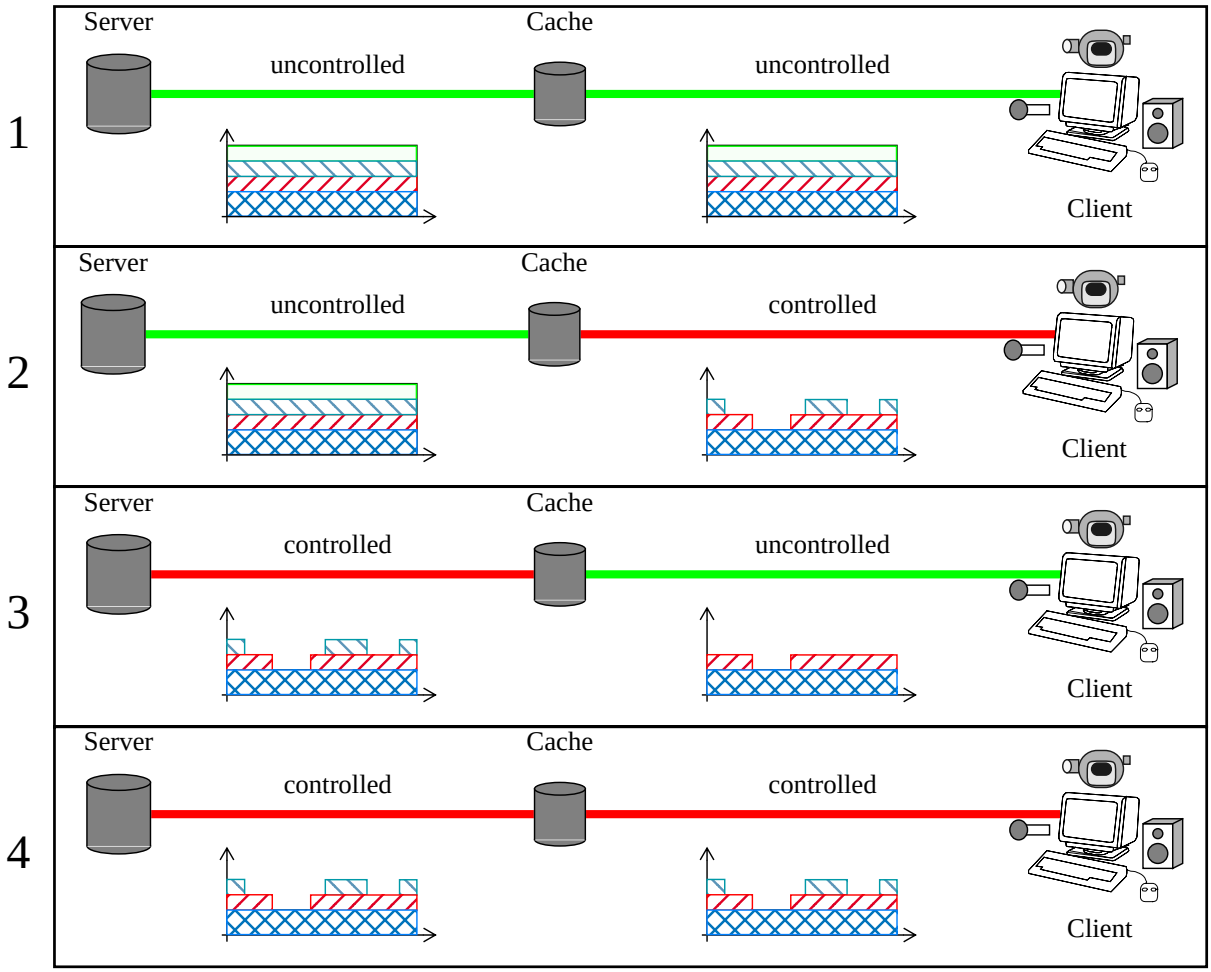


Figure 8.1: Combination of transport mechanisms

For each of the four scenarios an example is given which shows how a layer-encoded video could possibly be transmitted between server, cache, and client. For the sake of simplicity no losses are assumed on both links. Note that only scenario 1 and 3 allow the usage of standard RTP/RTSP clients. Scenario 1 allows the usage of standard RTP/RTSP server, cache, and client and represents commercial systems that are available today. In all other cases (2-4) at least one of the components has to be extended as described in the following. Scenario 2 shows an example in which data between server and cache is streamed uncontrolled while congestion controlled streaming between cache and client is performed. This scenario allows the usage of an unmodified server while cache and client must be able to perform congestion controlled streaming. In scenario 3, an example is given in which the client signals the maximum rate it can receive data via

RTSP to the cache. Since this maximum bandwidth allows only the transmission of up to two layers, segments from layer 3 are not forwarded from the cache to the client. Nevertheless, the video is streamed from the server to the cache with a higher rate in order to increase the quality of the cached video object. This approach is appropriate in cases where cache storage is sufficient and the possibility that other clients request this object in a better quality is high. The signaling of the maximum transmission rate by the client is necessary, since no congestion control is performed between cache and client and this is a possibility to prevent the cache from sending with a rate that cannot be consumed by the client.

In scenario 4, both links are congestion controlled thus, sending signaling information about the maximum rate from the client to the cache is not necessary.

8.3 Scalable Streaming Implementations

Due to the limited information about commercial products [57, 55, 153] the focus is on implementations created by the research community.

Rejaie et al. [112] mainly focus on the design and implementation of a cache and the goal to adaptively adjust the quality of a cached stream based on popularity and available bandwidth. The main difference to the SAS approach is the fact that clients in their architecture always have to be able to perform congestion control. In addition, the congestion control mechanism in [112] is not integrated in RTP but set on top of it. Another implementation of a TCP-friendly partially reliable video streaming approach is presented in [92] by Feamster et. al. No caches are envisioned in this architecture.

In [154], Race et al. present the implementation of a RAM based video cache which is designed for the caching of MPEG-2 streams. The usage of RAM instead of a hard disk circumvents a bottleneck on the disk's channel. DSM-CC is used for stream control and the streaming is performed in non-congestion-controlled manner via UDP.

8.4 Implementation

In the following, the design and implementation of the SAS architecture that is capable of supporting heterogeneous clients are presented. The implementation described here is based on the KOMSSYS [155] streaming system which was built during our research on wide-area distribution systems. In a basic version [152], it consists of server, cache, and client and is based on the standard protocols RTP/RTCP, RTSP, and SDP. To verify new ideas for scalable distribution systems it is constantly extended by new functionality and used for experiments. LC-RTP ([120] and Appendix A) and *Gleaning* [4] are two examples for new functionality that was integrated into KOMSSYS (see Appendix C).

In the remainder of this section, the modifications and new functionality for the data-path related and control-path related parts of the streaming system are presented. Finally, it is shown how all of the new pieces are orchestrated in the cache to offer the new functionality.

8.4.1 Data Path

A) TCP-friendly Congestion Control Mechanisms

In Chapter 7, it is already motivated why TCP-friendly rate control (TFRC) is well suited as congestion control algorithm for streaming in the SAS architecture. It is also presented how the integration of TFRC in RTP can result in a congestion controlled streaming mechanism.

B) Layer-Encoded Video Format (Layer Dummy)

The *layer dummy* (LD) format that is used for the experiments shown in Section 8.5 is presented in this section. Afterwards, the modifications that have to be made to perform lossless transmission into caches using layer-encoded video are shown. The decision to use the LD format is made because at first it should be investigated whether the modifications on the data and control path proposed here meet the expectations. In addition, the LD format makes measurements easier, since the specific RTP payload format allows an extensive logging (see Section 8.4.2). The way LD is designed allows an easy integration of layered formats like SPEG [80]. Thus, the LD format is designed with properties similar to SPEG. It is assumed that the format is hierarchically coded, i.e., a segment of higher layer data is worthless if the corresponding lower layer data segment has been lost. It is also presumed that the bit rate is constant, and that all layers have equal segment sizes and equal bandwidths. The RTP payload for this format includes a header as shown at the bottom of Figure 8.2, which includes a sequence number and a layer field. The latter specifies the layer of the video data that follows the payload header. The MTU size chosen by RTP can differ from the segment size, therefore, the payload of one RTP packet can contain several segments.

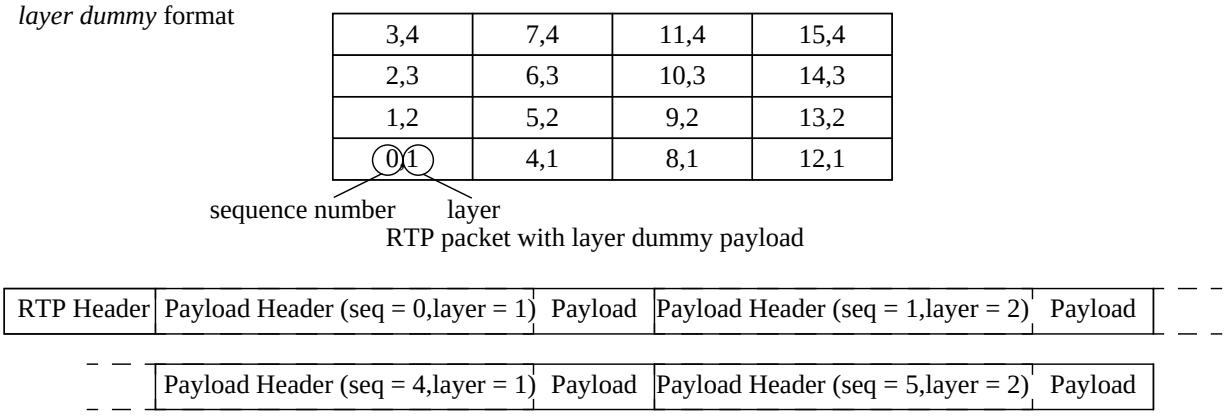


Figure 8.2: Layer dummy format and RTP packet

For the experiments (see Section 8.5) a four-layer format as shown at the top of Figure 8.2 is used, where the first digit specifies the sequence number and the second the layer of the segment.

C) LC-RTP Extensions

In preceding work (see Appendix A), an extension to RTP that provides lossless transmission of AV content into caches and concurrently, lossy real-time delivery to end-users is proposed. Here, an extension to LC-RTP is made such that two different types of transmission of missing segments are possible. Depending on different factors like popularity of the video and kind of client it

might be necessary to transmit only the losses that occurred during the transmission or in addition transmit the segments of the layer-encoded video that were not transmitted at all (e.g., in the case of congestion). For the second case no modifications must be made to LC-RTP, since all missing segments are transmitted as with the original LC-RTP. In the first case, two modifications to LC-RTP must be made:

- The sender stores a list of the segments that are actually sent into the network. With the aid of this list the server can identify which segments have to be transmitted and which do not. The client sends requests for the transmission of missing segments as long as all segments from its list of missing segments are received, a maximum number of retries is reached, or a BYE message from the server is received.
- When the server notices that the client request contains only unsent segments it sends the client a BYE message to stop it from sending further requests. Since the client is already in the loss collection phase it interprets this message differently from a BYE that is sent at the end of the initial transmission and stops sending requests for retransmission.

This extension to LC-RTP requires only minor changes to enhance its functionality and no new protocol messages must be introduced. In order to distinguish between the two retransmission methods the *subtype* field of the application-defined RTCP packet is subdivided as follows: The first bit is used to indicate which retransmission method should be applied (0 = only losses, 1 = losses and not initially sent segments) and the remaining four bits are used to identify the type of the packet. In Figure 8.3, an example transmission from server to cache is shown. Due to bandwidth limitations only three out of four layers are sent to the cache. During the transmission some of the segments are lost. These losses are recognized by the cache and requested for retransmission. Since the cache is not aware of the fact that the server did not send the segments belonging to the fourth layer, those segments are also requested. The server keeps track of the segments that have been sent to the client initially and retransmits only those segments. If the client requests only packets that have not been sent, the server sends a RTCP BYE message and the retransmission phase is terminated.

8.4.2 Signaling

In the KOMSSYS streaming environment, RTSP is used as application signaling protocol. Here, it is shown how RTSP can be used to allow members of a streaming session to negotiate if they are capable of a congestion controlled streaming session or not. To achieve this capability a new *transport-protocol* identifier in the *Transport* header, called **RTP/TFRC/UDP**, is introduced. This would, e.g., allow a non-RTP/TFRC-capable client to send a *SETUP* message as shown in Figure 8.4 and, therefore, initiate a non-congestion controlled session between itself and the cache. The cache on the other hand can modify the *transport-protocol* identifier and initiate a congestion controlled session between itself and the server.

If the cache is capable of establishing a congestion controlled session towards the server and the client, the tag stays unchanged and is forwarded to the server. An equally extended server replies

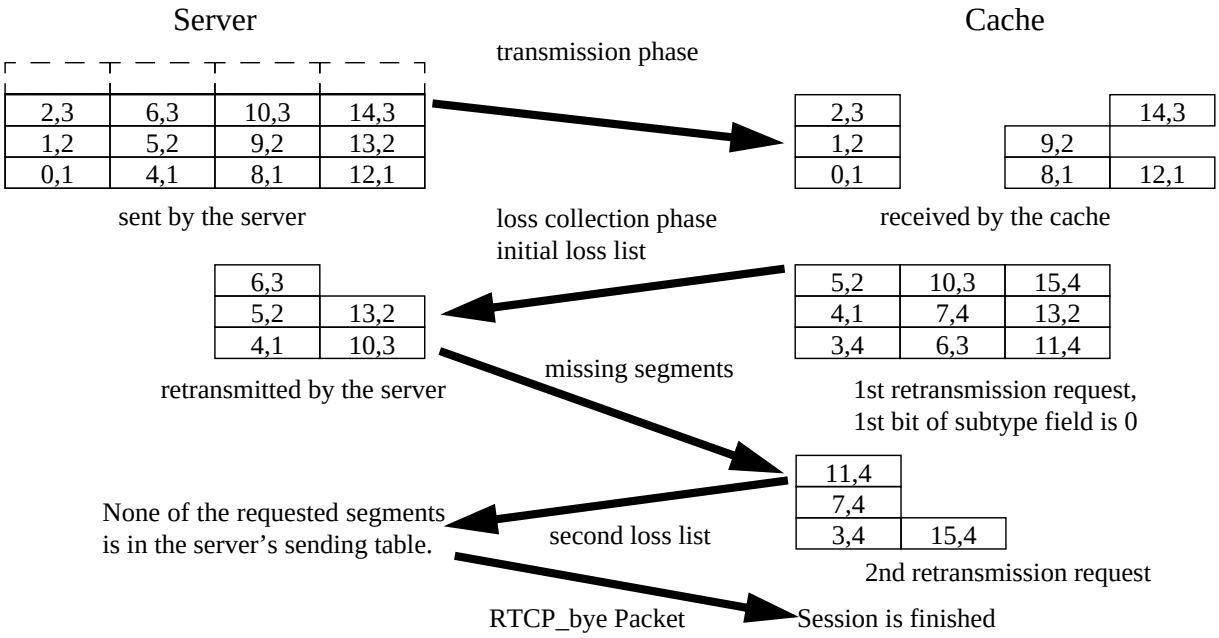


Figure 8.3: LC-RTP for layer-encoded video

with the appropriate *transport-protocol* identifier depending on whether it is RTP/TFRC capable or not. The cache modifies the identifier according to its own capabilities and the clients initial request and forwards the message towards the client. A complete communication between server, cache, and client regarding the RTSP *Setup* message is shown in Figure 8.4.

Since a cache should also support clients that are not capable of congestion control (see Section 8.1), it can automatically insert the *transport-protocol* identifier in the setup message that is forwarded to the server and remove it from the server's reply before it is forwarded towards the client. Thus, allowing a congestion controlled stream between server and cache and a standard RTP/UDP stream between cache and client.

An additional information that is signaled by RTSP is the maximum bandwidth at which a stream can be sent to the client or the cache. For this purpose, RTSP provides the *Bandwidth* request header field, that describes the estimated bandwidth available to the client, expressed as a positive integer and measured in bits per second [16]. In the case of the KOMSSYS implementation this header field is also added to the *SETUP* header and used by either the server or the cache to limit the maximum bandwidth TFRC used to stream the content. In Section 8.2, the functionality that allows a faster streaming than the default rate between the server and the cache and how the cache serves as a buffer for the client are described. To achieve this functionality, the information in the *Bandwidth* request header field is modified by the cache. The *Setup* signaling for this case is also shown in Figure 8.4.

8.4.3 Putting the Pieces Together: Cache

In Appendix C, the stream handler architecture that builds the basis for the KOMSSYS streaming platform is presented. Stream handlers are media processing units which can be bound together

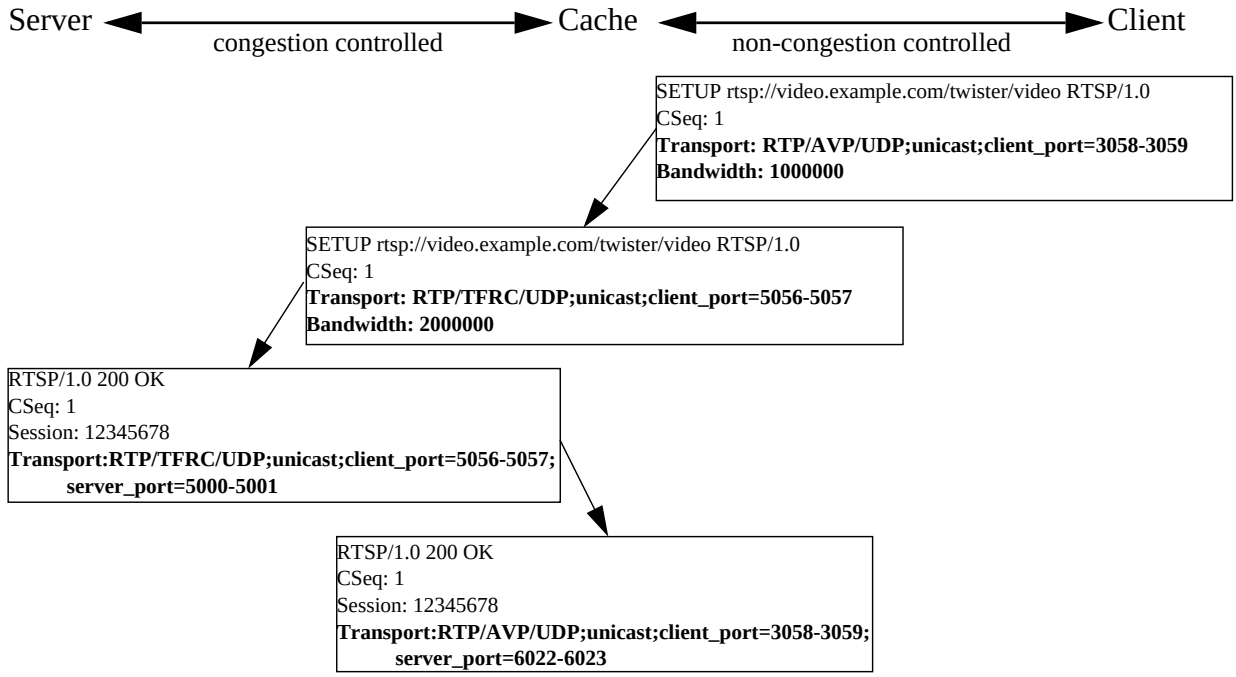


Figure 8.4: Modified *Setup* messages

by a controlling entity into a directed graph. This approach allows the creation of new functionality in either server, cache, or client by reusing existing stream handlers.

Figure 8.5 depicts the streaming graph at the cache including both alternatives for congestion controlled and standard RTP/UDP-based streaming towards the client. Conditional write-through caching is enabled by the use of the PacketMultiplierSH that creates a copy of each received segment, which is then stored at the local disk. This part of the streaming path is only activated if a positive decision to store the video in the cache is made by the caching strategy. The two alternative paths that handle the data forwarding towards the client are created based on the RTSP information that is received from the client. If the client RTSP *SETUP* message contains the RTP/TFRC transport protocol identifier, the upper path in Figure 8.5 is created, if not (in the case of RTP/AVP) the lower path is created. The PushPullSH in the upper path implements a queue with limited length that allows controlled dropping. In the lower path, all data is forwarded as it arrives at the cache.

Both paths have additional functionality that is only needed for the measurement in the following experiments. If logging is requested, a log entry is generated in the respective trace files for each layer of a frame that is either received at the cache or the client.

8.5 Experiments

8.5.1 Setup

The goal of the experiments is to verify that the functionality described in Section 8.2 can be realized by an implementation. Therefore, the KOMSSYS streaming platform is extended by the

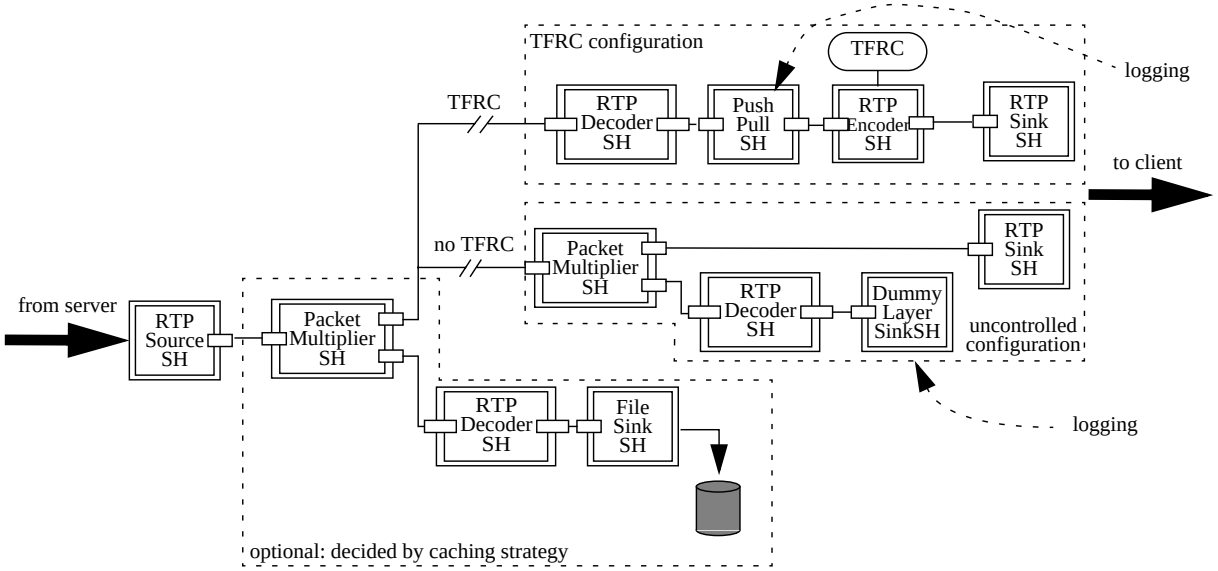


Figure 8.5: Cache configuration for TFRC downlink and for uncontrolled downlink

functionality described in Section 8.4. Two sets of experiments were conducted with the modified platform. The first one was a testbed-based experiment. The five computers (Server-NISTNet1-Cache-NISTNet2-Client) used for this experiment were standard Pentium-III PCs (850 MHz) with 256 MB of RAM and Linux 2.4 as operating system. During the experiment, KOMSSYS server, client, and cache were used. The use of two additional NISTNet [156] network emulators allowed the reduction of the link conditions between server and cache and between cache and client from the original capacity of a switched 10 MBit/s Ethernet. The measurements described in the following section were executed in an environment without additional network traffic, since the testbed is not connected to any other network. For the experiments a bandwidth of 1 Mbit/s for the server-cache link, and 512 kbit/s for the cache-client link is chosen. This allows full quality streaming between server and proxy, while the quality must be reduced on the link between cache and client due to a limited bandwidth. The bandwidth of the layer dummy video at full bit rate is 1 Mbit/s, with each layer making up an equal share of 256 kbit/s.The testbed setup is shown in Figure 8.6.

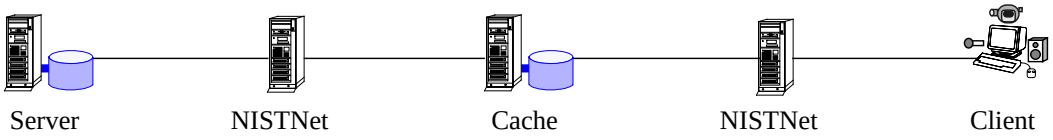


Figure 8.6: Testbed setup

To complement the artificial network setup with real world data, a measurement over the Internet between Oslo and Darmstadt was performed in addition. The setup for this measurement is shown in Figure 8.7. The test machines are different Pentium-III Linux 2.4 machines. The network was fairly unloaded during the test.

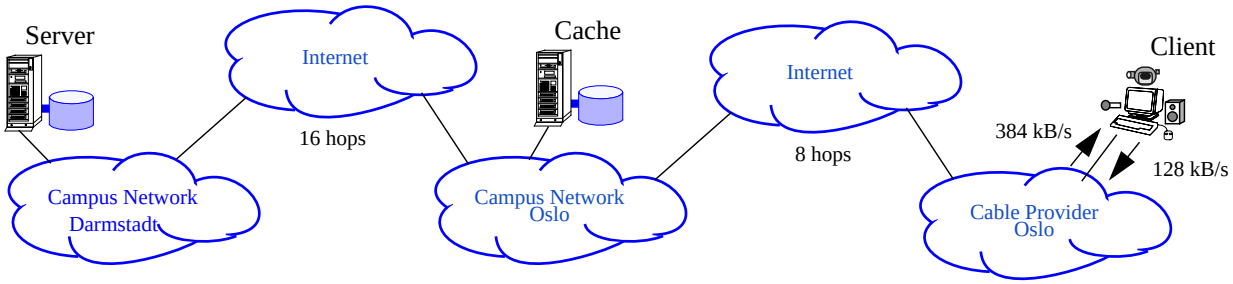


Figure 8.7: Internet-based measurement

8.5.2 Measurements

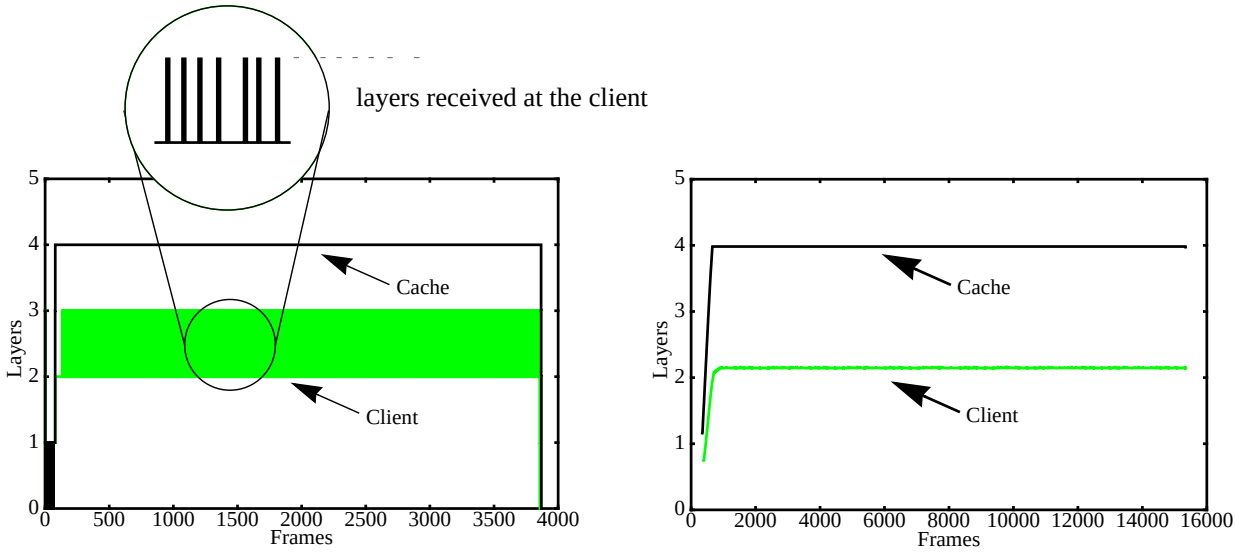
In the following, measurement results from both test setups are presented. The results are shown in graphs that depict the number of received layers at the cache and at the client. With the logging mechanisms presented in Section 8.4.2 the number of layers that are either stored at the cache or received at the client is measured. In the case of hierarchically layered codecs, the loss of a segment with a lower layer number implies that all arriving segments with higher layer number, which belong to the missing segment, have to be considered lost as well. Under this consideration, one graph that shows the valid, arrived layers for every individual frame in the video is created. The frequency of layer changes makes it hard to visualize the average number of layers that are received. To provide a better insight, a second graph is shown, which presents the same quality in terms of layers once for each frame in the movie, but instead of just showing the number of layers for the individual frame, the average of the frame and the previous 100 frames is shown. This presentation hides short-term quality changes, but makes it easier to identify mid- and long-term changes in the quality development.

A) TFRC in the Access Network

In this experiment, the client demands less bandwidth from the cache than the cache demands from the server. This allows the cache to be filled with a higher quality version of the video object, and clients that request the same object later in time are able to receive higher qualities as well. In Figure 8.8, the testbed-based results for this experiment are presented.

The detailed observation of layers received at the client shows a very low loss rate for packets that contain layer 1 and layer 2 data. This implies that TFRC chooses a sustainable bit rate, which can in this case support a load of 2 layers. Additional packets of layer 3 are inserted into the stream for bandwidth probing, and have a considerable probability of getting lost.

The use of TFRC allows the usage of this additional bandwidth above the sustained bandwidth in any application-defined way. It is inappropriate to use it for faster-than-real-time transmission of the base layer because the high loss probability would make the additional use of retransmission necessary. For a hierarchically encoded layer-encoded video, however, this additional layer, which is too unstable to be used for transmitting another layer, may be used to implement retransmission (as proposed for fair share claiming in Chapter 7) or forward error correction for the base layer.

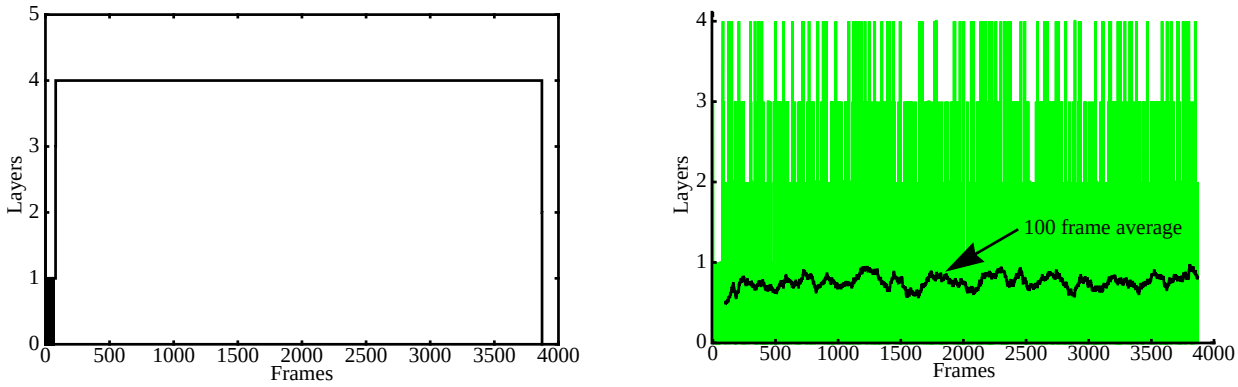


Received layers at cache and client

Average number of layers per 100 frame interval

(a) (b)

Figure 8.8: Congestion controlled by TFRC in the access network (testbed)



Received layers at the cache

Received layers at the client:
individual number of layers received for each frame, overlaid with 100 frame average

(a) (b)

Figure 8.9: Uncontrolled UDP in the access network (testbed)

By examining all scenarios of TFRC use, a strong instability in the early phase of the connection is observed, compared to the stable operation when the appropriate bandwidth has been found. Since the duration of this start phase is relatively long, it seems reasonable to enter the initial slowstart phase with the bandwidth of a previous connection rather than a full slowstart as proposed in [157].

B) UDP in the Access Network

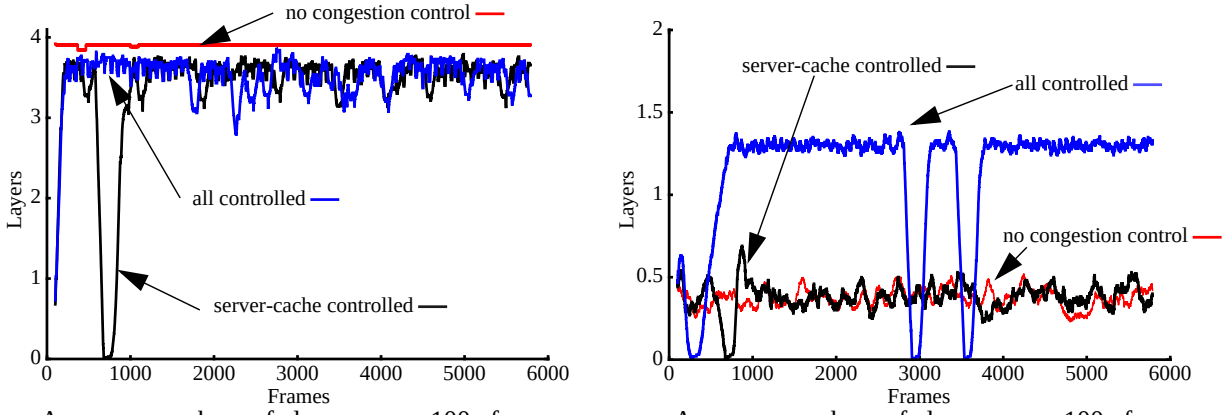
In a second measurement, the effects of using UDP in the access network combined with TFRC in the backbone is investigated. This question is legitimate for several reasons. First, bandwidth in

access networks that use DSL or cable modems is frequently restricted by shaping but provides a guaranteed minimal throughput in the provider's network. If caches are deployed at a provider's site, the customers should be able to exploit this guaranteed bandwidth such that an additional consideration of TCP friendliness in the access network is not required. Second, standard-compliant clients do not use TFRC. Thus, it is important to understand the interplay of using TFRC in the backbone and regular RTP/UDP in the access network. The results of the experiments show that when the access link becomes the bottleneck link, random loss occurs in case of UDP. This should be compared to the controlled loss that is possible when the available bandwidth can be estimated based on TFRC information. It becomes apparent from the detailed graphs in Figure 8.8 and Figure 8.9 that the UDP approach suffers from one problem. The rate of valid packets is not sustained, which results in frequent quality changes. Since the access link can support approximately two layers and the cache forwards approximately 4 layers, the probability of a successful arrival of a layer 1 block is 0.5 and a layer 1 and a layer 2 block 0.33. This yields an estimated average number of 0.66 valid layers. The results in Figure 8.9 support this estimation. The difference in TFRC performance between the testbed-based and the cable-based access network becomes visible by comparing the testbed results with the real world traces in Figure 8.10. Whereas the congestion controlled approach yields about 2/3 of the uncontrolled throughput on the access link, it is just 1/3 in the real-world scenario. This limited throughput is likely due to artificial limitations of the TCP throughput to 384 kbit/s, which are defined in the service agreement. However, it can be observed in both scenarios that the short-term quality changes with congestion control are considerably lower than without. In the observed case, it does not seem viable to use the bandwidth available between the server and the cache for a complete transfer of movies into the cache, because of the contractual limitations of the downlink. However, since a bandwidth very close to the demanded bandwidth is made available, it is appropriate to allocate it for faster-than-real-time transfer and retransmission.

The incursion of the bandwidth can only be explained by some unexpected behavior on the access network in Oslo.

C) Hierarchical and Non-hierarchical Codecs

Since the packet droppings are entirely random, it is important to notice the disadvantage of a hierarchical layer-encoded video codec: a codec that uses independent layers (as described in Section 3.3.1) would have considerably more stable results. A comparison with the results that a non-hierarchical codec would yield is presented in Figure 8.11. The two graphs in the figure show two alternative interpretations of the same trace files, a hierarchical interpretation on the left and a non-hierarchical interpretation on the right. Since no filtering takes place in the TFRC case unless the forwarding queue in the cache is full, there is no need to consider the preference of forwarding packets with lower layer number. Also, in the non-hierarchical case, the number of forwarded packets would be identical even if the queue would not prefer selected packets. Obviously, the congestion controlled approach behaves identical for both kinds of encoding. Both cases that are uncontrolled in the access network achieve the same throughput in this case, which is very close to the connection's limit. Nevertheless, the completely uncontrolled case allows for a more stable



Average number of layers per 100 frame interval received at the cache

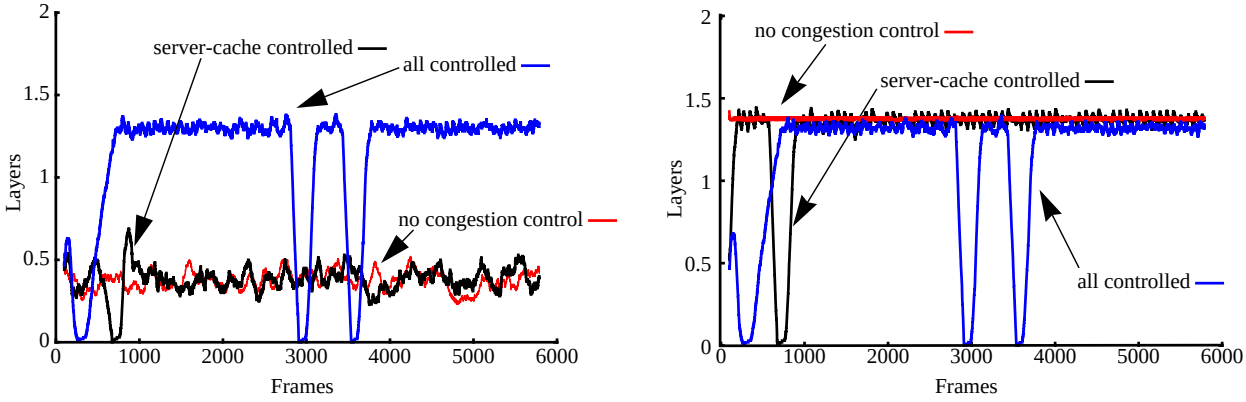
(a)

Average number of layers per 100 frame interval received at the client

(b)

Figure 8.10: Results for different control approach (Internet traces)

bandwidth in this case, i.e., less quality changes than with the congestion-controlled approach occur. The most surprising observation in the non-hierarchical graphs is certainly that this smoothness is not achieved for the approach that uses congestion control in the backbone and no congestion control in the access network. Although the number of packets that are forwarded onto the access link in an uncontrolled manner far exceeds the available bandwidth, the rate fluctuations of the controlled approach are disseminated to the access network. This observation leads to the conclusion that even if simple shaping is performed in a cache that separates the backbone from the access network, a queue should be introduced into the forwarding path to hide the bandwidth fluctuations.



Average number of layers per 100 frame interval received at the client for hierarchical layer-encoded

(a)

Average number of layers per 100 frame interval received at the client for non-hierarchical layer-encoded video

(b)

Figure 8.11: Hierarchical vs. non-hierarchical layer-encoded video

8.6 Summary

In this chapter, the architecture and implementation of a scalable, TCP-friendly video distribution system for heterogeneous clients is presented. This architecture allows the usage of clients with and without congestion control while a congestion controlled transmission is always performed between server and cache. Congestion control in our system is achieved by the integration of TFRC in RTP. Next to the modifications in RTP an extension to LC-RTP to support the lossless distribution of layer-encoded video into the cache is presented. In addition, it is shown how RTSP signaling can be used to negotiate the type of streaming (congestion controlled or not) and maximum bandwidth between the single entities of the streaming system. Based on these implementation experiments in both, a testbed and the Internet, are conducted.

The results obtained in the experiment indicate that all connections should be congestion controlled, but certainly if hierarchical layer-encoded video is used. With standard clients that cannot perform congestion control on the access link the usage of non-hierarchical layer-encoded video would be an option. It must be mentioned that in the non-hierarchical case the uncontrolled approach performs best, but is not a TCP-friendly approach.

Chapter 9: Conclusions and Outlook

9.1 Conclusions

The major conclusion that can be drawn from this thesis is the fact that two-dimensional scalable streaming in today's Internet is possible. A new architecture for Scalable Adaptive Streaming (SAS) is presented which combines system and content scalability. The SAS architecture consists of mechanisms which allow the delivery of layer-encoded video in an acceptable quality to the user while the overall amount of network traffic is reduced and, thus, the overall scalability of the system is increased. The new mechanisms are based on Internet technology that is available today. The applicability of these mechanisms is shown in the preceeding chapters by applying the three methods; assessment, simulation, and implementation.

One major issue is the introduction of quality variations introduced by the combination of adaptive (due to network congestion) streaming and scalable content. These variations are a natural side-effect of the adaptive streaming mechanisms which should be kept to a minimum in order to keep the quality of the transmitted stream acceptable. Since caches are involved in the distribution of the video objects, additional mechanisms can be applied to reduce the variations of a layer-encoded video.

Due to the lack of a subjective assessment of variations in layer-encoded video, part of the thesis is dedicated to this issue. This investigation is the first of its kind and results in a set of guidelines being used to develop transport mechanisms in SAS and an objective quality measure. The results from the subjective assessment influenced the development of a new objective measure, the spectrum, that is more suitable than the well known Peak Signal-to-Noise Ratio (PSNR).

To improve the quality of layer-encoded video either stored on a cache or streamed through a cache to the client, new mechanisms to reduce layer variations are developed. It is shown that an optimal solution is computationally infeasible and, thus, new heuristics are proposed. The development of these heuristics is also influenced by the insights gained from the subjective assessment. A simulative investigation shows the performance and applicability of the new heuristics. To compare the heuristics with each other, the spectrum is used as a quality metric. In addition to the heuristics, different retransmission scheduling approaches are introduced which either maximize the quality of the cached layer-encoded video or the one that is streamed to the client. The results of the simulation show that retransmission scheduling is an effective means to reduce variations in layer-encoded video. With the fair share claiming (FSC) technique, a new transport mechanism is presented which allows the retransmission of missing segments from the server to the cache.

To support heterogeneous clients in a video distribution system extensions for signaling and data transport are made. These extensions are implemented in the existing experimental streaming platform built during this thesis. On the basis of this implementation measurements in both, a test-bed and the Internet, are performed to prove the applicability of the proposed protocol extensions. The results of these measurements clearly show the benefits of congestion controlled streaming in combination with layer-encoded video.

Finally, an investigation on the reduction of variations on layer-encoded video in the case that retransmissions cannot be performed resulted in the polishing technique. This technique was further developed into a fine-grained cache replacement algorithm that can perform replacement for layer-encoded video based on the popularity of the video objects.

Altogether, the contributions made in this thesis allow for VoD systems in today's Internet that outperform existing ones. The results of the performed investigations can either be used combined to build a new video distribution system or existing systems can be extended by one or more mechanisms presented in this thesis.

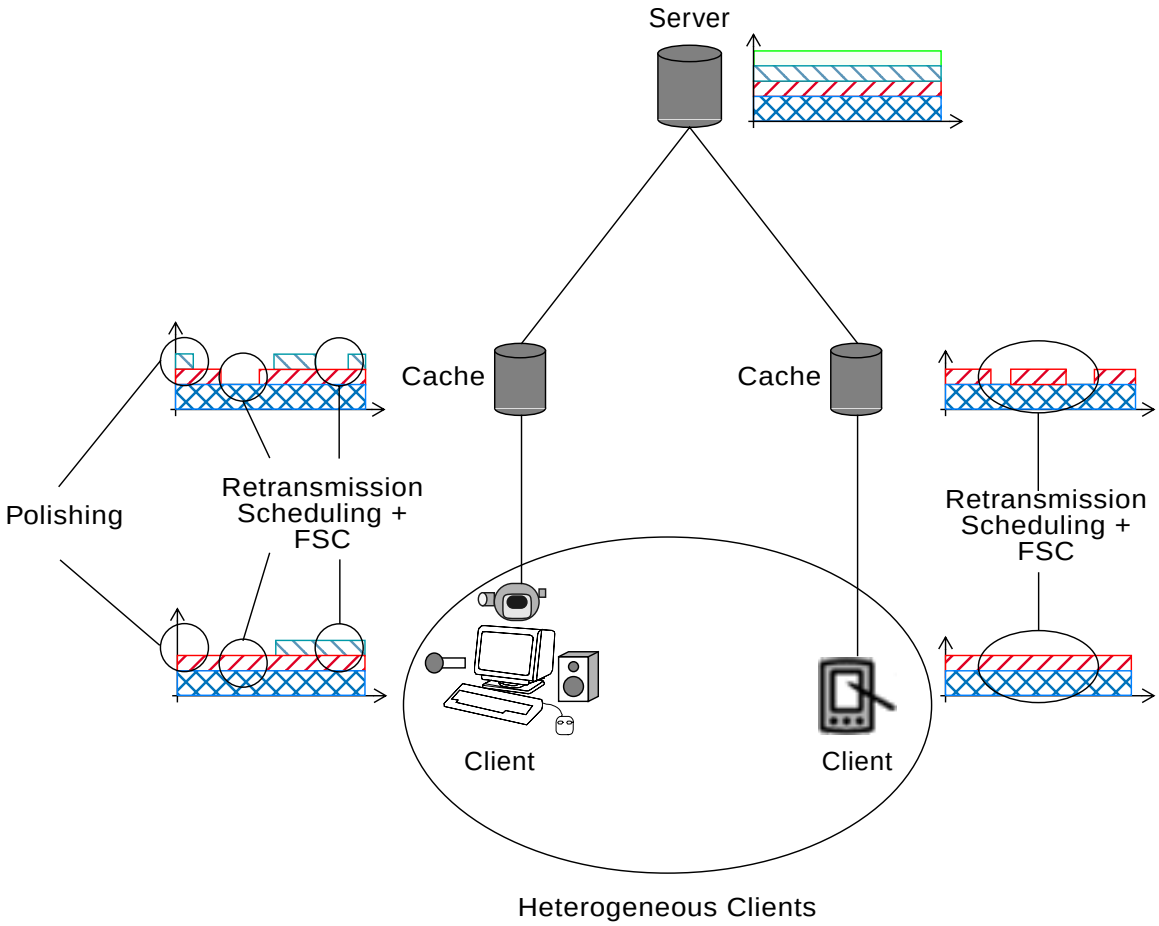


Figure 9.1: Contributions in this thesis

9.2 Outlook

Applying the results obtained in this thesis can lead to more efficient and better quality streaming of video data in today's Internet. The scalability of the distribution infrastructure is increased by the usage of caches on the one hand and layer-encoded video on the other. The combination of both scalabilities, system and content, does not only increase the quality of the received stream at the client but also allows the support of heterogeneous clients. Thus, using these new mechanisms, Video-on-Demand systems can be created that are more scalable, offer a better quality, and support a large variety of clients.

As already mentioned above, the presented approaches can either be used in combination or stand alone. For example, the results of the subjective assessment might also be valuable for researchers who investigate objective quality metrics for layer-encoded video. The retransmission scheduling and polishing mechanisms can be integrated in already existing systems. Especially the polishing-based cache replacement method can be applied in caches fairly easy since no other parts of the cache like, e.g., communication protocols are affected.

A possible future task could be the investigation, if layer-encoded video can be replaced by multiple description coded (MDC) video. The results of the measurement performed in Chapter 8 indicate that the usage of non-hierarchical encoding formats (like MDC) can be advantageous in comparison to layer-encoded video in certain cases. Unfortunately, this task is not simply executed by exchanging layer-encoded video through MDC video. Rather it starts with a new subjective assessment for MDC video to verify if the assumptions that retransmission scheduling and polishing are based on are also valid for this new encoding format. Another interesting research area is the realization of streaming in a peer-to-peer environment. Some of the existing problems were identified in [158] and it is shown that MDC is a well suited encoding format for streaming in peer-to-peer networks, but there are still many open issues to solve.

This thesis already provides solutions for the support of heterogeneous clients which also includes wireless clients. Yet, since the channel characteristics of the wireless channel are often different from wired channels, new or modified transport mechanisms are necessary. Initial work [159] has shown that the modification of TFRC can improve the quality of a stream in an UMTS network. Further investigations are necessary to show if caches can provide additional support for wireless clients. For example, caches could also act as a gateway which is located on the transition between wired and wireless network and allow a differentiation between congestion-based and fading-based losses. New research issues in this area will also depend on the technologies 4th generation wireless networks will be based on.

Very important issues in relation to Video-on-Demand are digital rights management (DRM) and copyright protection. There is a high demand for such mechanisms from the content providers, but none of these are available so far.

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2003	SPIE/ACM - Multimedia Computing and Networking as publicity chair
2003	ACM - Multimedia Conference as programm commitee member
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Appendix A: LC - RTP (Loss Collection RTP)

A.1 Motivation

In this appendix a protocol is presented which allows the lossless transport of audio and video data into caches while, in parallel, this data is streamed to one or several clients. In contrast to existing approaches on reliable multicast (see Section 3.3.4) the protocol presented here was especially designed for use in content distribution infrastructures for multimedia data.

An important characteristic of A/V content is the fact that it should be transmitted in real-time. This implies that a client cannot wait too long (assuming a limited buffer) for any resent packets instead of displaying the current data, so the normal data flow must persist and any retransmission must happen aside of the normal data flow. The situation is different for caches where the arrival of data is not time critical (at least not as critical as at the client). Therefore, additional functionality in server and caches in combination with LC-RTP allows a lossless transport into caches while data is streamed to clients as usual.

LC-RTP is designed as a RFC-compliant extension to RTP for reliable file transfer that requires no infrastructure modifications except on servers and caches. LC-RTP provides lossless transfer of real-time data by using loss collection (LC). The sender sends RTP-packets via multicast to all receivers (clients and cache servers) in the multicast group. If a cache server detects a packet loss during the transmission it will be memorized in a list. At the end of the session caches which are storing the video from this multicast transmission request the missing parts from the sender. The sender retransmits all missing blocks and waits until no more packets are requested. LC-RTP works also in simple unicast mode where data is streamed from the server through the cache to the client.

In A.2 the standard protocol set for A/V streaming in the Internet and the loss collection extensions are presented. The design and the general functionality of LC-RTP are presented in A.3 while A.4 shows the extension of standard RTP to support LC-RTP. These protocol extension were implemented in the KOMSSYS streaming platform (see Appendix C) and measurements based on this implementation are presented in A.5.

A.2 Protocol Set for Streaming Media

In the Internet, one set of protocols is currently adopted -partially or completely- by companies in their products for streaming media (Apple, Real Networks, SUN, IBM, Cisco, FVC.com, ...). These protocols are the combination of RTSP/SDP for stream control and RTP/RTCP for streaming.

A.2.1 RTSP/SDP

The Real Time Streaming Protocol (RTSP, [16]) is an IETF RFC that is supposed to be used in conjunction with various other protocols. Its functionality is not generic but rather concentrated on stream control. It references elements of HTTP to which it is weakly related. It can be used with either TCP or UDP as an underlying transport protocol. The data transfer protocol that is mentioned in the RFC and that interacts most closely with RTSP, is the Real-Time Transfer Protocol (RTP, [61]). The same approach applies for the session description protocols; although no fixed session protocol is defined, the RFC specifies the interaction with the Session Description Protocol (SDP, [64]).

SDP is originally considered as a companion protocol for SAP, the Session Announcement Protocol. However, besides this mode of distribution for session information, others like download from the web or E-mail distribution are also compatible with this kind of information.

Table A.1: Protocol set

reliable file transfer & real-time streaming	
LC-RTP • RTP-compatible until RTCP BYE message • use RTP header extensions • continuous byte count • retransmission after reception of loss lists	**LC-RTCP** • RTCP-compatible • user application-defined RTCP packets • loss-list report receiver to sender • retransmission request after random waiting time
stream control & sequencing	
RTSP • standard protocol • use SDP	**SDP** • standard protocol • specifies play range • different sources for data segments

A.2.2 RTP/RTCP

RTP (Real-Time Transport Protocol) was created to transport real-time data over the Internet. VoD, Internet telephony, MBone-conferences, and all video- and audio-conferences make specific time restrictions on how the data is delivered. RTP provides payload type identification, sequence numbering, time-stamping, delivery monitoring, and supports multicast if the underlying protocol provides this service.

Usually it is used over UDP, as UDP allows multiplexing and does not have any retransmission schemes like TCP. RTP is used together with RTCP (RTP Control Protocol [61]) which allows a

quality monitoring of the network connection and has minimal control over the session. Furthermore, RTCP can be used to identify the sender. The main task of RTCP is to send periodic control packets to all members of the session using the same distribution mechanisms as the data packets. The resulting protocol set is listed in Table A.1, including the tasks that are handled by each protocol.

A.2.3 LC-RTP

RTP with Loss Collections (LC-RTP) implements a unified protocol for stream transmission that is compatible with RTP, and reliable transfer of content into the cache servers. It solves these problems by making RTP reliable, while the ability is maintained that non LC-RTP capable clients (standard RTP clients) can receive an LC-RTP stream as well. The functionality of LC-RTP is described in Section A.3

A.2.4 LC-RTCP

Just as RTP has a companion protocol RTCP for the exchange of information about the data transfer, LC-RTP requires a companion protocol LC-RTCP, which needs to be RTCP-compliant. In application-defined RTCP packets, the receivers inform the sender about their losses after the reception of the BYE packet, unless all of its missing packets have earlier been reported by another receiver.

A.3 LC-RTP Design

In an environment for AV-caching it is absolutely necessary that the cached version of the content in the cache is stored 100% correctly to avoid error propagation towards the client. With the use of standard RTP on top of UDP, information that gets lost during transmission is also lost to the caches. The problem is that these errors would be transmitted with every stream that is forwarded from the cache server to a client. In any case that should be avoided, since it has to be regarded as a degradation of the service quality. During each transmission data can get lost and, thus, lead to a higher error rate in stored copies.

LC-RTP solves these problems by making RTP reliable, while the ability is maintained that non LC-RTP capable clients (standard RTP clients) can receive an LC-RTP stream as well.

To describe LC-RTP the transmission process is divided into two parts. The first part works almost like a regular RTP transmission and ends after the transmission of the original content following by the transmission of a BYE message. The second part follows this BYE message and is used to retransmit all lost data. In this scenario, the receiver is a cache that has received a request from a client but that has recognized that the requested content is not stored locally and, therefore, a request forwarding to the original or to a cache located upstream towards the server is per-

formed. Figure A.1 gives a general overview of the different steps that are executed during a LC-RTP session.

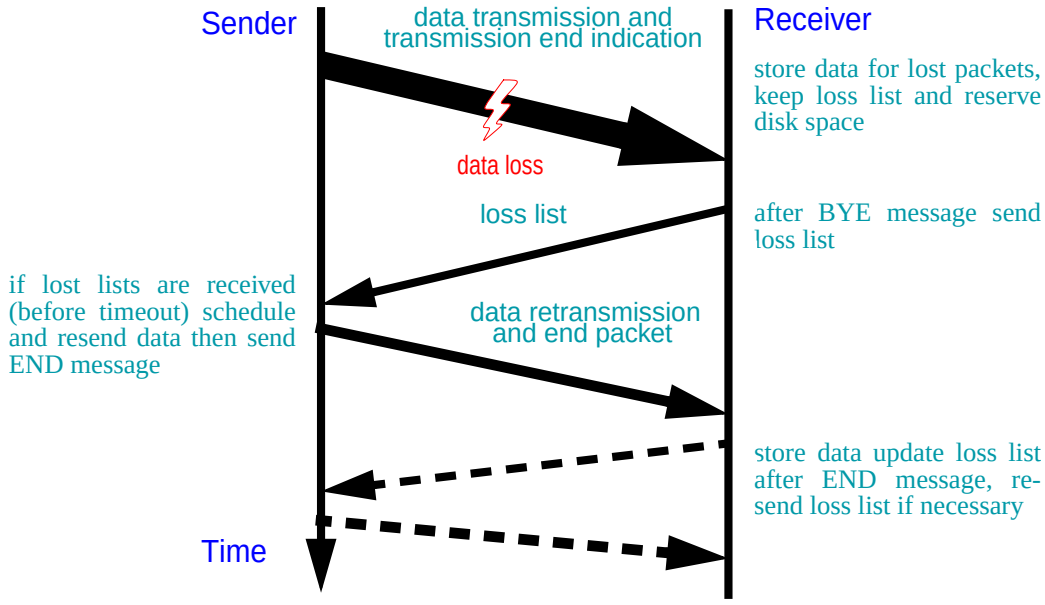

Figure A.1: LC-RTP communication

A.3.1 Unicast vs. Multicast

Depending on the popularity and the availability of IP multicast it might be required to perform the data distribution either via unicast or multicast. Therefore, it was an additional design goal to support both transmission modes.

The unicast mode is simpler than the multicast mode, since only one cache and one client are served. Also the retransmission phase is simplified because loss reports are only generated by one receiver. Compared to the multicast model the data must be forwarded through the cache towards the client, while with the multicast method data can be streamed in parallel to several caches and clients (see Figure A.2). Additional measures that had to be taken to allow multicast in combination with LC-RTP are explicitly outlined in the following sections of this chapter.

A.3.2 Actions During the Content Transmission

The sender streams the content that is requested by a client as a multicast stream to all receivers of a multicast group including the clients. In order to give the caches the possibility to reserve exactly the required disk space in case of data loss, it is necessary to send information beyond the regular information of a RTP packet. In the case of LC-RTP this information consists of a byte count which is included in each RTP packet. This mechanism facilitates the synchronization between byte count and the data which is represented by it. If the byte count were sent in an extra

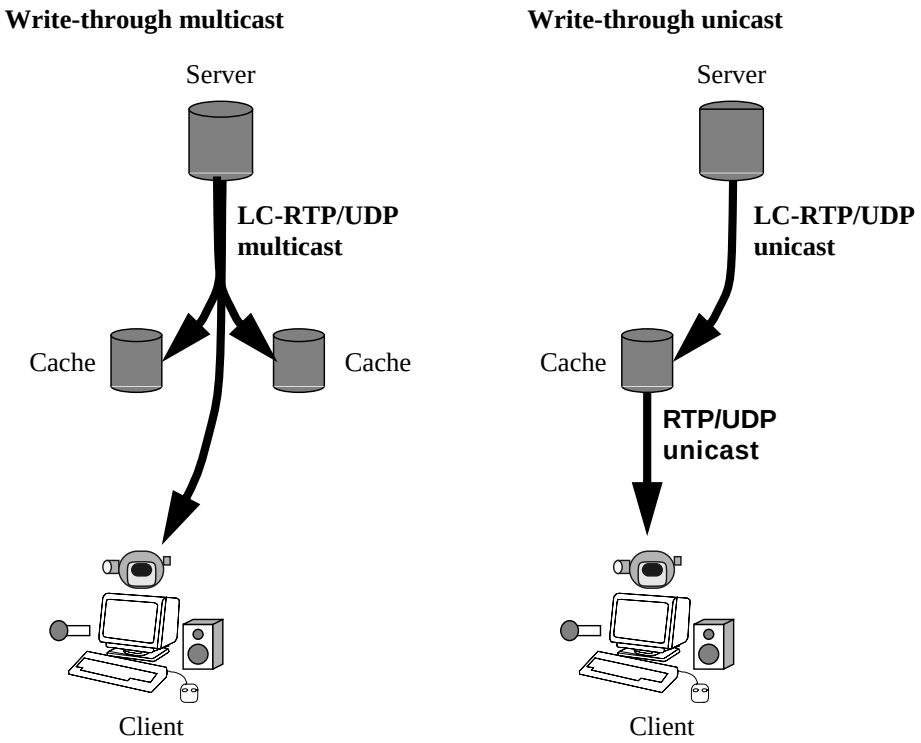

Figure A.2: Multicast and unicast LC-RTP

packet, e.g. via RTCP, the sequence of the byte count and data packet can be interchanged, or the byte count packet can get lost.

The receiver stores the data and detects a loss by checking the byte count with the last memorized byte count. If a packet loss is detected, the difference between the two byte counts and the length of the actual packet is computed and this computed size can be reserved on the disk for a later insertion of the retransmitted data (see Figure A.3). The received payload of the packet is then stored after this reserved gap. Furthermore, the loss must be written to a loss list. If no loss is detected, the received data is stored on the disk immediately.

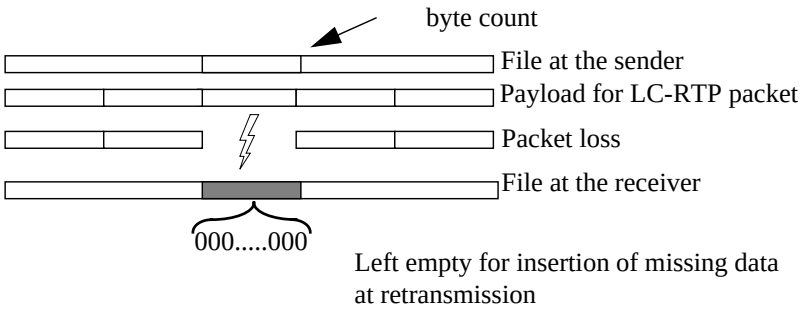

Figure A.3: LC-RTP byte count supports retransmission

Each cache server implementation has to transform the byte count value into its own file indexing information. As a consequence it is possible to have different file layouts on the server and

cache. For example one cache implementation stores the file as raw data and another stores some header information with it.

As a consequence of including the byte count in the data packet, and the requirement of serving regular RTP clients, only an RFC-conforming protocol extension was an option; including the byte count in the RTP payload of the packet would cause problems for standard receivers (see Section A.4).

At the end of the transmission, an end packet is sent including the last byte count, in order to inform the receivers of the normal end of the transmission including information to check whether data preceding the end packet was lost.

Reserving the computed space in the file in case of a loss detection has advantages for several reasons. The approach of reserving the correct amount of space on the hard disk is very simple and efficient, because it preserves the sequential nature of the stored data. This property is essential for an efficient use of a hard disk, as seeking on a disk importantly diminishes its throughput. Furthermore, this allows LC-RTP to be compatible with multimedia file systems ([160], [161]) which are penalized by inserting or do not support it at all.

A.3.3 Actions After the Content Transmission

After sending the end packet the sender starts a timer. This timer should be a multiple of the worst case RTT (Round Trip Time) between the sender and the known receivers. This RTT can be computed with the periodic RTCP packets that are sent for calculations of the network quality. During this timer period at least one loss list has to be received from a cache that has detected packet losses, or the session ends.

With the reception of the end packet the cache finishes the normal procedure of the transmission of the content and starts the procedure for initiating retransmissions. To avoid a possible overload of the sender, loss lists are sent from receivers after a random amount of time in the case of multicast LC-RTP. The loss list includes all ranges of the detected data losses. If ranges are direct neighbors, they are combined into one range, in order to keep the size of the list small.

If a loss list arrives at the server, the requested data ranges are stored in a schedule list. This list includes a counter for each range to indicate the number of requesting clients. This allows the use of a strategy for building a retransmission schedule (e.g. most frequently lost first).

Resent packets are of the same size as the packets that were sent during the first transmission to simplify storing at the receiver. The resent data range is deleted from this list. The client saves each requested, retransmitted packet at the position that is indicated by the byte count. Concurrently, the loss list is updated. If the byte count is not included in the loss list the packet is discarded.

When the last entry of the list is processed and deleted, the sender resends the end packet in order to inform the receivers that this retransmission cycle is over. This procedure is repeated until an application-specific retransmission counter has reached its threshold value or until no more loss lists are sent.

To avoid the blocking of a receiver a timer is necessary that terminates the session if no end packet or other resent packets are received after a considerable period.

A.4 Use and Integration of Protocols

The design of LC-RTP was made within the constraints of an RFC-conforming RTP implementation. This section describes how the standard RTP protocol is extended to meet the goal described above.

A.4.1 LC-RTP as an RTP Extension

The main problem in mapping LC-RTP into RTP is the byte count, as it has to be included into the header of RTP (see Section A.3). This is necessary in order to keep content of LC-RTP packages compatible with RTP-related packaging RFCs and therefore to make it possible for standard RTP clients to receive LC-RTP streams. One possibility to insert the byte count into the RTP header and not into the payload is the use of the extension header of RTP (Figure A.4). By setting the

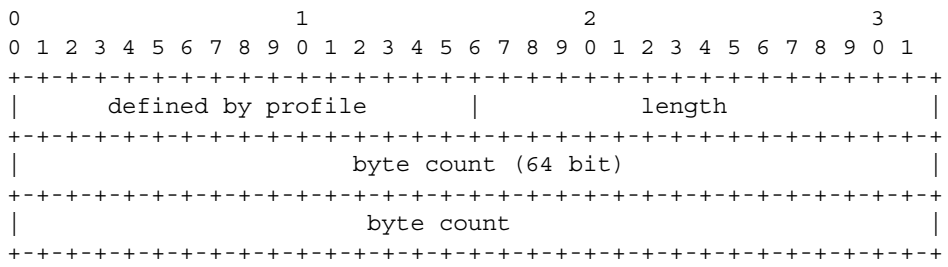

Figure A.4: RTP header extension

extension bit a variable-length header extension to the RTP header is appended. LC-RTP defines two kinds of header extensions. They are defined to easily distinguish whether a packet is sent as part of the regular stream or during a retransmission phase. The only difference between them is the value in the identifier field. Each extension header consists of the two RTP dependent extension fields plus an additional byte count field. For a current video streaming application this field should be 64 bits long, as a cyclic byte count must be prevented.

During the usual transmission, the RTP transmission is made as usual, except for the byte count which is included in the RTP extension header. At the end of the transmission an end packet is sent. An appropriate way to do this is by sending an RTCP packet. This packet should not be the normal RTCP BYE packet, as this is used for other meanings. Thus, an application-dependent extension RTCP packet must be created as shown in Figure A.5.

LC-RTP defines two application defined RTCP packets. The first one is the end packet and the second one is the loss list packet. The only additional data transmitted in the end packet is the last byte count of the session. The name of the packet itself is enough information for the receiver to interpret this as the end of the normal transmission. The list appended into the loss list packet should be appended as a list of byte count ranges.

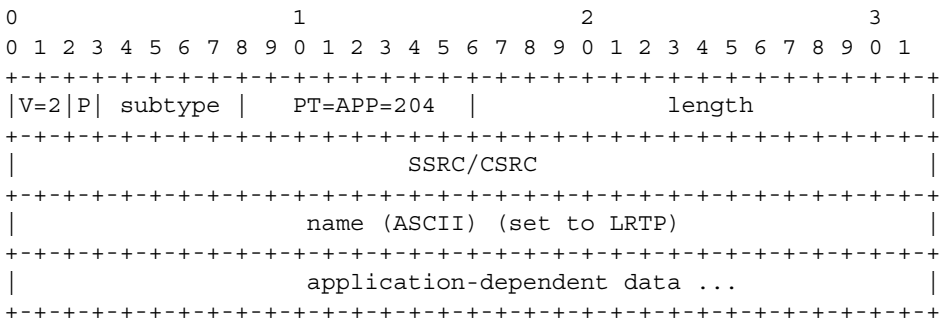

Figure A.5: Application defined RTCP packet

The extension to RTP is minimal and should be ignored by other applications. This is very important, because it ensures that a cache update can be made in parallel to a customer request.

Experiment of LC-RTP in combination with *vic* and *vat* resulted in the rejection of all RTP packets with an extension header. A closer look at the source code of both revealed that the RTP implementation is not standard compliant.

For the intended application class, the header extension introduced by LC-RTP is sufficiently cheap with an overhead of 8 to 12 bytes per packet. Furthermore, this type of extension is defined in the original RTP RFC ([61]) and should -theoretically- be implemented by all RTP implementations.

A.5 Tests

RTP and LC-RTP were implemented in the KOMSSYS streaming platform. This implementation was used for an Internet-based experiment which is described in detail in the following.

A.5.1 Test Scenario

The goal of this experiment is to show that LC-RTP performs as well and reliably as other data distribution protocols (e.g. FTP) and can be used for the reliable distribution of AV content both via unicast and multicast.

Two different video objects (6MB and 20MB of MPEG-I Movie) were transmitted from locations in Germany, the US, and Canada to a receiver located at our institute (Darmstadt, Germany). The results from transmissions between the US (National Institute of Standards and Technology) and Canada (University of Ottawa) (both acting as senders) and a receiver located in Darmstadt are presented in Table A.2. Single experiments were repeated 5 times for each file from both locations each time with a different transmission bandwidth.

The decision to perform long-distance experiments was made because of the higher likeliness of packet losses. During preliminary experiments on the campus network and in between Germany no or only very few losses occurred.

For each LC-RTP session information about the retransmission was logged at the receiver and the original file and the transmitted file were compared to assure that the transmission completed

successful. The comparison for all tests was positive thus, all transmissions were finally made without any errors.

Table A.2: Test results (Bandwidth, Duration)

BW [kbit/s]	File Size [MByte]	Max. BW [Bit/s]		Duration [s]	
		NIST	Ottawa	NIST	Ottawa
1000	6	1047552	1022800	41	42
	20	1024048	1024000	160	160
2000	6	2147480	2045216	20	21
	20	2048104	2048000	80	80
4000	6	4294968	3904512	10	11
	20	1561080	4096000	105	40
8000	6	8593216	1169880	5	37
	20	8192008	1058392	20	151
12000	6	8589936	1213296	5	36
	20	5461336	487968	30	337

A.5.2 Test Results

The results obtained from the logging that was performed during the LC-RTP sessions shows the occurrence of retransmissions in almost all of the test. The logging information also confirmed that the number of retransmissions increases with the size of the bandwidth it has tried to send the files with. If the bandwidth is set much higher than the actual bandwidth of the link between sender and receiver multiple retransmissions for one packet are more likely. However, also in these cases the files were transmitted without any errors.

Table A.3: Test results FTP

File Size [MByte]	Max. BW [Bit/s]		Duration [s]	
	NIST	Ottawa	NIST	Ottawa
6	576000	328000	71	126
20	568000	304000	273	512

During the tests it also became clear that the quality of the link between the US and Darmstadt is of a higher quality than the one between Canada and Darmstadt. We also transmitted both files via FTP from both locations to Darmstadt to obtain some information about the performance of a traditional file transfer protocol. The comparison of the transmission times shows that, with LC-RTP,

data can be transmitted faster than with plain FTP. This is caused by the nature of the UDP-based transmission which does not, in comparison to TCP, back off in the case of congestion in the network. Thus, the performance of competing TCP traffic is affected by LC-RTP transmissions in the case of congestion. In Section 9.2, a TCP-friendly approach for LC-RTP is presented.

A.6 Summary

Caching and prefetching of A/V content are powerful methods to increase overall performance in the Internet. LC-RTP is a simple and efficient reliable multicast protocol compatible with the original RTP, which is stated by the experiment presented in this chapter. It needs to be implemented only in servers and caches. These servers have to be adapted to LC-RTP and they need mainly a list implementation, so the adaptation is a very simple procedure. Clients are not affected at all by LC-RTP.

All resources are used carefully and the extension permits an implementation to use a simple method to keep the sequential nature of the stored data without buffering. This method considers hard disk performance and possible network structures without wasting resources (like main memory and CPU power). Its intention is to allow a maximum number of concurrent streams handled by the caches. As no additional packets are sent during the regular session and the packet sizes are hardly bigger than those of an standard RTP sender, all access control mechanisms and network quality computations can remain unmodified. The only difference to a normal transmission is the fact that after the session, a retransmission of the lost packets to between server and cache with LC-RTP extensions is performed. A conforming, standard RTP receiver would recognize this as a normal session termination and, thus, would not be affected.

Multicast ensures a minimum load increase on the network, because the packets are sent only to members of the multicast group, during a transmission to a regular customer.

Appendix B: Preliminary Subjective Assessment

The main goal of the subjective assessment from Chapter 4 is to get an answer to the question: How do variations in a layer-encoded video influence its perceived quality? Since the answer of this question should have an influence on the work on retransmission scheduling, the main idea is to compare impaired sequences directly with each other. In general, two of the presented test methods from Section 4.3.3, double stimulus impairment scale (DSIS) and stimulus comparison (SC), are applicable in this case. With DSIS information about the influence of an impairment in comparison with a reference sequence can be obtained. Thus, the comparison of two impaired sequences can only be performed indirectly. For example, to compare two impaired sequences (sequence 1 and sequence 2) two DSIS tests have to be performed. In the first test, sequence 1 and the reference sequence are compared while in the second test a comparison between sequence 2 and the reference sequence is made. In contrast to DSIS, the SC method needs only one single test for the comparison of sequences 1 and 2.

Since it was not clear which of the two tests should be used for subjective assessment, the decision was made to perform a preliminary assessment in oder to find out which method is suited best. An additional goal of this test was to investigate, if the test candidates would recognize the quality changes that occur in the test sequences. Negative results on this investigation would have implied a modification of the test sequences.

B.1 Execution of the Preliminary Assessment

This assessment was executed manually, since the test application described in Section 4.3.4 should make use of the results gained in this investigation, and naturally could not be available at this point in time. (The implementation of the application should not start before the final decision on the test method was made in order to avoid implementing to methods.)

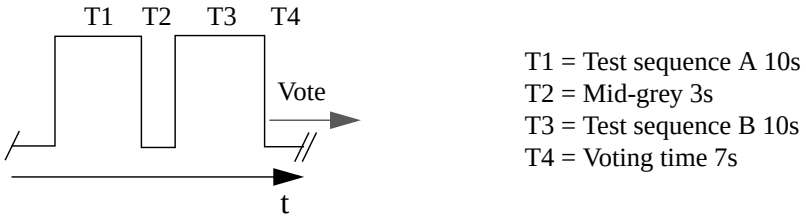

Figure B.1: Single test procedure

The procedure of the test was as follows: first of all the test procedure was explained to the test candidate. The questions and possible answers for each single test were given to the candidate. The candidate was given sufficient time to read and understand all questions and answers. After-

wards the actual test sequence was presented to the client with the time constraints as shown in Figure B.1. Figure B.2 shows the sequences used for the assessment.

After the presentation of the two sequences the candidate had to answer the according question in the next 7 seconds by marking his assessment. Six single tests were executed during the whole assessment in which 14 test candidates took part. The candidates should compare M&C 1 with M&C 2 and T-Tennis 1 with T-Tennis 2 using both test methods. With the DSIS test method for single tests are necessary while the SC method requires only two tests. A final comparison between both test methods seems to be valid, since equal content and quality variations were used for each test method. The way the candidates assessed the quality of the sequences is shown in the following two sections.

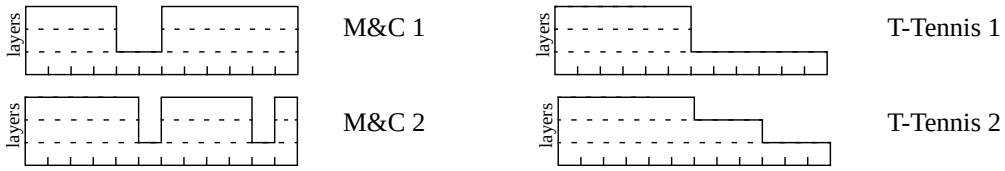

Figure B.2: Shapes for the preliminary assessment

B.1.1 DSIS Method

*The same video sequence will be shown to you twice. The quality of the two sequences **may** differ. Please, answer the following question (within 7 seconds) **immediately after the second sequence**.*

Did you perceive the second sequence in a worse quality than the first and if so was it annoying?
(You have to answer this question within 7 seconds)

	imperceptible
	perceptibel, but not annoying
	slightlty annoying
	annoying
	very annoying

B.1.2 SC Method

*The same video sequence will be shown to you twice. The quality of the two sequences **may** differ. Please, answer the following question (within 7 seconds) **immediately after the second sequence**.*

Mark on the shown scale (shown below), within 7 seconds after you have seen the second sequence, how you perceived the 2 sequences relative to each other.

Sequence 1 much better | Seq.1=Seq.2 | Sequence 2 much better

B.2 Selection of the Test Method

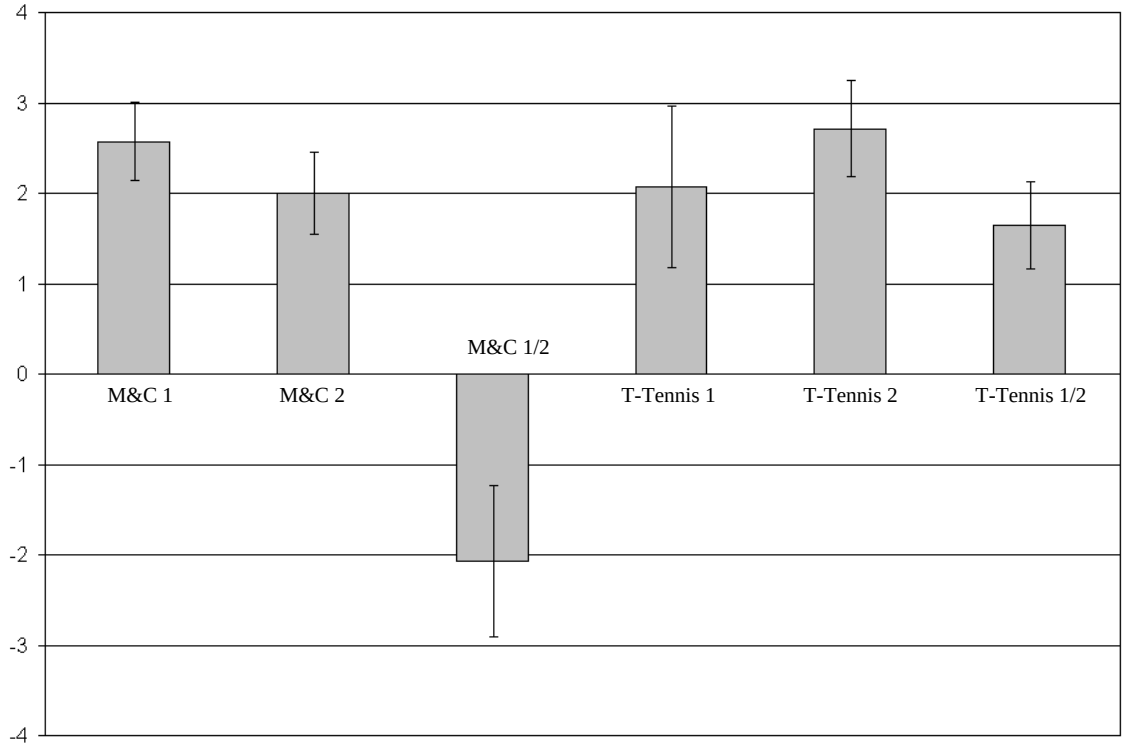

Figure B.3: Average and 95% confidence interval for the different tests of the experiment

The outcome of the statistical analysis (performed as described in Section 4.5) of this preliminary assessment is shown in Figure B.3.

The first positive results of the assessment is the significance of the statistical analysis. The tendencies that can be derived for the single tests met the expectations. At first, the tests with the M&C sequence as content for the experiment are discussed in more detail. The results of the statistical analysis (two DSIS and one SC) are shown in Figure B.3. M&C 1 shows the result of the DSIS method where the original sequence and sequence M&C 1 (see Figure B.2) had to be compared with each other, while M&C 2 shows the result of the comparison between the original and the M&C 2 sequence. M&C 1/2 presents the result of the SC test at which sequences M&C 1 and M&C 2 are compared directly with each other. The negative result of this test indicates that the test candidates assessed the quality of the first sequence (M&C 1) better than the second (M&C 2). In addition to the average result of the assessment also the 95% confidence interval of the T-test is shown. For M&C 1/2 the confidence interval is quite large but does not cross the neutral axis. Thus, it is of statistical significance. The same is true for the results of M&C 1 and M&C 2. The test candidates realized an impairment in the modified test sequences, yet the results for both test are quite similar. To be able to compare the results of both tests (M&C 1 and M&C 2) the difference between both results must be calculated. The confidence intervals of both tests are over-

lapping each other and, therefore, are not as significant as the result of the single SC test. If only the average values of M&C 1 and M&C 2 are regarded, a tendency similar to the result of the SC test (M&C 1/2) can be recognized.

Similar results were obtained for the second pair of sequences that should be compared with each other. While the result of the SC test method (T-Tennis 1/2) is statistically significant, the conclusions that can be drawn from the two DSIS tests (T-Tennis 1 and T-Tennis 2) do not allow such an interpretation. Regarding only the average of the assessments reveals an equal tendency for both test methods; T-Tennis 2 is assessed a higher quality than T-Tennis 1.

The initial goal of this preliminary assessment was to make a decision on which of the two test methods (DSIS or SC) should be used in further assessments. First the advantages of both tests are given in the following and then the decision for the SC test is justified.

- In comparison to the DSIS method the results of a statistical analysis based on the SC method can be statistically significant.
- The SC method is well tailored for the goal of this investigation: Two sequences, each having a different, impaired quality should be compared with each other directly.
- With the DSIS method the mean value of one test can be located in the boundaries of the confidence interval of the other (peer) test and vice versa. This effect can lead to an overall increase of the confidence interval and, thus, the possibility to obtain statistically significant results is reduced.
- Compared to DSIS the amount of context effects [162] is reduced. Thus, results of the SC method are more stable.
- The overall duration of the assessment is shorter with the SC method, since only half of the amounts of tests are required compared to the DSIS method.

The DSIS method has the following advantage:

- ITU-R BT.500-10 [129] states that the DSIS method is well suited for a comparison of the original and an sequence which impairment is small. For example, quality degradation is due to bit-errors occurring during the transmission of the video compared to quality degradations that are introduced due to an adaptive streaming.

In general, the goals of the two test methods are slightly different. While the DSIS method is used to assess more fine-grained impairment, SC aims on more general statements about the quality of a video sequence. This is also reflected by the two different scales (see B.1.1 and B.1.2) to assess the single tests. Nevertheless, this difference should not be a reason that influences the decision for one of the two tests, since both methods can produce meaningful results.

The decision to use the SC method was due to the amount of advantages it offers in comparison to the DSIS method. The most important fact is the possibility to obtain statistically significant results more easily than it would be the case with DSIS. The only benefit of the DSIS, assessing the quality of an impaired sequence in comparison to a full quality sequence, is not necessarily needed for the assessment that should be performed in this thesis. The fact that the duration of a complete assessment is doubled by the DSIS method should also not be neglected.

B.2.1 Content

In discussions that followed the assessment, test candidates stated the influence of the content on the way they performed their assessment. According to many of the candidates a sequence is assessed differently if the content of this sequence is watched for the first time compared to later assessments when the content is already known. This effect might be caused by the fact that new contact is distracting the test candidates. Thus, test candidates are concentrating on the content instead of the impairment of the sequences. To avoid this phenomenon, according to [162] initial sequences were included in further assessments (as in Section 4.4). Those test sequence make the user aware of the content but are not used for further statistical analysis.

Appendix C: A Toolkit for Dynamically Reconfigurable Multimedia Distribution Systems

C.1 Motivation for a Video Distribution Testbed

In recent years a substantial amount of work has been performed on investigations of wide-area audio and video (A/V) distribution in the Internet. Most of this work has been theoretical and simulative work on new mechanisms that reduce the consumption of network and server resources by streaming data in A/V distribution systems (see Section 3.3). During our work on transport mechanisms for such systems we realized that a testbed for implementing the mechanisms that are used on the data path is necessary in order to perform measurements and analysis. Therefore, the decision was made to build a toolkit for the testing of A/V streaming and distribution mechanisms (KOMSSYS [155]). With the knowledge that the development of this infrastructure would be performed as a research project at universities and, thus, supported by contributions from student projects (master theses), the system should be easy to extend, have reusable components and well-defined interfaces. Therefore, a toolkit was created that allows implementors to build prototypes for multimedia distribution systems. Such a distribution system prototype comprises simple applications that fulfil the basic functionality of video servers, video caches and clients. These applications can be built from the toolkit or they can be existing applications such as the RealPlayer [52] or the Quicktime Streaming Server [163]. The components of the toolkit provide abstractions for data and protocol handling functions that are required on the data path of such distribution systems. To allow interoperability with existing applications the networking components of the toolkit were built around a standard-based architecture that supports the existing standards RTP/RTCP [61], RTSP [65] and SDP [64].

Existing approaches for configurable distributed multimedia systems [164, 165, 166, 167, 168, 169] implement mostly data path components that are connected into graphs which remain unchanged while data is flowing, or they consider middleware frameworks that allow the specification of an end-to-end behavior for complex multimedia systems. In the latter kind of systems, functionality is described at the level of cooperating distributed components [170, 171]. Achieving network transparency is neither a goal nor a possibility of the toolkit. Firstly, third-party applications are included in the distribution systems, secondly, standardized and extended protocols for the data path should be investigated. These standards require support for reconfigurations of the data path. Reconfigurations are necessary because the data path can be influenced by control information that is implicitly contained in protocols and payload. In RTP, for example, packets

may arrive unpredictably from new sources, or MPEG transport multiplexes may contain other streams than expected.

In the following, the toolkit design, its performance aspects, and experience made during its use by the original developers and others who have implemented several distribution system mechanisms are presented. Experiences include the example of the gleaning distribution mechanism [4], which was implement by a master student [172] by adding just two new components for the data path. Performance measurements with both a monolithic and modular architecture for the toolkit showed that the performance penalty for the latter approach is only around 8%.

C.2 Terminology

Several terminologies have been used in the past to describe multimedia distribution subsystems. The *stream handler* terminology has been used for many years and is also adopted for the toolkit presented in this appendix. The example for a data path that is set up in a simple streaming server is used to explain several terms and their functionality as shown in Figure C.1. In this specific example three components are used that read video data (in this case MPEG 1 System) from disk, packetize it and add RTP header information, and send it out on the network. Based on this example, several terms are explain in more detail in the following:

Stream handlers (SH) are components that can be bound together dynamically by a controlling entity into a graph. A SH must either produce or consume data units, or data units can enter and

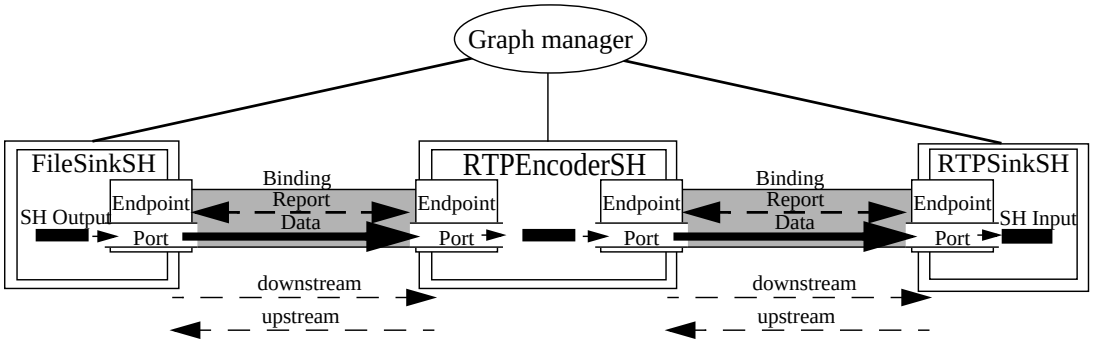

Figure C.1: Terms

leave the SH. Data units that enter the SH are ***SH input***, data units that leave are ***SH output***. A ***port*** of an SH is an interface for sending or receiving data units. An ***endpoint*** encapsulates ports of a SH. Endpoints provide the information that allows one to decide whether ports of two SHs can be bound to each other or not. Only if one SH's output port is bound to another SH's input port data units can be exchanged between the SHs. These ***bindings*** are logical entities in our case, implemented as function calls. The terms ***upstream*** and ***downstream*** are meaningful because the resulting graphs are directed. Upstream qualifies SHs that are bound to an SH's input port, downstream qualifies SHs bound to an output port.

The graph shown in Figure C.1 consists of 3 SHs, ***FileSinkSH***, ***RTPEncoderSH***, and ***RTPSinkSH***. In this example, data is only sent downstream from the ***FileSinkSH*** through the ***RTPEncoderSH*** to the ***RTPSinkSH***. Control data, e.g., RTCP info can be sent up- and down-stream

between the SHs. Data units are, e.g., video data that leave ***FileSinkSH*** as ***SH output*** and enter ***RTPEncoderSH*** as ***SH input***.

A ***stream graph*** (or simply ***graph***) is a set of connected SHs and the bindings between them. In graph terminology, we consider the SHs nodes and the bindings edges. Here, the movement of data units in stream graphs is directed and non-cyclic. ***Sub-graph*** describes a subset of SHs that are bound directly to each other but have unbound ports. A ***trunk*** is a sub-graph of a stream that has either only open input or open output ports.

The controlling entity mentioned in the SH definition is called the ***graph manager (GM)***. On behalf of an application, the GM creates a graph of SHs to form a media processing subsystem. The GM is responsible for the setup and destruction of the SHs, determines the interaction between the individual SHs and is the interface to the application. Since the focus of this work is on the data path of distribution systems, these GMs are application specific. They use predefined sub-sequences of SHs that are required for a specific task, such as data forwarding, writing to and playout from disk, buffering, or sequencing. The applications and the subsystem communicate only through this GM.

In the toolkit, a ***stream*** consists of data that is logically one entity and that is processed by a single stream graph. The same stream graph may handle several streams in parallel.[1]

C.3 Design

KOMSSYS, the streaming platform that makes use of the toolkit, offers the possibility to make practical experiences with various distribution system mechanisms. This requires that mechanisms that are implemented using the toolkit can achieve comparable performance to monolithic, specialized implementations. The toolkit must allow the creation of mechanisms in such a way that run-time performance is equivalent to a dedicated implementation. This goal requires also the ability to reconfigure the graphs that are built from the components of the toolkit, because distribution mechanisms require changes to the data flow that can be handled internally by monolithic implementations, but require reconfiguration in a graph of interconnected components.

Since the toolkit should allow other researchers to build prototypes for multimedia distribution systems, it must be easy to extend, have reusable components and well-defined interfaces. In Section C.3.3, it is described how the toolkit supports the implementor in building new prototypes.

C.3.1 Equivalent Design

The prototype applications that build on the basis of the toolkit should allow investigations on the performance of the data path. Therefore, the implementation should have a low influence on this performance, i.e., the toolkit should not force an implementor of a mechanism to build a solution that consumes more system resources than a monolithic implementation of the same mechanism. Such basic limitations would arise from a toolkit-defined threading model and memory model that are mandatory for all components. These influence fundamental issues in approaches such as the

[1] In the remainder of this chapter flow is used as a synonym for stream.

amount of queuing of data and control information, and the number of mandatory context switches. Besides the support for passing arbitrary structures of data between components after an initial negotiation of capabilities, the threading model and the handling of control information are large determined by these performance considerations.

A) Threading Model

The threading model of a SH system is concerned with the question, which entity is processing which amount of data at which time. Advanced, open middleware approaches that implement functions by concatenating functional modules into arbitrary graphs of independent components are able to attach scheduling mechanisms to arbitrary sub-graphs [173]. While this approach is highly flexible, it requires either an operating system abstraction layer to allow arbitrary grouping, or information about the potential grouping capabilities of modules. For example, it is problematic to support a module that listens to BSD sockets with a module that waits for the release of a POSIX semaphore in the same thread.

The toolkit consist of both, very fine-granular SHs with minimal functionality such as logically copying a data unit, and course-granular ones, such as an SH that controls a zero-copy kernel implementation [174] which handles everything from data retrieval from disk to sending onto the network. The first example makes it undesirable to have one thread per SH, the second requires that threading can be controlled by SHs. To fulfil both requirements the optional creation of threads inside each SH and also optional sharing of threads among SHs with assistance of the GM is supported.

To organize the creation of graphs from SHs that have various demands for concurrency, they can be referred to as active-capable, passive-capable, and through-capable SHs. The terms active, passive, and through are defined as follows:

- **Active:**
  Active SHs determine their own timing, usually by waiting for events of some kind, such as time-outs or packet arrivals. They can push data downstream actively (by calling a push function of the downstream SH) or they can pull data from an upstream SH, or both. Two active SHs cannot be connected directly because each one tries to control synchronicity. A passive SH must be inserted between them. The ***RTPEncoderSH*** in Figure C.1 acts as an active SH, it pulls data from the ***FileSinkSH*** and, based on its own timer, pushes it to the ***RTPSinkSH***.
- **Passive:**
  A passive SH does not determine timing. If it acts as a sink, an upstream SH may push data to it, if it acts as a source, a downstream SH may pull data from it. If it implements both source and sink, it must also provide buffering capacities that suit the needs of the graphs that it is likely to be included in. Either in-band reporting or GM notifications can be used to warn of over- and under-runs of the buffer. Passive SHs cannot be connected directly because no data would be exchanged between them. ***FileSinkSH*** and ***RTPSinkSH*** act as passive SHs in Figure C.1.

- **Through:**
  Through SHs are meant for tasks such as on-the-fly transcoding, packet duplication, or filtering. They do not generate timing and should not introduce buffers beyond those necessary for their operation. They must always implement a source as well as a sink. An arbitrary number of them can be concatenated. An active SH that is located upstream will push data through this kind of SH, potentially through several more through SHs until a passive SH is encountered. The pull operation is used in the same way by an active SH located downstream. Whether a through SHs can be used in push or pull mode is determined by endpoint settings. The matching function of the endpoints can restrict it to one direction. In Section C.4.4 an example for a through SH is given.

A SH may be capable of one or more of these operation modes. During the configuration or reconfiguration of a stream graph, the matching procedure must determine whether the desired sequence of SHs can be connected without violating these threading conditions. The mode of a SH that supports several or all of the three modes can only be determined when it is initially added to a graph, but it remains fixed during reconfiguration.

Independently from these modes, each SH is allowed to create as many threads internally as are necessary for its processing. Only if these threads perform data retrieval from upstream SHs or data delivery to downstream SHs, the SHs identify themselves as active SHs to the matching process that is initiated by the GM during configuration and reconfiguration. Otherwise, the GM remains oblivious to internal threads. This approach limits control over thread use in the system but it simplifies the integration of third-party modules, even closed-source modules (see Section C.4.5). To allow better control and a reduction of the number of concurrent threads when necessary, SHs can on the other hand offer an interface to the GM to provide them with a thread object, which can then be shared among several SHs at the discretion of the GM.[1]

A large number of other options were considered but discarded for of the following reasons. Using one thread per graph has the disadvantages of a single thread for the entire system but the additional problem of loosing control over the complete system. An approach that assigns one thread to each data packet wastes threads when processing is delayed, and it fails when packets are copied, split, or merged. Running each SH in a separate thread can encapsulate third-party software well, but a simple packet forwarding in a graph requires an unnecessary number of context switches. Special thread SHs provide good speed control to the GM but require that all other SHs are passive and adaptive to speed changes.

B) Feedback and Parametrization

SHs have been devised to ease the integration of streaming media into applications. The original assumption is that SHs are connected and configured at application startup, resource needs are negotiated for all relevant components, and the streaming is than handled by the SH graph trans-

[1] Optionally, also a single-threaded approach for simplified debugging only is supported, which is however not generically applicable to all SHs.

parently. State-of-the-art multimedia systems support resource-adaptive media processing, for example in the control-theoretical approach, where a set of SHs cooperates logically to form a control loop [175]. This requires the distribution of feedback information to the SHs that must adapt according to this information.

The toolkit allows two different types of feedback distribution, one that allows the exchange of feedback between SHs directly, and the other one between an SH and the GM. The direct exchange between SHs (in-band) saves resources and context switches when SHs generate a large amount of feedback information and the number of existing SHs in a stream graph is high. This mechanism does not involve a context switch. Nevertheless, there are cases when the application must be notified and, therefore, feedback information can also be sent to the GM (out-of-band). Acceptance of this feedback will usually require context changes.

In the in-band approach, each SH implements a return channel that takes control messages from a report interface and tries to interpret and process them. If they cannot be interpreted, they are forwarded to all SHs that are connected to the other side of the SH, i.e., to all upstream SHs if the report arrived from the downstream side and vice versa. If they can be interpreted, the SH may be able to complete the processing, it may decide to notify the GM, or it may forward a modified report to the upstream SHs. Since the GM has created the stream graph, it can change the parameters of the relevant SHs directly, bypassing potential intermediate SHs. The in-band approach is most appropriate for sets of SHs that are usually all present in a stream graph and that cooperate in a predefined manner, whether they are separated by other SHs or connected directly. The out-of-band approach is more useful for handling of exceptional situations, atypical configurations, or the delivery of feedback over machine boundaries, e.g., signaling information from RTSP that requires a modification of the behavior of one or several SHs.

An alternative approach, the handling of control information in an additional graph of SHs, was not pursued because of the complexity it would add. This approach would double the number of SHs that need to be managed and it would introduce cycles in the SH graph.

C) SH Performance

The SH architecture of the toolkit requires that data is forwarded between arbitrary SH endpoints. This implies a dynamic identification of data handling functions which is realized through the typical C++ approach of virtual functions. To evaluate the efficiency of this approach, a pipe of three dynamically plugged SHs is compared with a SH that hard-codes the same functionality in a single module. The experiments are performed on an unloaded system, but in multiuser mode. The consumed time is measured in CPU cycles using the Pentium-specific mnemonic RDTSC [176], the compiler is a GNU g++ 2.95.3 (with -O2) on Linux 2.4.14. The chosen encoding format is a dummy format that does not require disk access when data is “read from file” to avoid that disk reads falsify the measurement.

Figure C.2 shows the performance increase that can be achieved by merging a pipe into a single, specialized SH. In the pipe comprising three SHs, the RTPEncoderSH is an active SH, while the other two SHs are passive. When the streaming graph is active, the encoding function of the

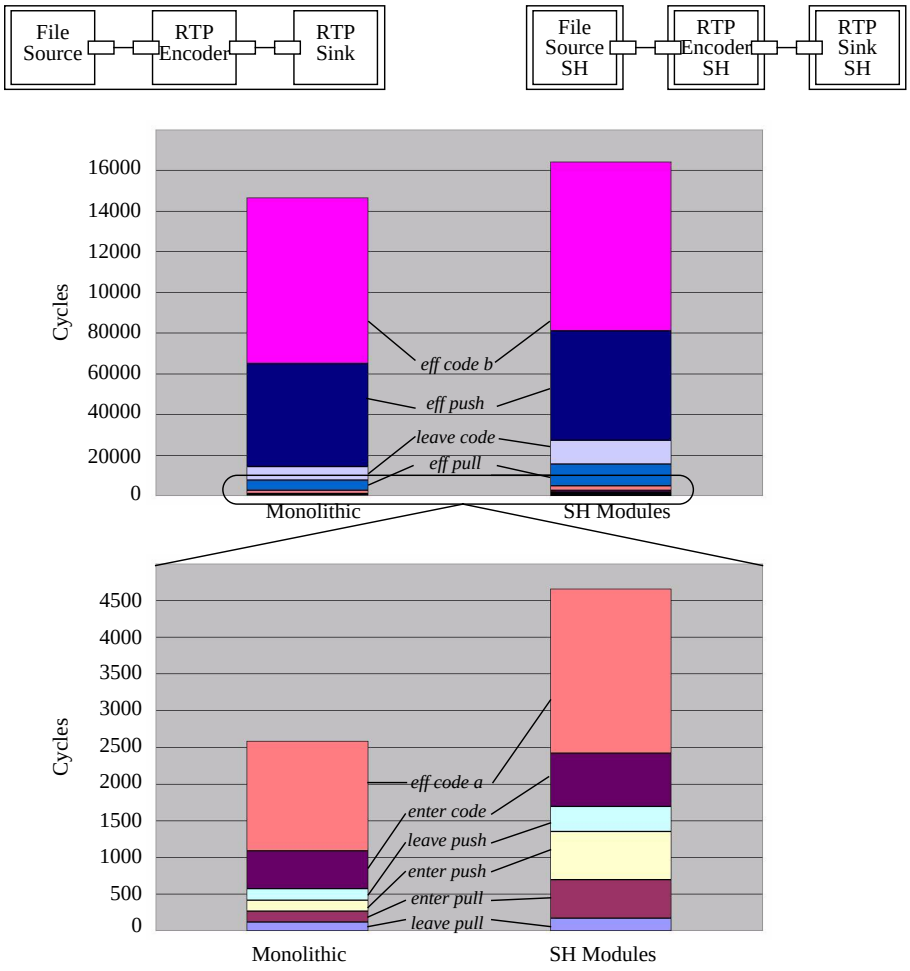

Figure C.2: Cycles

RTPEncoderSH calls a referenced, codec-specific object at times that are determined by the RTP packaging format for the specific encoding. This codec checks its prefetch buffers (*code a*) and if it needs additional data, pulls chunks of data from the FileSourceSH via the sink endpoint of the RTPEncoderSH and the source endpoint of the FileSourceSH (*pull*). Then, it processes the data chunk (*code b*) and returns it to the RTPEncoderSH. It is then sent to the RTPSinkSH via the source endpoint of the RTPEncoderSH and the sink endpoint of the RTPSinkSH (*push*).

The chart on the left side shows single function calls, while two virtual function calls and an indirection through source and sink endpoints are performed on the right side. The time for each call is denoted *enter code*, *enter pull*, and *enter push*, and it contributes an essential portion of the performance penalty on dynamic pipes of SHs. While this penalty was expected, it was unexpected that there is also a considerable penalty in leaving virtual function calls, at least partly due to conservative smart pointer handling.

The hard-coded SH saves time in three ways: by replacing virtual function calls with direct function calls, by increasing data locality due to collecting all member variables in a single object, and by implementing all functions within a single object instead of separate objects for SHs, SH

endpoints and codecs, and by additional smart pointer construction and destruction. The performance penalty of approximately 8% in a nearly empty pipe is acceptable when compared to the inflexibility of the integrated approach.

C.3.2 Streaming Graph Reconfiguration

In architectures that use coarse-granular SHs for the creation of stream graphs such as the one used for the toolkit, reconfiguration of active stream graphs is frequently a matter of reconfiguring the behavior of individual SHs rather than the graph itself. An example of the options that are available through this design is given by [175] in a system that is adaptive through feedback control loops.

Many applications do not need any dynamic reconfiguration of data paths and, therefore, do not require reconfigurable stream graphs. In the following, cases in which modification of individual SHs' behavior will solve many requirements are identified. Under the following conditions, reconfiguration of the graph is an interesting option:

- **Long life-time:**
  Long life-time of a stream graph makes the necessity of changes in the topology more likely.
- **Fine granularity:**
  Fine functional granularity increases the flexibility of graph creation but requires very complex graphs if the topology cannot be changed.
- **Unpredictable behavior of data sources and sinks:**
  If data sources, such as an RTP receiver, can produce data units that have other processing needs than earlier units, new functions have to be provided. Data sources can also change their behavior, e.g., by modifying the display speed or requesting a different data format.

The toolkit supports reconfiguration of the active stream graph, meaning that data continues to flow through the graph while sub-graphs are added to or removed from the graph. These operations are both time- and resource-critical. They are especially resource-critical because the disconnected but still active sub-graph may suffer from buffer over- and under-runs if it cannot be deactivated in time.

One reason is the support of RTP over IP multicast and, therefore, RTP packets from several sources may arrive at the same data port. In this scenario, it is up to the application to react to such a situation. If the decision is, e.g., to cache this unexpected data, a new trunk must be created downstream from the RTP receiver. Furthermore, a variable number of clients must be served, if stream scheduling approaches like patching [39] are implemented. In patching, new receivers must be able to join and leave an ongoing transmission. Since different flows cannot use the same active stream graph concurrently, graph configurations must be changed. The reason for not sharing graphs among several flows is the necessity of keeping per-flow information in SHs, such as encryption keys, file handles, encoding format choices, and many more. The implementation of

this approach is simpler if one graph per flow is used. Since sockets, semaphores, and SHs with optional threads are supported, several waiting threads exist in each graph. Therefore, this flexible configuration option was chosen.

This requires SHs that can offer endpoints with a variable number of ports that can be opened and closed dynamically. Each endpoint of an SH specifies the number of connections that it can support, and whether or not they can be connected and disconnected dynamically. To allow for fast connection and disconnection of trunks from graphs, it must be possible for the GM to check whether a graph is 'nearly' functional. With the toolkit, each endpoint of an SH can be matched, connected, and checked for completeness separately.

C.3.3 Implementor Support

The toolkit is meant to allow the building of a variety of prototype applications for delivery systems that are based on Internet standards. Due to the interaction of RTP and RTCP and the possibility of receiving data from several sources at a single port a directed, non-cyclic graph of modules is an appropriate streaming model.

On the most part, packet loss and duplication, delay and jitter as usual, but also packets from unexpected sources or from peers that should not be sending must be considered. The toolkit's infrastructure has to support dynamic reconfiguration of the data path, which influences the SH design, as well as the controlling framework, in order to deal with these (sometimes) unexpected behavior. The ability to reconfigure the graph dynamically is required, e.g., for the implementation of a cache: it is necessary to handle user interaction if that cache implements a conditional write-through mode. For example, the client decides to stop watching a video object while the caching process for this object should be continued. In this case, the subgraph (at the cache) that was used to forward the data to the client can be removed. In [177] dynamic reconfiguration of stream graphs is investigated to adapt to changing resource availability by reconfiguring SHs. Our requirements are orthogonal to these abilities: in our delivery systems, caches must be able to handle unexpected new streams from the uplink side, pause and continue requests from the client side. To perform this task the GM must split a graph or merge graphs on behalf of the application without disrupting the active data forwarding of a stream.

To allow management of the graphs, a layer of basic classes is given (see Figure C.3), that provides templates and interface definitions for the creation of new SHs. Parent classes with a set of virtual functions ensure the interoperability between SHs. The basic classes are the following:

- **SH:**
  SH must be inherited by all new SH classes. It provides all interfaces to the GM for configuration, notification, and status information.

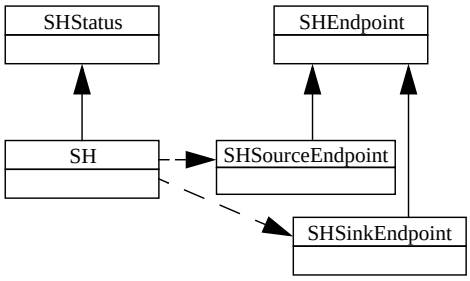

Figure C.3: Parent classes

- **Endpoints:**
  The Endpoint classes provide standard interfaces between the SHs. Each new SH must also include a class that implements its endpoints and inherits from SHEndpoint. SHs can provide both sink (SHSinkEndpoint) and source (SHSourceEndpoint) endpoints.
- **Attributes:**
  Attributes of a SH are modified by the GM to specialize a SH before it is connected into a graph.
- **Reports:**
  To implement in-band control reports are used that provide direct feedback in both directions along the data path. It allows notification between SHs without involvement of the GM. Each endpoint must provide report interfaces that are non-blocking. SHs may communicate via specialized reports even if intermediate SHs cannot interpret them (see Section C.3.1).
- **Notifications:**
  Notifications allows SHs to inform the GM of events, using a non-blocking notification function of the GM. A typical notification concerns the crossing of threshold in an SH that implements a queue which connects two active sub-graphs. Another use is the notification about RTP packets that arrive from unexpected sources.

C.4 Evaluation

In this section, the efficiency of the toolkit design is demonstrated by presenting different prototype applications. First, a simple client server application that is able to simply stream an MPEG-1 is shown. Subsequently, the implementation of a cache is presented that is able to either cache data on its local storage or serve a client if the requested object is already in its storage. To be able to measure the efficiency of the toolkit in terms of code reuse the *Frakes and Terry* metric is applied to the cache example. In addition, the implementation of a gleaning capable cache, which shows why reconfiguration is necessary is presented and the section is concluded with an example on the integration of third party libraries.

C.4.1 Client-Server Application

An example of the interaction between SHs is the delivery of an MPEG-1 (system stream) movie to a client. Figure C.3 shows the SHs that are used in this simple scenario. The movie is stored on the server's disk. Thus the starting point of the stream path is a *File Source SH*[1] that reads the data from the disk. In this example, data is requested from the *File Source SH* by the *RTP Encoder SH* which determines the timing in this stream. The *RTP Encoder SH* has knowledge about the actual encoding format of the data and the transport protocol that is used for data transmission. It determines time and amount of data to *pull* from the *File Source SH* and *pushes* it to the *RTP Sink SH* to meet the existing constraints for data rate and delay, and to create a reasonable stream. In the

[1] This is described as a source because it is the source of the stream path.

case of an MPEG-1 system stream this means that the *RTP Encoder SH* requests data chunks of equal size and *pushes* those to the *RTP Sink SH*. RTCP receiver reports are interpreted by the *RTP Sink SH* and statistics are forwarded to the *RTP Encoder SH* using the report interface.

The actual stream path is determined by the existing stream graph which represents the layout of the streaming architecture. In Figure C.3 the stream graph at the server consists of GM, *File Source SH*, *RTP Encoder SH*, and *RTP Sink SH*. The GM is responsible for the setup and destruction of the SHs, determines the interaction between the individual SHs and represents the interface towards the application. The stream handler functionality of the client is explained in Section C.4.5.

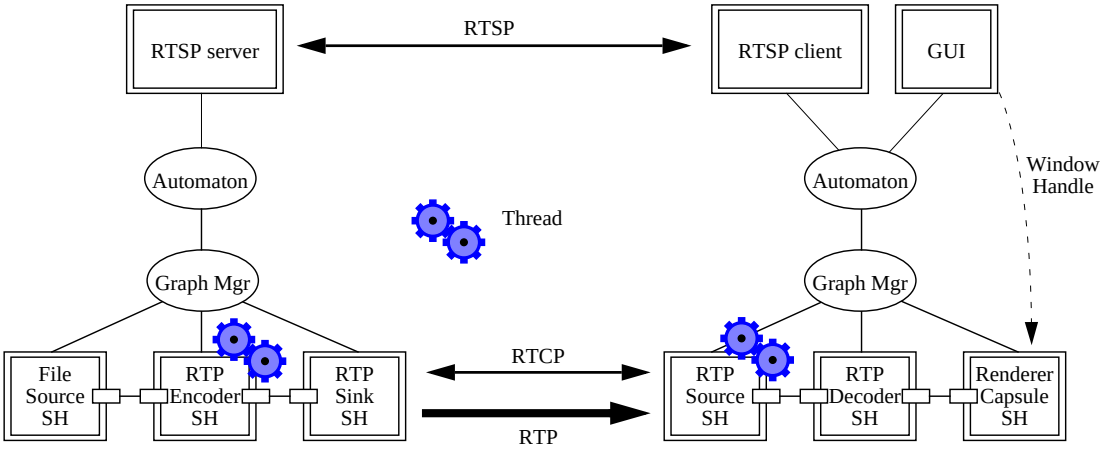

Figure C.4: Client server configuration overview

C.4.2 Cache

To enable caching functionality in KOMSSYS only one new SH had to be created. The task of this new SH is to create a copy of the RTP payload and send it on one path to the *RTP Decoder SH* and on the other to the *RTP Sink SH*. An overview of the used SHs and their relation is shown in Figure C.5. The new *Multiplier SH* was built by one of our students in the scope of his thesis [178]. In the stream graph for the cache, the existing SHs (*FileSink SH*, *RTP Decoder SH*, and *RTP Sink SH*) could be used without any modifications. Since the model for the GM is rather static, i.e., a GM cannot be created by, e.g., a configuration file but its characteristics are defined by the code that builds the graph manger, a new GM had to be created from scratch. The example of the cache demonstrates how, by the creation of a new SH, caching functionality could be implemented on the stream graph. It must be mentioned that, in order to have the complete functionality of a cache, additional modifications and extensions on the signaling protocol (RTSP) had to be made, but this is not related to the SH subsystem work that is presented here. The code reuse is formally evaluated in Section C.4.3.

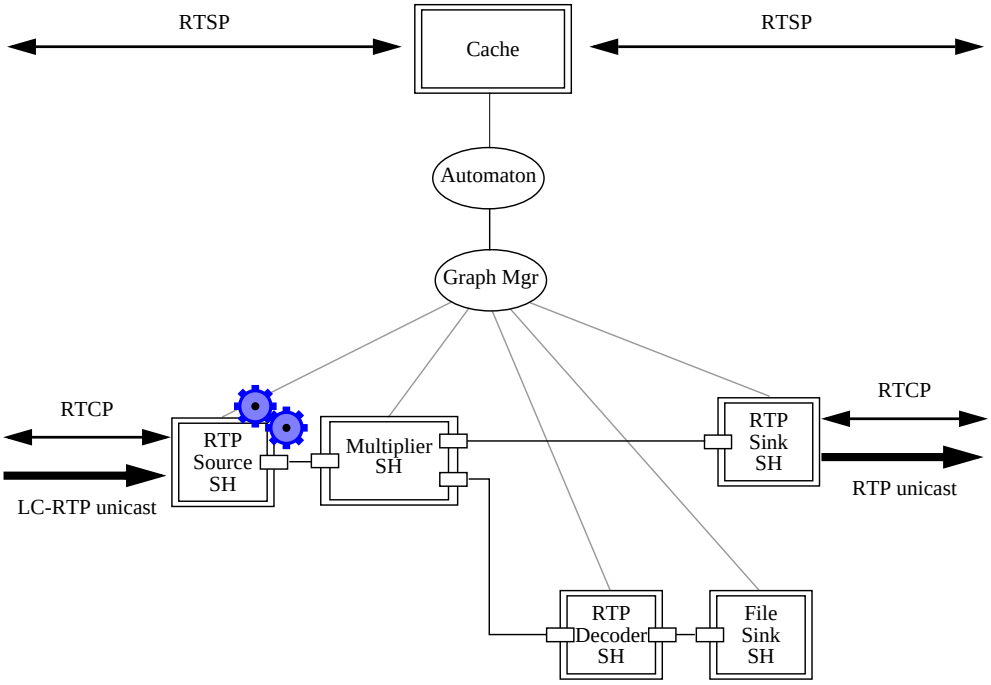

Figure C.5: Cache stream graph

C.4.3 Reuse

To measure the efficiency of the toolkit in terms of code reuse, two reuse metrics that are described in [179] are used, the object-based metric of Banker et al. [180], and the "Frakes and Terry metric with adjustment for complexity" [181], which considers lines of code as well. These metrics are chosen because they do not require a development cost factor, which cannot provide for components of the toolkit. The reuse metrics for the cache, the gleaning capable cache, and the clients are shown in Table C.2. Table C.1 gives an overview of the code statistics of these applications. Since C++ is used, the number of objects differs from the number of classes that are implemented, so values for both objects and classes are provided. The measures used in this work are shown in Figure C.6.

In the measurements presented here, only code on the data path that is composed from SHs and the controlling GM is considered. Other components like RTSP functionality which is also necessary to create streaming applications are not considered.

The first row of Table C.1 show the code statistics for the cache example, while the reuse metrics are shown in the first row of Table C.2. As Figure C.5 shows clearly, the graph of SHs that is required for the caching functionality requires only five SHs and their endpoints. For this reason, the reuse level in terms of classes and objects is low, while the high reuse level in lines of code shows that only little additional code was required to build the *Multiplier SH*. The second line considers the existing codecs for RTP payload types, as well as the newly written graph manager

for the cache. This increases the number of existing objects considerably but due to the size of the graph manager class, the reuse level that considers lines of code is not increased.

Banker et al.:

$$\text{reuse percentage} = \left(1 - \frac{\text{new objects built}}{\text{total objects used}}\right) \times 100\%$$

$$\text{reuse leverage} = \frac{\text{total objects used}}{\text{new objects built}}$$

Frakes and Terry:

$$\text{reuse level} = \frac{\text{objects above ITL + objects above ETL}}{\text{total objects used}}$$

ITL - internal treshold level

ETL - external threshold level

Figure C.6: Reuse measures

Table C.1: Code statistics

Application	existing objects	new objects	existing classes	new classes	exising LOC	new LOC
Cache SH pipe only without codecs	4	1	11	3	2271	218
Cache with graph manager and codecs	22	2	36	4	5130	291
Gleaning cache SH pipe only without codecs	6	1	17	3	3506	435
Gleaning cache with graph manager and codecs	13	2	42	4	6365	947
mpeglib support SH pipe only without codecs	3	10	10	6	1690	1082
MPlayer support SH pipe only without codecs	2	1	7	2	2176	273
Gstreamer support SH pipe only without codecs	3	1	10	3	1690	814

Table C.2: Reuse metrics for specific applications

Application	Banker et al reuse leverage and percentage				Frakes and Terry ITL=0, ETL=0	
	objects		classes		classes	lines of code
	leverage	percentage	leverage	percentage	reuse level	
Cache SH pipe only without codecs	5	80	4.667	78.57	0.7857	0.9124
Cache with graph manager and codecs	12	91.67	10	90	0.9	0.9463
Gleaning cache SH pipe only without codecs	6	85.71	5.667	85	0.85	0.8896
Gleaning cache with graph manager and codecs	6.5	86.67	10.5	91.3	0.913	0.8705
mpeglib support SH pipe only without codecs	1.3	23.08	2.6	62.5	0.625	0.6097
MPlayer support SH pipe only without codecs	2	66.67	3.5	77.78	0.7778	0.8885
Gstreamer support SH pipe only without codecs	3	75	3.333	76.92	0.7692	0.6749

C.4.4 Gleaning Cache

Reconfiguration plays no role in the example of Section C.4.1 but it is a basic requirement for a cache that implements *gleaning*. Roughly, a gleaning cache works by delivering a movie linearly to a client via unicast, which the cache itself receives in two pieces: a short start sequence via unicast and the remaining portion via multicast. For a detailed description of gleaning the interested reader is referred to [4].

A detailed implementation design of a gleaning cache can be found in [172]. The cache is not an RTSP proxy as understood in the RFC [61], which caches and redirects only control information. Rather, it is an RTSP/RTP proxy cache that stores content in addition to handling RTSP requests. RTSP messages from different RTSP sessions are multiplexed onto one connection between a server and a cache. RTSP Session IDs are the keys to de-multiplex sessions. A cache installs an RTSP connection to a server on-demand when a request for the particular server is received from a client. The connection is torn down when no more active RTSP sessions between cache and server exist.

There exist two possible situations that require dynamic reconfiguration of the data path. If the cache does not keep the entire movie, a second client must be served from the same multicast

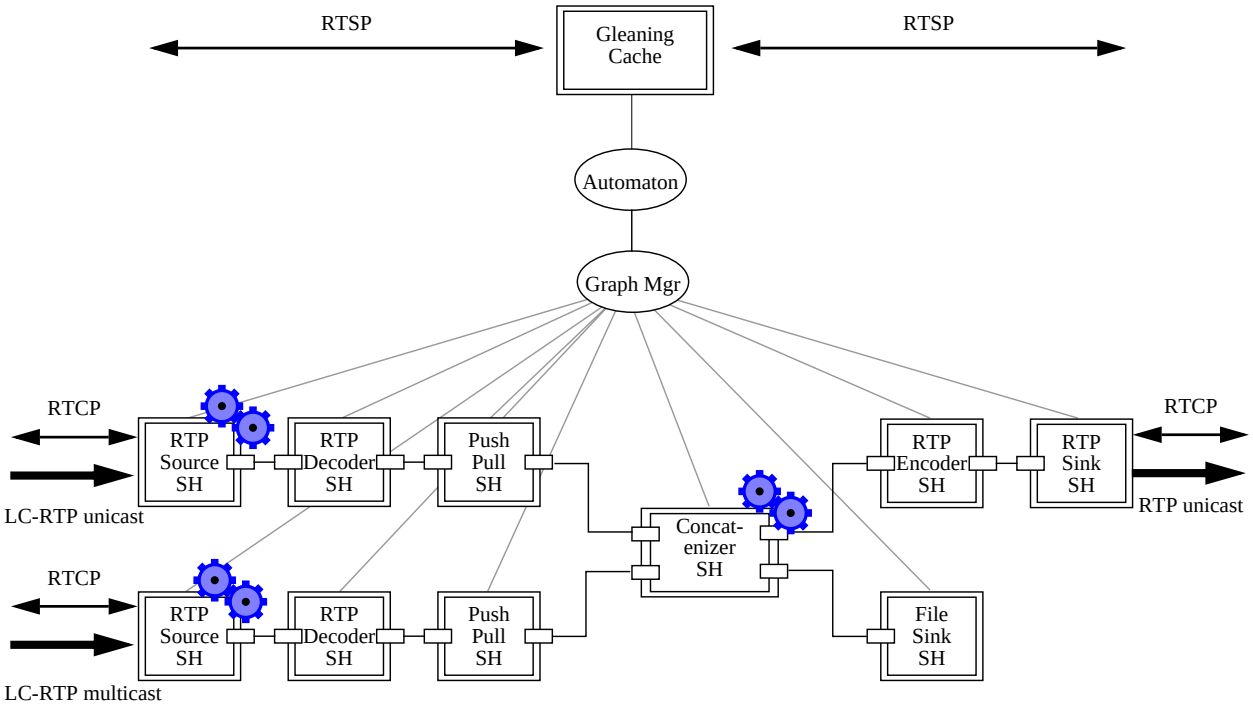

Figure C.7: Gleaning cache stream graphs (caching)

stream and an additional unicast stream, and if the cache keeps the entire movie, the client may decide to pause. In this case, the delivery path to the client must be suspended.

Figure C.7 and Figure C.8 show two possible stream graphs for the gleaning cache with a single client. When a client requests data from the server through the cache, the gleaning cache is often able to join an existing multicast session and request only the missing part of the movie via a unicast stream. When the missing part arrives in a unicast transmission, this data is immediately forwarded to the client while the multicast stream is buffered cyclically and streamed to the client after the unicast stream is finished. If the cache also decides to cache this movie (Figure C.7), both streams are stored linearly on its local disc. If a second client requests the same stream later on, a new graph for playing from the cache is created and reconfiguration is not necessary. In case the cache has decided not to cache the title and the server decides to deliver only a patch stream for this session, the same multicast can be used for the client, but another patch stream has to be requested. The light grey path in Figure C.8 is dynamically created in this case.

If the client pauses and the gleaning cache application decides to continue the caching operation, the trunk of the graph that forwards data to the client must be cut, while the trunk that stores data on disk must be maintained. If the client resumes viewing later, the application must create a new graph, which retrieves the data from the cache. Therefore, two stream paths on the receiving part of the cache are needed: one for the unicast stream and one for the multicast stream. These paths consist of an active *RTP Source SH*, a passive *RTP Decoder SH*, and *PushPull SH*. The latter is needed because the *Concatenizer SH* is active. This is the case because it determines both the order and the timing with which data is forwarded to the client or to the local disc. The main

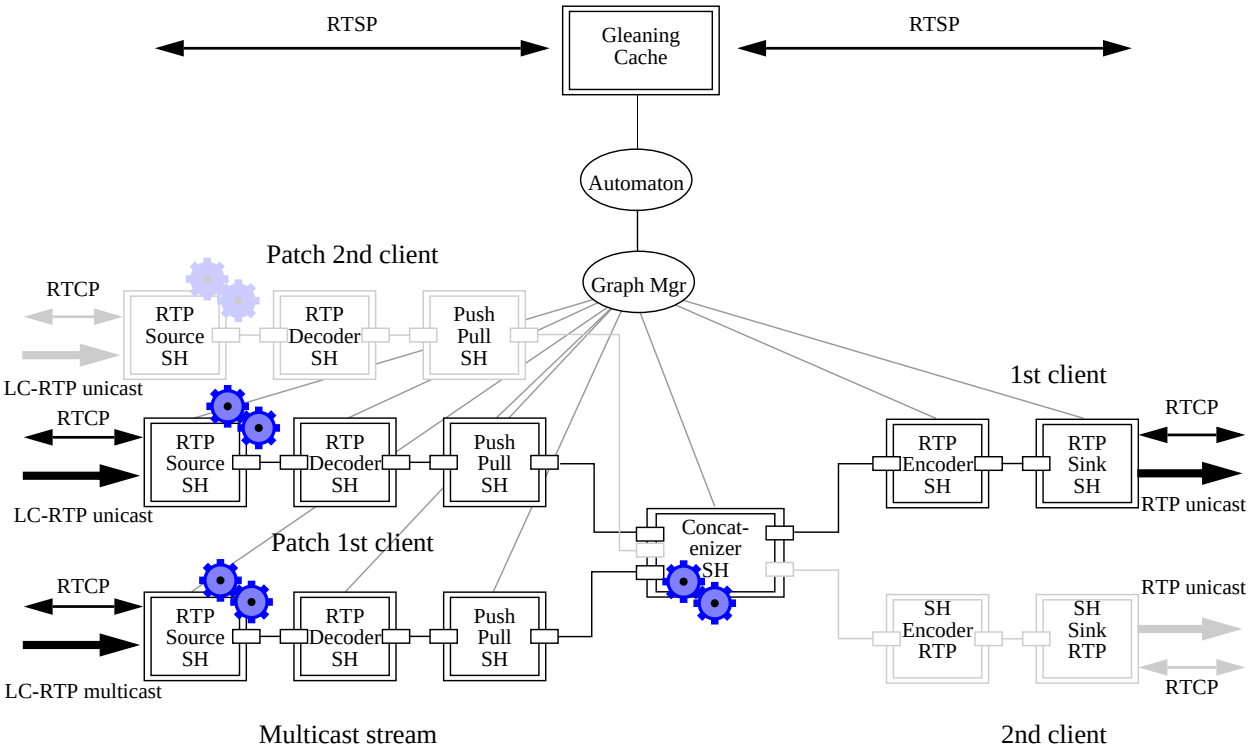

Figure C.8: Gleaning cache stream graphs (two clients)

task of the *Concatenizer SH* is to serialize the two incoming streams (multicast and path) and forward them to the client. Data from the multicast stream will be buffered as long as the patch stream is active.

On the data forwarding path to the client, an *RTP Encoder SH* can be seen in through mode, in contrast to its use in active mode in Figure C.3. If active mode is used instead, as in the previous example, the *Concatenizer SH* and the *RTP Encoder SH* would have to be separated by another *PushPull SH*, and both would re-create the required timing of the RTP stream independently.

Also this implementation was performed by a student [172]. The two new modules that had to be created from scratch for the data path are *Concatenizer SH* and *Patch GM*. All other SHs that were needed (shown in Figure C.8) to realize the *gleaning* functionality existed already and could be reused in the stream graph. The code statistics and the reuse metrics for the SH and GM are shown in Table C.1 and Table C.2, respectively. Although the reuse level in terms of objects is much higher than in case of the simple cache (see Section C.4.2), the reuse level in terms of lines of code is lower.

C.4.5 Third Party Libraries

In this section, an example is given that demonstrates how third-party software can be combined with the toolkit. This approach was used to create several KOMSSYS clients. The variety of decoding and presentation options for MPEG-1 system streams is a good example of the architec-

ture's flexibility. Currently, three alternative tools that can be integrated into stream graphs to perform this task are available with KOMSSYS.

The three tools are the KDE2 library *mpeglib* [182], the stand-alone playback program *mplayer* [183], and the GNOME streaming subsystem *gstreamer* [184]. Figure C.9 depicts the stream graphs that can be set up at the client in the case that it receives a simple unicast MPEG 1 system stream. As in the examples above *RTP Source SH*, *RTP Decoder SH*, and (in case of *mpeglib* and *gstreamer*) *PushPull SH* are reused.

The new SH that had to be built for *mpeglib* is the *MpeglibSink SH*, which works as a wrapper for the library. The *mpeglib* decoder pulls according to the timing that it parses from the arriving MPEG stream. It blocks the SH's thread to maintain this timing.

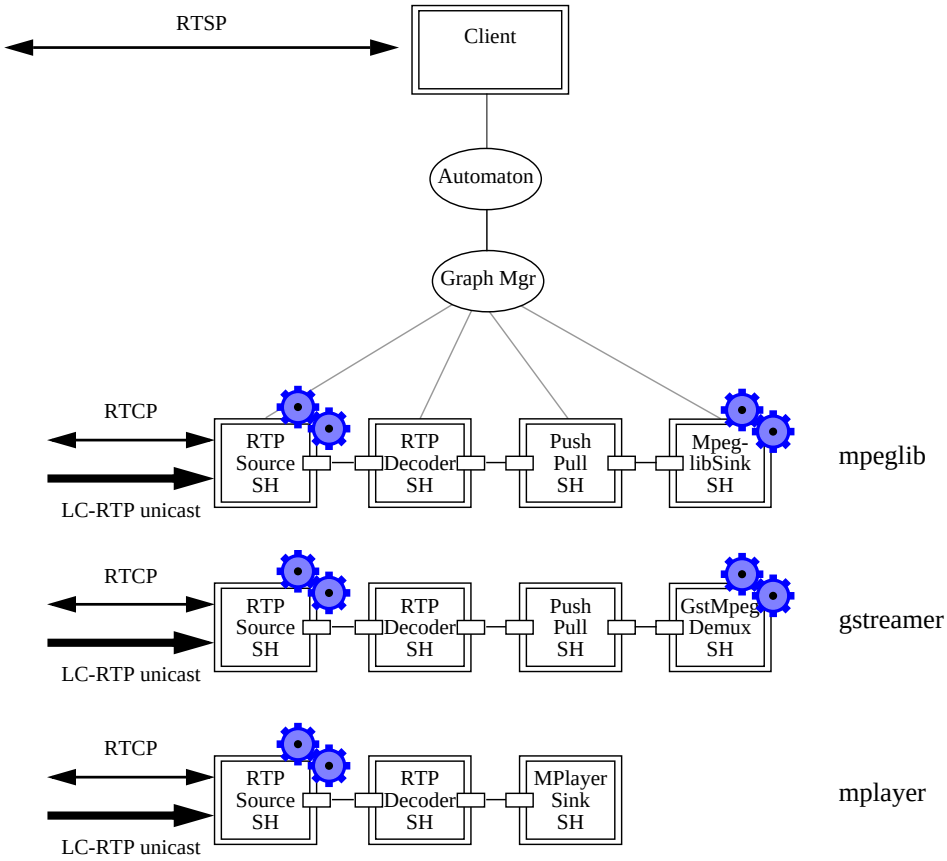

Figure C.9: Client stream graph

A similar graph is used for *gstreamer* integration, where an entire open-ended pipe of *gstreamer* SHs is encapsulated by the active *GstMpegDemux SH*, and the SH sink endpoint implements a *gstreamer* sink pad. In this case, *gstreamer* comes with a co-thread package that is not compatible with POSIX threads and, thus, must be encapsulated and hidden by the SH.

In the third approach, the stand-alone executable *mplayer* is encapsulated by the *MPlayerSink SH*. The SH forks to start the player, and forwards data to it through a Unix socket. Since this works in push mode, the *PushPull SH* is not required in this configuration. Thus by simply creating a different stream graph, three different MPEG 1 decoding libraries can be used in the client.

Code statistics and reuse levels of the three SHs can be found in Table C.1 and Table C.2.

C.5 Summary

The KOMSSYS platform is constantly extended to develop research prototypes in the area of wide-area distribution systems for streaming media in the Internet. We consider the platform an appropriate abstraction for developing streaming applications, especially since a lot of recent implementation work was done by students ([185, 178, 172]) who were not involved in the actual SH design. Based on the results presented in Section C.4.3 one can claim that KOMSSYS is easy to extend, for newcomers, as well as the original developers.

Two special requirements are needed to perform investigations in VoD distribution infrastructures: the ability to integrate third-party software and the ability to share resources among user-initiated session. The first requirement prevented, e.g., the use of co-routines and the limitation to a single mechanism for awaiting events. The second requirements led to the intention of reconfiguring the stream graph dynamically. The resulting toolkit is most similar to the InfoPipe [164] design, although InfoPipe does not support dynamic reconfiguration of the data path. One central difference is that InfoPipe components use a special co-routine model while with the toolkit used for KOMSSYS native threads are used. The presented approach simplifies the integration of third-party libraries and tools. The InfoPipe approach is more efficient but it requires conformance to a programming model.

In KOMSSYS, the initial code base considered mainly the distribution of CBR MPEG-1 system and MP3 streams in a caching hierarchy and to receiving clients. From the start of the development, the integration of third-party software was an important element, e.g. alternative streaming subsystems are the commercial servers VideoCharger and RealServer. The initial formats were chosen because they combine hardware- and OS-independent playback capability with an appropriate quality. More recently, H.261 and VBR MPEG-I audio, video were added.

The need for dynamic data paths was not an original requirement. The intention of building a gleaning prototype led to a redesign, which resulted in the decisions that are presented in Section C.4. An important observation is that the new design does not increase the number of context changes or the buffer requirements for the data path if one of the simple data paths is configured. On the contrary, the attachable threads extension allows SHs to get rid of some threads that were not exposed to the interface before. The reason is that attachable threads can be shared between SHs that are separated by a sub-graph that must use an incompatible mechanism to wait for events. An example is the RTCP thread of the RTP sink SH, which appears as a passive SH to the GM. Its thread, not exposed to the GM, is responsible for receiving and processing control messages.

Although the applicability and extensibility of the approach has been shown with the gleaning cache prototype, a better handling by integrating the SH approach consequently into the client was achieved recently.

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