Improving Efficiency and Enhancing Utilizability of Media-on-Demand Systems

Dem Fachbereich 03 — Mathematik/Informatik —
der Universität Bremen
zur Erlangung des akademischen Grades eines
Doktor-Ingenieurs (Dr.-Ing.)
eingereichte

Dissertation

von
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aus
Berlin, Deutschland

Datum der Einreichung: 2008/07/01

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Tag der mündlichen Prüfung: 2008/09/05

Abstract

Nikolaus, Boris:

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Media-on-demand is a method which extends traditional media transmissions by an important service: While consumers depend on the schedule of broadcasting organizations in case of traditional media transmissions like television or radio, they are granted the possibility to select a media stream from the offer of the provider and request it at arbitrary time in case of media-on-demand.

Certainly, this surplus service creates additional requirements to the sender and receiver systems and to the network between these systems, especially when a large number of receiver systems are simultaneously active. To keep these requirements low, many advanced media-on-demand systems and transmission techniques have been proposed in recent years. Besides of following different approaches, these systems differ by their goals and focus, their efficiency and utilizability, and each of them brings its own advantages and drawbacks with it.

The aim of the present thesis is to improve the efficiency of media-on-demand transmissions and to enhance the utilizability of media-on-demand systems. To achieve these goals, many media-on-demand systems have been analyzed, and from the gained experience and knowledge, a new and improved system has been developed. This new system distinguishes itself from existing systems by incorporating nearly any of the advantages and extensions of the other systems while operating at an efficiency near to the theoretical optimum and by providing a high flexibility and robustness at the same time. To prove the practical benefit of the proposed system in real environments, an embedding into an existing real-time transport protocol has been proposed and a comparison of the cost-effectiveness opposed to traditional media-on-demand systems has been provided.

Media-on-Demand, Video-on-Demand, Interactive TV, Media Streaming

Zusammenfassung

Nikolaus, Boris:

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Steigerung der Effizienz und Verbesserung der Einsetzbarkeit von Media-on-Demand-Systemen

Media-on-Demand ist ein Verfahren, das traditionelle Medienübertragungen um einen wichtigen Dienst erweitert: Während bei traditionellen Medienübertragungen wie z. B. Fernsehen und Radio die Sendeanstalten das Programm festlegen, ist es dem Konsument bei Media-on-Demand möglich, einen Medienstrom aus dem Angebot des Anbieters auszuwählen und zu beliebiger Zeit anzufordern.

Dieser Mehrwert erzeugt natürlich zusätzliche Anforderungen an die Sende- und Empfangssysteme und an das Netzwerk zwischen diesen Systemen, insbesondere wenn eine große Anzahl von Empfangssystemen zeitgleich aktiv sein kann. Um die Anforderungen möglichst gering zu halten, wurden in den letzten Jahren viele hochentwickelte Media-on-Demand-Systeme und -Übertragungstechniken vorgestellt. Neben der Verfolgung verschiedener Ansätze unterscheiden sich diese Verfahren auch in ihren Zielsetzungen, Schwerpunkten, ihrer Effizienz sowie Einsetzbarkeit und bringen unterschiedliche, systemimmanente Vor- und Nachteile mit sich.

Die vorliegende Arbeit hat die Zielsetzung, die Effizienz von Media-on-Demand-Übertragungen zu steigern und die Einsetzbarkeit von Media-on-Demand-Systemen zu verbessern. Um diese Ziele zu erreichen, wurden eine große Anzahl bestehender Media-on-Demand-Verfahren analysiert und mit den daraus gewonnenen Erfahrungen und Erkenntnissen ein neues, verbessertes System entwickelt. Dieses neue System zeichnet sich dadurch aus, dass es nahezu alle Vorteile und Erweiterungen der bestehenden Verfahren in sich vereint, eine Effizienz nahe dem theoretischen Optimum besitzt und zugleich äußerst flexibel und robust in seinem Einsatz ist. Um den praktischen Nutzen des Systems in realen Einsatzgebieten zu belegen, wurde zusätzlich eine Einbettung in ein existierendes Echtzeittransportprotokoll vorgestellt und in einem Kostenvergleich die wirtschaftliche Rentabilität der von traditionellen Media-on-Demand-Systemen gegenübergestellt.

Media-on-Demand, Video-on-Demand, interaktives Fernsehen, Medienstromübertragung

Acknowledgments

This thesis is the result of the research I carried out from 1998 to 2007 on the topic of media-on-demand. Although I am the sole author of this thesis, many people have contributed to it, to my education and to my live, and it is my greatest pleasure to take this opportunity to thank them.

First and foremost, I would like to express my greatest gratitude to Prof. Dr.-Ing. Jörg Ott. After receiving much guidance from him when I wrote my diploma thesis at the Technische Universität Berlin, I am very pleased that he decided to continue attending my academic career. Although living about 300 km apart (and since 2005 even around 1100 km), he managed to meet me countless times (while being always available by telephone) for our innumerable and fruitful discussions which not only brought new ideas and approaches into view but also delighted and motivated me much to continue my work on this long and sometimes stubborn path.

Furthermore, it gives me great pleasure to thank Prof. Dr.-Ing. Ute Bormann and Prof. Dr.-Ing. Carsten Bormann for participating in many enlightening discussions and for their organizational and professional guidance to a successful thesis.

Additionally, I would like to thank Prof. Dr.-Ing. Ute Bormann for accepting supervision of this thesis and Prof. Dr.-Ing. Jörg Ott for taking the role of the second thesis advisor.

There is one other friend I owe special thanks: During countless meetings, I got an invaluable help from Dipl.-Inf. Frank Tscheuschner by proofreading my publications and parts of this thesis, making critical remarks and useful suggestions and besides this for improving my English. I also would like to offer my thanks to Dipl.-Ing. Stefan Schwabe, Dipl.-Ing. Martina Liggesmeier, Dipl.-Ing. Tobias Poschwatta and Dipl.-Wirtschaftsinf. (FH) Christian Zanoth for proofreading this thesis.

Finally, I would like to thank my parents and especially my wife for their patience, their encouragement and all the support they gave me to perform my researches and to successfully complete this thesis.

Contents

List of Figures

List of Tables

Chapter 1

Introduction

SINCE the invention of the first electromechanical television system in 1884 by Paul Gottlieb Nipkow, it took a long time until television became a full grown-up technique: The first public demonstration of a working television system was not performed before 1925, and the first fully electronic system which was able to transmit open air captures went online in 1932 [Wik08c].

From this time, the number of television (TV) households was steadily increasing, e. g. passing the 10 % border in 1951, 50 % in 1954, 90 % in 1962, 98 % in 1978 and 98.2 % in 2002 in the United States (US), and it is estimated that more than one billion TV sets have been sold worldwide until 2005 [Tvh05, Usc05a, Man05].

TV systems get most attraction from supplying entertainment and serving recreational purposes, e. g. by transmission of movies, TV series, studio programs and sporting events, but also for providing up-to-date information, e. g. daily news, stock market information or weather forecast. But in addition to these two main types of broadcast, media transmissions can also be used for education, research or home shopping. Even democratic rights like freedom of speech and free political forming of opinion can benefit from media transmissions if the content providers and broadcasting organizations are politically independent.

Concurrently to the spread of TV systems, technical enhancements in transmission techniques have been encountered. For example, progresses in high frequency electronics and broadcast networks allowed to increase the number of channels by a magnitude: While a US household was able to receive an average number of 18.8 TV channels in 1985, this number increased to 92.6 channels in 2005 [Man05]. Advancements in digital video compression of the last recent years may increase this number to several hundreds in near future.

But even if the number of channels increases, not all consumers can be satisfied because transmissions are still scheduled at times which are laid down by the broadcasting organizations. The only choice which is left to the consumer is the selection of the channel he wants to tune in and

— for the rare case of request programs — the voting for one movie from a very limited list of typically less than five movies he would like to see soon.

Although the broadcasting institutions optimize their program to serve an audience as large as possible, most viewers demand a higher flexibility: They want to select themselves *what to see* and the time *when to see* it. Using video cassette recorders (VCRs) for delayed playback of transmissions is one popular solution for this problem (91.4 % of all US TV households owned a VCR in 2002 [Usc05a]), renting movies at video stores just another. But in all cases the possibility for a playback *on demand* is very limited: For the VCR solution, the customers have to wait for the next transmission of a movie if it is not contained in their home video archive, and to get a movie from a video rental store they have to drive to the store (or request a postal or messenger delivery if offered) and depend on fortune that the favored movie is currently available.

But with the growing bandwidth of broadcasting networks (e. g. satellite and cable TV networks) and home Internet access (e. g. based on DSL techniques), better solutions are conceivable: The viewer is sitting at home in front of her TV system, selects the movie she wants to see by her remote control, and when she made her choice, the movie is transmitted almost immediately to her TV system.

This idea — requesting the playback of a selected audio-visual stream at a consumer chosen time — is generally known as **video-on-demand** or **media-on-demand**. Many providers offer media-on-demand services today, probably a result of the convergence of audio, video and data onto a single network (TriplePlay, QuadruplePlay, TV-over-DSL), but these services are usually based on simple stream transmissions for each active customer which require a lot of bandwidth and server resources. The goal of this thesis is to examine the advantages and drawbacks of this simple solution and of alternative techniques and to propose enhancements and improvements to increase efficiency and extend utilizability of media-on-demand transmission systems.

1.1 Media-on-Demand

To provide a definition for media-on-demand, the search has to be routed back to the more ancient terms video-on-demand and interactive television and to the term media. Unfortunately, there is not a single, exact definition for **video-on-demand** available: Depending on the consulted encyclopedia, different aspects of video-on-demand are outlined to distinguish video-on-demand from other video transmitting systems.

For example, Wikipedia [Wik08e] currently defines video-on-demand systems as

> *[. . .] systems [which] allow users to select and watch video and clip content over a network as part of an interactive television system. [. . .]*

whereas video-on-demand is defined in the Computer Desktop Encyclopedia [Com07] in a more general way which does not rely on a network for the transmission of video streams:

> *Video on Demand: The ability to deliver a movie, sports event or other video program to a TV set whenever the customer requests it. [. . .]*

Although this would mean that a "call-a-video" service (i. e. a video rental service which accepts request via telephone and supplies the requested videos through a delivery service) has to be entitled as video-on-demand service, video-on-demand is used in this thesis only in the context of non-physical video delivery techniques.

Interestingly, some commercial companies even define video-on-demand in such a way that the definition incorporates their main business. For example, Cisco [Cis05] defines video-on-demand as

> *A system that uses video compression to supply programs to viewers when requested, via ISDN or cable.*

limiting the type of network just to ISDN and cable and making video compression a requirement for video-on-demand.

Condensed, video-on-demand means that a *video* is supplied *on-demand* with no or short service delay to the viewer for playback using non-physical transport. Accordingly, a video-on-demand system describes a setup of one (or sometimes several) video-on-demand sender systems, (typically) several video-on-demand receiver systems and a network which allows the transmission of video streams from sender to receiver systems and of control data between sender and receiver systems (sometimes but not necessarily in both directions).

Interactive television is a term which is often used in the context of video-on-demand although this term describes a more general type of interactivity: As currently defined in Wikipedia [Wik08h],

> *Interactive television describes a number of techniques which allow viewers to interact with television content [. . .].*

By this definition, the interactivity supplied by interactive television systems is not only limited to the selection of the content but includes also actions *after* the playback has started. Typical examples for interactive television which exceed video-on-demand are view angle selection, voting in TV shows or TV-based ordering of product samples when presented in commercials.

Recognizing that video-on-demand systems are a subtype of interactive television systems allows delimiting them from **content distribution systems**: For content distribution systems, the content is transmitted to the receiver system at arbitrary times (depending on the schedule of the

broadcasting organization) and is used some time after the transmission completed. For video-on-demand systems, the content is played as result of a request from the consumer, so it has to be *transmitted on-demand* to the receiver system and (typically) *played concurrently to its reception*[1]

Although the term video-on-demand is commonly used, the more general term **media-on-demand** is preferred within this thesis: As the proposed services are not limited to the transmission of video content, they are described more accurately by media-on-demand services than video-on-demand services. The term **media** in the context of media-on-demand means **streaming media**, i. e. content which

- has a *continuous* (or at least piecewise continuous) *structure* (thus the term **stream**), and
- imposes *real-time constraints for playback*.

This means that e. g. video, audio and real-time text ticker data (as well as any combinations of them, e. g. audio-visual data or video data with subtitles) are valid media types for a media-on-demand system while images, newspaper, magazines and World Wide Web sites are not[2].

These requirements also hint at the problem which has to be solved by media-on-demand systems: A media stream has to be transferred from one or several sender systems to typically several receiver systems on request in such a way that playbacks concurrently to the receptions (probably with initial delays) are possible.

1.2 Motivation for Media-on-Demand

Up to now, media streams are transmitted in most cases using traditional media broadcasting systems where media streams are broadcast at times which are chosen by the broadcasting organizations or rented from video stores. Although this provides entertainment to many people at the same time, this has to be viewed as an archaic system which does not fit the demands of the service society of the twenty-first century: Today, people are used to be served with what they want and at any time they want.

Ordering a media stream electronically using a media-on-demand system seems to be the logical successor to traditional media transmissions as this provides the best possible service to the viewer:

[1] By theory, it is also possible to use content distribution systems to provide media-on-demand services by pushing the whole media library to all receiver systems, so a playback of any media streams is possible on demand. In this setup, the differentiation between video-on-demand system and content distribution system is not possible from the type of transmission. Hybrid solutions where part of the content is transmitted using a content distribution system and the remainder using video-on-demand system approaches are also possible and reviewed later.

[2] It is possible to define additional constraints so that the latter media types are also valid media types for media-on-demand systems, e. g. by adding a read out system at the receiver side which reads the newspaper at a predefined speed or using a slide show system which presents images with specified delays, but these cases can be regarded as conversions into one of the aforementioned media types. On the other side, even media-on-demand systems can theoretically be used to transmit non-media content, ignoring the real-time capabilities for media stream transmissions of media-on-demand systems.

Flexibility: The viewer attains a high flexibility as he can view the media stream he just wants to consume and at the time he wants to consume it.

Spontaneity: The viewer gains a high level of spontaneity as consuming a media stream needs not to be planned in advance.

Independence: The viewer gets independent from the transmission schedules of the broadcasting organizations.

Topicality: As media streams can be updated by the providers at any time, the media assortment can contain the most up-to-date media streams. This includes both the selection of streams (e. g. the provider can include the newest movies in its offer) and updates to streams itself (e. g. slots in news transmissions may be updated with new facts and evolutions).

Convenience: The viewer can request the media stream from its home, even without leaving her arm chair.

Time savings: As no additional efforts are required, no time has to be spent to run to a video rental, to maintain a personal tape archive or to observe TV programs for a long awaited transmission.

The idea to provide a media-on-demand service is not new: Even in 1990, a feasibility analysis has estimated that media-on-demand becomes possible before year 2000 [Sin90]. Many companies have recognized this trend in recent years and extended their business accordingly: Software companies started to develop media-on-demand solutions (see appendix E), hardware producers began building systems which provide high data throughput for media-on-demand sender systems and designed low-cost set-top boxes (see section 6.1.3) with integrated media decoder chips for receiver systems, network providers set up simple media-on-demand services for their networks (e. g. [Deu, Arc, Han07], content providers and producers entered negotiation for media-on-demand based distribution licenses with media-on-demand providers [Goo05], market research institutions observe the service and provide recommendations for entering the market (e. g. [Poi04, loo01]) and research journals publish new approaches and advancements in media-on-demand transmission techniques (see Bibliography section for many examples).

Although good progress has been made by each of these groups, media-on-demand services have not spread widely until now: For example in the USA, an adult person consumes an average of 1 701 hours per year watching TV but only two hours per year consuming video-on-demand in 2002 [Usc05b].

This leads directly to an examination of the requirements for media-on-demand to spread:

Technical feasibility: The most important precondition is surely the technical feasibility. First of all, this means that the media-on-demand system must fulfill the theoretical requirements of

on-demand media streaming, i. e. a requested media stream must be transmitted from the media-on-demand sender system to the receiver system in such a way that it can be played without interruptions (except for unforeseen network problems) after a preset playback delay. But technical feasibility hereby not only means that the proposed technique is capable to function theoretically or work in a prototype environment, it must also be possible to use it in "real-world" environments. These environments may include setups with huge numbers of recipients, possibly inhomogeneous receiver system equipment and varying network quality.

Cost effectiveness: The major reason for companies to provide a new service is expected gain, and media-on-demand will not make an exception. Hence it is important to analyze the costs and the requirements for providers in order to examine how a media-on-demand system can be used with high economic efficiency. Another important subject to mention here is security: If the provider does not protect its service from unlicensed viewers, escaped profits would lower the cost-effectiveness and possibly encumber the media-on-demand system unnecessarily.

Availability: In any case, cost-effectiveness cannot be reached if the provider cannot acquire enough customers for its service. Therefore a media-on-demand service should provide a high connectivity which allows connecting a large number of customers. Certainly, this has also implications on the used transmission technique as it must be able to support a huge number of recipients simultaneously in an efficient and cost-effective way.

Attractiveness: But even if many customers can be reached through the network, the customers have to subscribe the service to keep the total costs per customer low and competitive to video stores or other traditional media transmission systems. Therefore expenses for the customers must be kept low and the service should be made as manifold and attractive as possible (e. g. by providing a large and up-to-date media assortment with high media quality and short playback delays).

Utilizability: Additionally, the media-on-demand server system should be capable to handle common requirements of providers: For example, it should offer efficient ways for adding media streams to the media assortment or removing others from it or for modifying parameters (e. g. playback delay, bandwidth assignments, media quality) of active streams without interrupting ongoing transmissions. For better acceptance and easier integration in existing systems, the system should also support standardized, quality-proven components (e. g. for media formats or for transmission of real-time data).

Other aspects which may influence the spread of media-on-demand services (e. g. acquaintance

of the service) are general problems of new branches of trade. As they are independent of media-on-demand, a further study of these topics is out of the scope of this thesis.

Currently used media-on-demand transmission techniques lack in one or several of these requirements: For Point-to-Point transmissions, each receiver system is served with a dedicated media stream transmission (see section 3.3.1). This allows to provide a very good playback delay to the customers but as the media-on-demand server throughput and the server bandwidth requirements are proportional to the number of concurrently active transmissions, this raises the costs of the service considerably for huge systems and limits the use on networks which cannot be extended when the number of recipients increase (e. g. cable TV networks or satellite networks).

In contrast, the Staggered Broadcasting scheme transmits the media stream on several channels in parallel with a different phase shift for each channel (see section 3.4.2). Although the bandwidth requirements for this transmission scheme are independent from the number of active receivers, the needed number of channels for acceptable service latency is too high to make the service profitable.

For the Batching transmission scheme (see section 3.3.2), requests for the same media stream are aggregated and served with a single transmission. This lowers the number of concurrently active transmissions but increases the playback delay at the same time. In the marginal case where requests are performed at a high frequency, Batching even converges to the Staggered Broadcasting scheme with the same problems as described above.

Conversely, when using Complete Preloading (see section 3.2.4), the media streams are transmitted in advance to the receiver systems and stored for playback at a later time. Thus Complete Preloading benefits from an increasing number of recipients (assuming a broadcast network), has very low bandwidth requirements and provides immediate playback, i. e. the best possible playback delay. Unfortunately, it is unemployable in many practical setups because the costs for receiver systems are far too expensive: To provide a real media-on-demand service this way, the receiver system would have to store all media streams on its storage device which requires a giant hard disk at each receiver system for media assortments of acceptable size.

Limiting the drawbacks of the different approaches of the transmission schemes and finding a suitable compromise is one of the main goals of this thesis.

1.3 Scope of this Thesis

In order to facilitate the spread of media-on-demand, this thesis proposes methods for *improving the efficiency* and *extending the utilizability* of *media-on-demand systems*. In detail, this comprehends the following topics:

Analysis of the structure of media-on-demand systems and extensions thereof:

To get an overview of the functioning and the comprising *components of a media-on-demand system*, the structure has to be analyzed. This allows to identify the functional parts of the system and to apply some general extensions.

Derivation of a measurement for efficiency of media-on-demand transmission schemes from theoretical bandwidth requirement analysis:

To compare the utilizability of media-on-demand transmission systems, it is not sufficient to enumerate only functional features and drawbacks of the schemes: As the resource requirements influence the acceptance and deployment of media-on-demand services, it is very important to compare these requirements, too. One of the most critical resources in many media-on-demand systems is the bandwidth of media-on-demand transmissions. To get an unbiased bandwidth comparison of media-on-demand transmission schemes, the theoretical bandwidth requirements for a transmission have to be analyzed and a measurement for the *efficiency* of the transmission scheme has to be defined which takes all parameters of the transmission into account.

Development of algorithms for measurement of memory requirements of media-on-demand transmission schemes:

Besides of the bandwidth requirements, other characteristics determine the *cost-effectiveness* and *utilizability* of the system. One of such parameters is the size of storage which is needed at the receiver system. Unfortunately, this is a transmission scheme immanent parameter which cannot be retrieved in a simple way for all schemes, so a method to determine it has to be developed.

Evaluation of a wide range of media-on-demand transmission schemes for advancements, drawbacks and efficiency:

Many approaches have been proposed in recent years for media-on-demand. Although not all of these transmission schemes are in practical use, it is beneficial to evaluate the existing schemes to get an overview of *enhancements and drawbacks* and to evaluate their efficiency.

Proposal of a new media-on-demand transmission scheme which combines most of the features of the existing transmission schemes while providing an increased efficiency and enhanced utilizability:

One main topic of this thesis is the proposal of a *new transmission scheme*. This transmission scheme, the **Generalized Greedy Broadcasting Scheme**, is highly generic which allows incorporating most of the features and extensions of other schemes into this single scheme. At the same time, this transmission scheme provides a very high efficiency (near to the theoretical optimum), making this scheme a “one-fits-all” solution for media-on-demand.

Adaption of standardized real-time transport protocols to be applicable to media-on-demand:

For the transmission of real-time media data, the Real-Time Transport Protocol [SCFJ03] has evolved to be a suitable and efficient solution. Although the new generation of media-on-demand transmission schemes have slightly different requirements (e. g. many of them have to split the media streams into small parts and transmit the parts repeatedly in a special order), they still need to transmit data in a real-time like manner. Providing a *new payload format for the Real-Time Transport Protocol* which extends the Real-Time Transport Protocol to the requirements of the new generation of media-on-demand transmission schemes allows combining the benefits of both worlds.

Cost comparison of the proposed media-on-demand system in different application cases:

To compare the proposed media-on-demand system with the simple, most commonly used Point-to-Point media-on-demand solutions, a *comparison of costs* is important.

Summary of Scope

Summarizing the above, the subject of this thesis can be defined as follows:

> Analysis, improvement and enhancement of media-on-demand systems and transmission schemes with respect to efficiency, utilizability and applicability in different use-cases.

1.4 Structure of this Thesis

This thesis consists generally of four parts: In the first part (chapter 2), the structure and capabilities of media-on-demand systems are analyzed. This involves a dissection of media-on-demand systems into components and a description of their cooperation, a classification of media-on-demand system types and a short overview of media content and formats.

The most significant task of a media-on-demand system is the transmission of media streams. As there are a lot of approaches for a more or less efficient transmission of media streams, the second part (chapter 3) takes existing media transmission schemes into focus. This involves an outline of the functioning of nearly fifty media transmission schemes, a description of their benefits and drawbacks and a comparison of their efficiency.

In the third part (chapters 4 and 5), a new media-on-demand transmission scheme is proposed which combines a high efficiency with most of the features and benefits of the existing transmission schemes. This task is performed by defining a generalization of an existing transmission scheme and by adopting the resulting scheme to different application cases which influence the bandwidth requirements or the utilizability of the transmission scheme (chapter 4). Another issue

which is covered in this part is the enhancement of the media transport by transmission scheme independent means, e. g. the introduction of security and an embedding into a standardized real-time transport protocol (chapter 5).

The last part (chapter 6) addresses financial issues of media-on-demand systems: Although media-on-demand is technically feasible for years, only few media-on-demand services have evolved, mainly based on simple and inefficient transmission schemes. Thus to verify the practical utilizability of the proposed media-on-demand system, this part performs a cost analysis comparison of a simple and an enhanced media-on-demand system.

Following these four parts, the last chapter (chapter 7) concludes this thesis with a summary of all obtained research and engineering results.

Chapter 2

Media-on-Demand Systems

THE introduction provided an overview of the objectives of media-on-demand systems as well as a reasoning of the needs for media-on-demand transmissions. To get a deeper look inside media-on-demand systems, this chapter covers the basic setup and functioning of media-on-demand systems.

As a starting point the next section provides an introduction into the general structure of media-on-demand systems.

In section 2.2 and 2.3, the functioning of media-on-demand systems is analyzed in further detail. Therefore, in section 2.2 a media-on-demand system is dissected into components and their cooperation is described and in section 2.3 the topology of media-on-demand systems is examined.

After the analysis of media-on-demand systems, classifications for media-on-demand systems are proposed. Classifications are especially useful for a justified comparison of the existing transmission schemes which will be presented in the chapter 3, and to identify the advantages of the newly proposed transmission scheme in chapter 4, and also for a clarification of the capabilities of media-on-demand systems in general and to delimit media-on-demand from other interactive media services. Therefore, section 2.4 provides a detailed list of technical and functional classifications.

Section 2.5 goes into some details of the most valuable element of a media-on-demand system: the media content. This includes the structure and encoding as well as some properties of it, e. g. bit rate variations and fault tolerance.

In the last section of this chapter, a summary of the results is presented.

2.1 Functioning and Capabilities of Media-on-Demand Systems

This section provides an introduction into the structure and capabilities of media-on-demand systems. For this, both a simple and a more complex media-on-demand system are proposed and compared:

- The presumably most simple media-on-demand system consists of a **media-on-demand sender system** and one or several **media-on-demand receiver systems** where each of the receiver systems is connected to the sender system via a bidirectional network connection.

  When a **consumer** requests a **media stream** from the receiver system, the receiver system sends a **playback request** to the sender system. As soon as resources are available, the sender system starts to transmit the media stream to the receiver system at media stream bit rate. When the media stream is received by the receiver system, the **playback** can commence by displaying the stream to the consumer (typically with minor delay for **jitter compensation**).

  The consumer may also control the playback, e. g. by using pause, rewind, fast forward or jumping to specific positions. These **playback control requests** are sent to the sender system where they are executed.

  Some media streams even support non-continuous content navigation (interactive TV), e. g. selection of alternative endings or different view angles. In this case, the **content navigation requests** are sent to the sender system.

- A more complex approach is selected in the following media-on-demand system: The sender system prepares the media stream in such a way that it can be transmitted permanently on a set of broadcast **channels**. When a consumer wants to join the transmission, the receiver system has to tune into the appropriate channels and starts the playback (usually after a short **playback delay**).

  To assure a **continuous playback**, the media-on-demand system has to adhere to a defined **transmission schedule**. For example, this schedule may define that the media stream is to be cut into small pieces — called **segments** throughout this thesis — and how these segments have to be arranged across the channels for transmission. By adhering to the schedule, it is assured that the receiver system gets each segment of the media stream in due time for playback.

  In this latter example, the communication between sender and receiver systems is only unidirectional: The sender system sends data to the receiver system but not vice versa. Consequently, the media-on-demand sender system has to work without processing playback

control (and content navigation) requests, these request have to be handled locally at the receiver system as far as possible[1].

These two examples should have given a small insight into media-on-demand systems and into the manifold of approaches which can be used by media-on-demand systems, although many topics — e. g. encryption, error correction and the exact functioning of transmission schedules — have been omitted in these examples to keep them simple.

To organize the features and functional units of a media-on-demand system, the following three sections provide a detailed look at the components of media-on-demand systems, their topological arrangement and a classification of the capabilities.

2.2 Structure of Media-on-Demand Systems

In the previous section, a simple introduction into the basic functioning of media-on-demand systems has been provided. Before looking deeper at the media-on-demand transmission schemes, this section describes the parts a media-on-demand system consists of.

As a first, very rough partitioning, a media-on-demand system can be divided into five parts in accordance to the Digital Audio Visual Council[2] (DAVIC) reference model [Dav99, MS02]:

- A **content provider system** (CPS) which offers and sells media streams to service providers,
- a **service provider system** (SPS) which stores media streams from the content provider system in its media database and which accepts and processes schedules (either fixed ones or based on requests) for transmission of media streams to service consumer systems,
- a **service consumer system** (SCS) which allows a consumer to receive a media stream from a service provider system and which is implemented in a customer premise equipment (CPE),
- a **CPS-SPS network** for the transmission of media streams from the CPS to the SPS and
- a **SPS-SCS network** for the transmission of media streams from the SPS to the SCS and of other data between the SPS and the SCS.

As the main focus of this thesis lies in the efficient transmission of media streams from the service provider system to the service consumer system, a more simplified division is used:

[1] As will be explained later, pro-active systems normally cannot provide the same level of interactivity as the aforementioned reactive system, therefore only a subset of playback control and content navigation requests can be supported.

[2] The Digital Audio Visual Council is a non-profit organization which has been created to provide end-to-end interoperability of broadcast and interactive digital audio-visual information and of multimedia communication. Unfortunately, it has been closed in 1999 and only focused on Point-to-Point (see section 3.3.1) and Staggered Broadcasting (see section 3.4.2) transmissions which were the most popular ones at this time.

- The content provider system, service provider system and the network between these two systems are combined to a **media-on-demand sender system**, which is responsible for providing the media stream data,
- analogously the service consumer system is called **media-on-demand receiver system** in this thesis and is liable for receiving one (in some cases several) media streams and displaying them for playback and
- the SPS-SCS network is split into a **media data transport system**, some kind of real-time transport connection from the sender to the receiver system, and
- optionally an **auxiliary data transport system**, a transport system for transmission of any non-media data between sender and receiver systems. The auxiliary data transport system can be used for transmission of media playback requests, playback control requests, content navigation requests, error recovery information, security means and/or accounting and payment information, but it may also be unavailable.

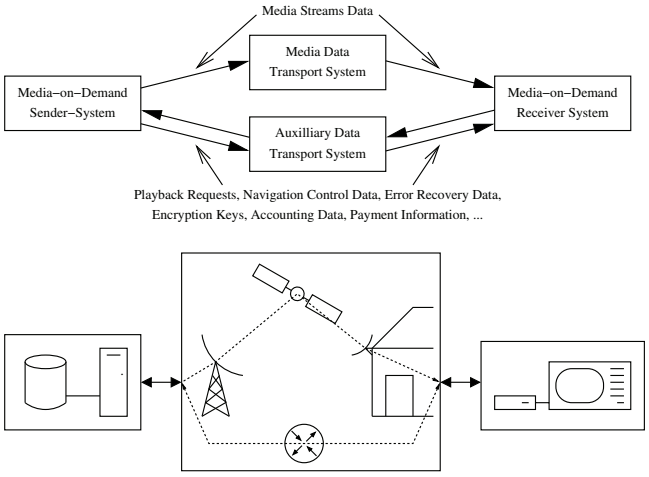


Figure 2.1: General structure of a media-on-demand system

The two transport systems differ much in their properties: While the media transport system is a high-bandwidth, real-time aware, unidirectional connection which has to provide a sufficient quality of service for media transmission, the auxiliary data transport system is typically only a low-bandwidth, but sometimes bidirectional and reliable connection. These requirements are not specifically for media-on-demand systems, thus their functioning is not described in more detail in this thesis.

Both the media-on-demand sender and receiver system parts can be further decomposed into several small components which are presented in the next two subsections. Not every media-on-demand system always contains all described components of the following subsections: Some

components may be missing as they are not required by the used transmission scheme (e. g. the segment storage at the receiver will not be necessary if the transmission scheme does not require to store any segments), other components may be left out on purpose (e. g. some components for encryption and error correction may be missing if these functionalities are not desired). Furthermore, the decisions where to put the cuts into identifiable components are artificial and may not always represent the internal structure of existing systems, in fact the parts which are discussed in this thesis in more detail are dissected into more and smaller pieces than other, surrounding parts. Nevertheless, this decomposition is proposed for a better understanding of the overall functioning of media-on-demand systems and for the determination of related terms which are used throughout this thesis.

2.2.1 Sender-Side Components of a Media-on-Demand System

This subsection describes several aspects of a media-on-demand sender system. As a side effect, many terms are defined and reasons for splitting the sender system at the chosen boundaries are provided at the end of this section.

Media Source

The first thing we have to observe on the way of a media stream from the sender system to the receiver system is the source of the media: Typically, each new media stream will be entered manually through a **media database administration** component into the **media database** of a media-on-demand system, e. g. it may be supplied using a data medium like a DVD or by downloading it from the content provider. The media streams will be saved in the media database, most probably on large hard disc arrays, but for seldom requested media streams larger (and slower) storage systems may also be used (e. g. disk arrays, juke boxes or tape robots)[3].

This feeding of the media system can also be automated, e. g. the system may look regularly for new media streams at some predefined places somewhere in the Internet or it may integrate new media streams automatically when they are sent to it by the content provider or an administrator from a remote place. To prevent unwanted manipulations, any access to the media database should be protected.

For **live transmissions** a complete other type of media source is used: Hereby the media stream has to be **captured and encoded in real-time** and saved into the media database, from where it can be transmitted to the recipient system almost without further delay. **Near-live transmissions** are possible the same way, and in section 4.17 the impacts of live transmissions and near-live transmissions on the transmission schemes and their efficiency is examined.

[3] The authors of [Wol04] show a table for different media storage types, pointing out the oppositeness of short access times and low costs.

If a media-on-demand system is an intermediate media-on-demand system (see section 2.3), media streams are received from other media-on-demand sender systems through an attached media-on-demand receiver system instead of being entered through the media administration or through live capturing.

Transmission Scheduling

Besides of the media source management, several further components are needed at the sender system. Firstly, for managing the available bandwidth and distribute it to the scheduled transmissions, a **transmission scheduler** is needed. The transmission schedule decides which media streams are transmitted at a time, depending on either a predefined schedule (the only solution if no back-channel is available) which can be managed through the **schedule administration** component, a dynamically created schedule (which considers customer requests for playback) or something in between (e. g. the provider may decide to transmit some popular movies without requests and leave the remaining bandwidth for individual requests). An **electronic program guide generation** component may be used to transmit the media stream assortment to the media-on-demand receiver systems.

Transmission scheduling biased on user requests is very difficult as soon as the resource limit is reached: The system has to find a balance between **fairness** (which means that the same maximum playback delay is granted to every request) and **efficiency** (which increases most when the media streams with the highest number of requests are transmitted first). Thereby none of these extremes provides the optimal solution (measured in number of satisfied requests per time unit): Serving all requests in a fair way keeps waiting times for seldom requested media streams low, but the efficiency is low as a high amount of bandwidth is consumed by only few consumers. On the other side, serving highly requested media streams first increases the efficiency, but media streams with low request rates have long service times, causing the customers to renege the service. Different batching schemes which focus on this problem are examined in section 3.3.2.

Media Stream Transmission

The next attention attracting component at the sender system is the media stream transmission: As already introduced in section 2.1, many of the media-on-demand transmission schemes split media streams into segments for transmission. To assure that all connected media-on-demand receiver systems will have received the segments in due time for playback, the segments have to be transmitted on a set of channels at specific times as defined in the **transmission schedule**. Determining the (sometimes very complex) transmission schedule for a media-on-demand transmission is the main task of the **segment scheduler**. In chapter 3, schedules of different types of transmis-

sion schemes are examined and compared and in chapter 4, an improved transmission scheme is proposed.

Some transmission schemes use segments of equal size and channels of equal bandwidth (or fractions of a base bandwidth). In this case, the duration it takes to transmit a segment on a (base) channel is called **slot interval** and the available capacity of each channel is divided into chunks (called **slots**), each capable to transmit a segment. The segment scheduler is then responsible for assigning segments of the media stream to slots of the channels.

The determined segment schedules are needed by the **segment sender**, which reads segments from the media database and transmits them to the receiver systems according to the arranged schedule.

Error Correction

Optionally, the provider may decide to improve the reception quality of the transmission by supporting **forward error correction**, **ARQ-based error recovery** or a combination thereof. Section 5.1 examines the different approaches for error correction and their impacts (e. g. to the playback delay) in more detail.

Security

If a back-channel from the receiver system to the sender system exists, media stream requests can be accepted by the sender system. For correct billing and to prevent requests from unlicensed customers, a **customer authentication** component can be used to verify incoming requests.

Another option is **encryption**, **signing** or **watermarking** of the media stream or using other means of security to protect it from unlicensed viewers, alterations or to prove the origin of a copy, resp. If the encryption and signing keys are not static, a **key distribution** system will also be needed for transmitting the active set of keys to all licensed viewers. Security related issues for media streams are examined more precisely in section 5.3.

Accounting & Customer Administration

To limit access to the system, a **customer database** can be used which stores authentication data of all licensed customers. To maintain this database, a **customer database administration** component may be used.

To support pay-per-view semantics, a **customer accounting** component will be required, which registers (authorized) incoming requests for media streams.

Sender System Components Overview

Figure 2.2 shows a possible dissection of a sender system in components and their communication. Table 2.1 summarizes all components of this figure in tabular form together with their function.

Component	Function
Media Source:	
Media Database Administration	Provides an interface for adding and removing media streams into/from the system.
Media Database	Stores media streams, delivers media metadata, reads media streams segment-wise.
Real-time Capturer and Encoder	Captures and encodes live media streams, saves them into the media database.
Media-on-Demand Receiver System	For intermediate systems in a hierarchically organized media-on-demand system, this is the interface to the receiver system part.
Transmission Scheduling:	
Transmission Scheduler	Manages available bandwidth for scheduled and requested transmissions, determines which media stream is transmitted at what time on which set of channels.
Schedule Administration	Provides an interface for defining schedules manually.
Electronic Program Guide Generation	Creates a list of all available media streams to offer them at the media-on-demand receiver system to the customer.
Media Stream Transmission:	
Segment Scheduler	Calculates transmission schedules for scheduled media transmissions, determines which segments are transmitted at what time on which channels.
Segment Sender	Transmits segments according to transmission schedules from the segment scheduler.
Error Correction:	
Forward Error Correction Generator	Calculates redundancy data for transmissions.
ARQ-based Error Recovery	Handles retransmission requests from receiver systems.
Security:	
Customer Authentication	Verifies authenticity of incoming playback requests from customers.
Watermarking, Signing & Encryption	Allows to prove the origin of a media stream, prevents alterations and provides protection from unlicensed viewers.
Key Distribution	Distributes encryption and signing keys to licensed viewers.
Accounting & Customer Administration:	
Customer Database	Provides data for authentication of customers, keeps accounting data for customers.
Customer Database Administration	Interface for maintaining the customer database.
Customer Accounting	Accounts playback requests to customers.

Table 2.1: Functions of the presented dissection of a media-on-demand sender system

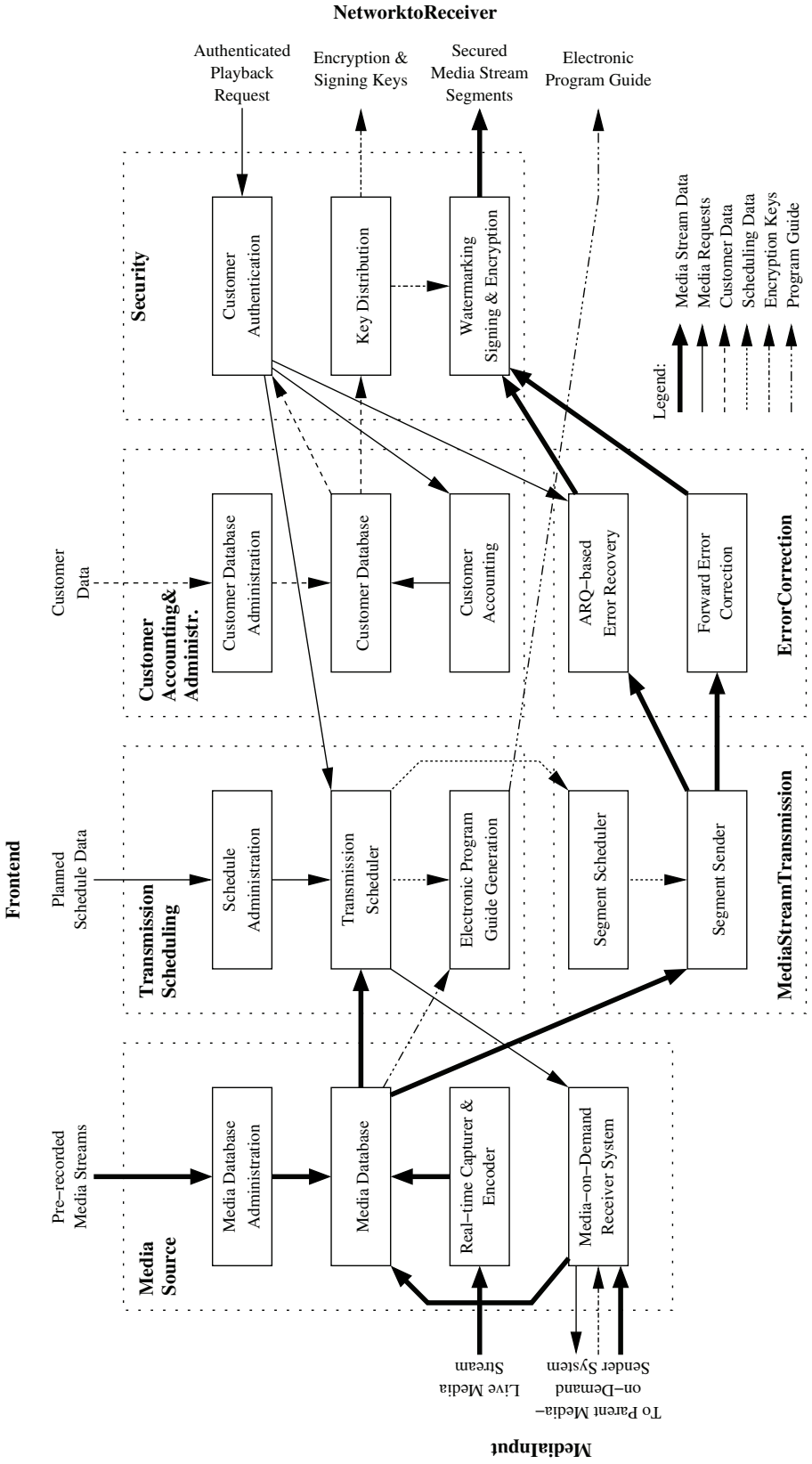


Figure 2.2: A dissection of a media-on-demand sender system

2.2.2 Receiver-Side Components of a Media-on-Demand System

On the receiver side, the system can be dissected in a very similar way into components:

Reception Scheduling

To start a media-on-demand reception, the customer has to select the media stream he wants to see through the **media stream selection** component, probably using data from the **electronic program guide storage** or information passed outside the scope of the media-on-demand system (e. g. a printed program guide, advertisements, a permanent program guide channel, ...). Additionally, the **reception scheduler** has to check if enough resources (e. g. bandwidth, storage) are available for the reception of the selected media stream and reserve the required resources for the duration of the reception.

Media Reception

When a media stream has been selected for reception, the **playback scheduler** has to evaluate the transmission parameters: Each media-on-demand transmission may use a set of channels which must be joined and left at specific times (esp. if the reception bandwidth of the media-on-demand receiver system is limited). Additionally, several further channels may be used for optional services (e. g. to increase the media stream quality or to add error correction redundancies). The playback scheduler selects the channels to join and instructs the **segment receiver** to receive segments from the channels.

Media Destination

The segment receiver then tunes into the selected channels and forwards the received segments to the **segment storage**. Depending on the type of interactivity and the used transmission scheme, this segment storage may be very small (only a buffer for compensating jitter of the media data transport system) up to very huge (e. g. to provide some support for playback control requests to transmission schemes which do not support any type of interactivity by themselves).

From the segment storage, the segments are read by the **media player** and presented to the consumer. For interactive media streams, the consumer can send playback control or content navigation requests to the aforementioned reception scheduler, so it can update its schedule and instruct the playback scheduler about the new schedule.

If the media-on-demand receiver system is an intermediate system (see section 2.3), the receiver system obtains its requests from the attached media-on-demand sender system and forwards all received data to this system.

Error Correction

To compensate for lost (or corrupted) data on the media data transport system, the sender system may have added **forward error correction redundancy**, support **ARQ-based error recovery** or a combination thereof. The receiver system may benefit from forward error correction if it can reconstruct lost or corrupted data with the help of the redundancy data or by requesting retransmissions.

Security

Analogical to the security components of the sender system, corresponding components at the receiver side are required: If a back-channel from the receiver system to the sender system is available, the media-on-demand sender system may require a playback request for the transmission of the media stream. To authenticate the customer to the media-on-demand sender system, a **customer authentication** component may be required. Accordingly, a **decryption & signature checking** component and a **key reception** component are needed to decrypt the media stream or to verify the stream integrity and to obtain the used encryption and signing keys if non-static keys are used.

Customer Identification

To identify a customer in the media-on-demand receiver system, a **customer login** component may be used.

Receiver System Components Overview

Figure 2.3 shows a dissection of a receiver system into the aforementioned components and their communication. Table 2.2 summarizes all components of this figure in tabular form together with their function.

2.3 Topology of Media-on-Demand Systems

Simple media-on-demand systems depict a **star topology**, where the center node is the sender system and the other nodes are receiver systems (see figure 2.4a). But for large media-on-demand installations it may be advisable to use a more complex topology where sender systems are replicated for reliability, load balancing or for locality reasons.

Higher reliability can be reached if several sender systems are located within the reception area of each receiver system. In case of a failure, other systems must be able to take over existing transmissions of the failed system or at least serve new media requests. Similarly, to lower load

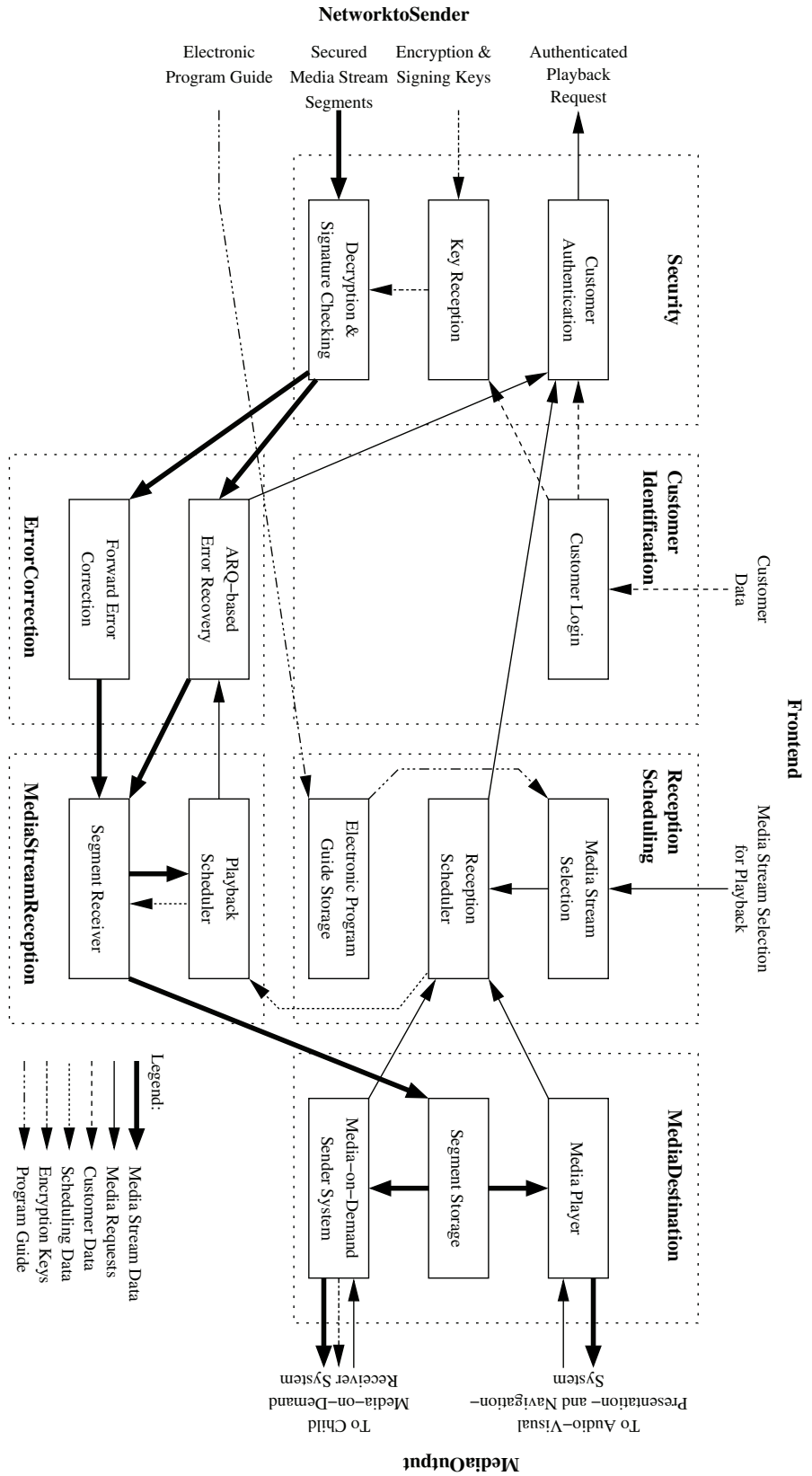


Figure 2.3: A dissection of a media-on-demand receiver system

Component	Function
Reception Scheduling:	
Media Stream Selection	Handles selection of a media stream to request.
Electronic Program Guide Storage	Keeps a list of the currently available media streams for playback.
Reception Scheduler	Manages available bandwidth for scheduled and requested receptions, determines which media stream is received at what time.
Media Reception:	
Playback Scheduler	Determines which segments have to be received until what time on which channels.
Segment Receiver	Receives segments and stores them into the segment storage.
Media Destination:	
Segment Storage	Stores media stream data segment-wise.
Media Player	Presents the media stream to the consumer, forwards navigation requests to the playback scheduler.
Media-on-Demand Sender System	For intermediate systems in a hierarchically organized media-on-demand system, this is the interface to the sender system part.
Error Correction:	
Forward Error Correction Recovery	Evaluates redundancy data for recovery of lost or corrupted segments.
ARQ-based Error Recovery	Requests retransmission for lost or corrupted segments from the sender system.
Security:	
Customer Authentication	Authenticates outgoing playback requests.
Decryption & Signature Checking	Decrypts encrypted transmissions and checks digital signatures of signed transmissions.
Key Reception	Receives encryption and signing keys.
Customer Identification:	
Customer Login	Accepts identification data of the customer.

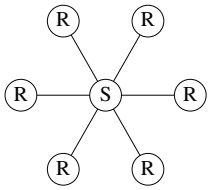
Table 2.2: Functions of the presented dissection of a media-on-demand receiver system

and bandwidth usage of sender systems or of the network, requests of receiver systems can be distributed to different sender systems, e. g. taking the current load or the distance between the sender and receiver system into account or using a random or preset assignment.

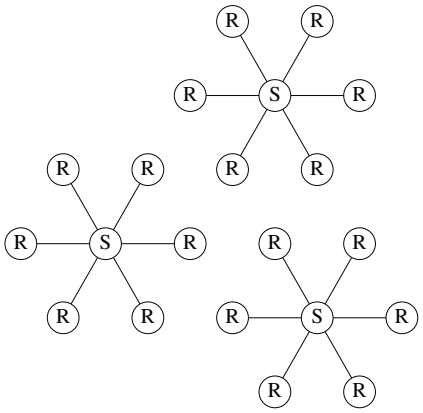
Reducing the network distance between sender and receiver systems can have several reasons which mostly depend on the type of the used network. One possible reason is to reduce the load of the network: The shorter the network distance between sender and receiver, the less routers or switches and the less data link connections are affected by the transmission. Sometimes locality is even required due to missing routing capabilities of the underlying network (e. g. if multicasting capabilities are required for the media stream transmission).

Besides of simply using several independent sender systems (see figure 2.4b), the sender systems can also be connected to each other. If a strict hierarchy is used for all sender systems (i. e.

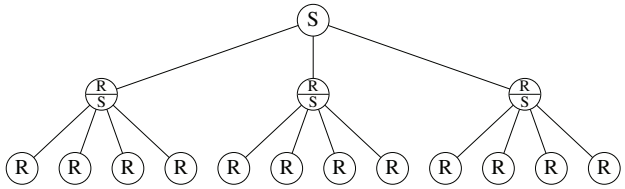
one designated sender system serves as master to all other sender systems, either directly or indirectly), this leads to a **tree topology** (see figure 2.4c): In this tree the **master sender system** is located at the root of the tree, the remaining non-leaf nodes work as **intermediate sender systems** and the receiver systems are positioned at the leaf nodes.



(a) Media-on-demand system using a star topology



(b) Several independent star topologies which serve receiver systems locally



(c) Hierarchically organized media-on-demand system using a tree-topology

Figure 2.4: Three example topologies for media-on-demand systems

Using a tree topology for organizing huge media-on-demand systems has the advantage that maintenance is much easier: Changes to the media assortment must not be performed on every media-on-demand sender system but only on the master sender system from where the changes are distributed to the intermediate sender systems. When a request for a media stream is received by an intermediate system, it can request the media stream on its **parent media-on-demand system** and send the received media data to its **child systems**. Besides of only forwarding media streams, intermediate systems can also cache the received media streams into their media database for answering subsequent requests for the same media stream without involving their parent media-on-demand system again.

But instead of using some file transfer software for transferring media streams between sender systems, it is possible to use media-on-demand transmissions at every level in the hierarchy. This gives two advantages: Firstly, by using media-on-demand transmissions even between sender systems, duplicate requests from several intermediate sender systems to a common parent system can be combined into a single transmission in the same way as duplicate requests from several

receiver systems to a common sender system, saving bandwidth for the transmissions between sender systems. Secondly, when a media stream is transmitted linearly from the parent media-on-demand system to an intermediate media-on-demand system at the time of the request from the media-on-demand receiver system, the intermediate media-on-demand system has no immediate access to the full media stream but just to the currently received part. This is comparable to a live (or near-live) transmission for the intermediate media-on-demand system which results in higher bandwidth requirements for some transmission schemes. Using the same media-on-demand transmission scheme at every level of the topology allows the intermediate systems to just forward the received data; no access to missing parts of the media stream is required at any time.

From another point of view, a hierarchical media-on-demand system can be regarded as an assembly of several simple media-on-demand systems, using a star topology at each subtree in the complete hierarchy tree: While the master sender at the root of the hierarchy serves as media-on-demand sender system only, intermediate systems are constituted out of a media-on-demand receiver system (for receiving media streams from their parent system) and a sender system (for sending media streams to their child systems). The receiver systems at the leaf nodes remain unchanged; the full topology is hidden at this level.

Several other topologies are possible, too, e. g. setups where all sender systems are connected in a **ring** or constitute a complete or incomplete (perhaps even irregular) **mesh** and receiver systems may be connected to one or several sender systems in rings or stars, but these setups loose most of the above mentioned advantages of the tree topology.

2.4 Classification of Media Transmission Systems

Classification of media transmission systems is a difficult matter as there are a lot of characteristics which can be used for a classification. Most commonly, media transmission systems are divided into groups depending on the supported level of user interactivity, but from a provider's point of view many other criteria may be more important, e. g. the required communication type of the network or the ability to transmit live streams.

In the following subsections, several classification schemes are introduced which help to benchmark and compare the transmission schemes which are proposed in the next chapter. For a better overview, the given classifications are presented in two groups:

- Technical classification schemes:
  This group is the larger one and includes classifications by communication type, network type, transmission scheme and many less important ones.
- Functional classification schemes:

This group contains classifications which concern the range of supported functions of the transmission system, examining both capabilities for providers and for consumers.

2.4.1 Technical Classification of Media Transmission Systems

The technical classification of media-on-demand systems is a very important issue as this classification allows a provider to perform a preselection of transmission schemes which fit to her prevailing conditions. For example, if a provider wants to use a wide-area broadcasting network for media-on-demand (e. g. using satellites), he cannot use transmission schemes which serve only a single customer for each transmission (or at least not cost-effective). Thus technical classifications characterize if a transmission scheme can (theoretically and practically) be used when the technical infrastructure is preset and immotile. Otherwise, if the infrastructure has not been specified, technical classifications can help to define the requirements.

Communication Type

One important characteristic for media-on-demand systems is the communication type: Traditional media-on-demand systems are based on **one-to-one** transmissions for the delivery of media streams. As a consequence, the bandwidth and throughput requirements for media-on-demand sender systems are proportional to the number of concurrently active media-on-demand receiver systems.

One-to-many transmissions provide a way to lower these requirements: The media-on-demand system sends the data once on a broadcast or multicast channel and it is up to the network protocol to deliver the data to all active receiver systems efficiently. This way, the sender system can transmit data to an arbitrary number of receiver systems without increasing the bandwidth requirements of the sender system proportional to the number of transmissions.

It is also possible to use a **combination of one-to-one and one-to-many** transmissions, e. g. sending often requested media streams using one-to-many transmissions and seldom requested streams using one-to-one transmissions. Some transmission schemes even combine one-to-one and one-to-many transmissions for a single transmission, e. g. a one-to-many transmission is used for the main part and one-to-one transmissions are used for missed parts of latecomers.

Nearly any communication network can be used for media transmissions (provided that the network offers a suitable bandwidth, short latency and a sufficient quality-of-service), so this section can only give a short overview of the most commonly used network types for media-on-demand.

Broadcasting Networks: Broadcasting networks have a long history for non-interactive media transmissions (i. e. traditional radio and TV) as they can be used to reach a huge number of recipients at once. Plain analog over-the-air (i. e. radio wave based) transmissions have

been used anciently, followed by analog broadcasts over cable TV (CATV) networks, and further followed by digital broadcasts over satellite, air, cable and cellular phone networks.

For media-on-demand, digital networks are required by nearly any of the transmission schemes, so the digital successors (e. g. based on DVB-T, DVB-C and DVB-S) are candidates for media-on-demand. But also cellular phone networks (e. g. using GPRS, UMTS or DVB-H) or short-range networks like local area networks of different type (e. g. Ethernet or wireless LAN) can be used for media-on-demand transmissions.

All of the aforementioned networks can be used for one-to-many transmissions in media-on-demand systems. They mainly differ in the number of recipients they can reach (satellite-based networks can typically reach more recipients than terrestrial networks, cable TV networks, cellular phone networks or even local area networks).

The more recipients a sender system can reach, the higher is the availability of the provided service. But at the same time, the more recipients are reached, the more recipients have to share the same bandwidth. Thus, reaching a high number of recipients through broadcasting is only advantageous if all recipients request the same small set of media streams. If many different media streams are requested, several small broadcasting networks are more beneficial.

Multicast-Capable Networks: Many networks provide a way which can be used to transmit a media stream from one sender to several recipients without sending the data several times over a link. The most common multicast-capable communication protocols comprise multicast-capable IP networks (e. g. the M-BONE, a satellite link using MPE, ULE or GSE encapsulation to transmit IP packets or local area networks) and ATM networks. Additionally, it is possible to setup tunnels through unicast networks to bypass non-multicast links. If the Internet provider where the home systems are connected to (e. g. by DSL) supports multicasting, it is even possible to provide multicast service up to the home[4].

Similar to broadcasting networks, multicast-capable networks can be used for one-to-many transmissions in media-on-demand systems.

Unicast Networks: Unfortunately, the major part of the current Internet does not provide multicast services yet. But all of the aforementioned multicast-capable networks also provide unicast functionality, and even the broadcast networks can be used for unicast transmissions, so all mentioned protocols can be used for one-to-one transmissions in media-on-demand systems. But as the available bandwidth is typically the more expensive the more recipients can be reached with it, large broadcasting networks are not cost-effective if they are used

[4]Unfortunately, many Internet providers do not provide multicast services to their customers yet, often not because of technical problems but just because of missing means for accounting multicast traffic at the sender side.

to transmit large amounts of data to single recipients. (Chapter 6 presents a comparison of the costs for unicast- and multicast-based media-on-demand systems in different network setups.)

Back-Channel Support

For some media-on-demand transmission systems it is necessary that requests for media streams are transferred from the media-on-demand receiver system to the sender system. These systems are called **reactive**, **online**, **close-loop** or **user-centered**. The counterpart of these systems, the **pro-active**, **offline**, **open-loop** or **data-centered** media-on-demand systems, perform permanent broadcasts, independent of any signaling of receiver systems.

Both types of systems have their own advantages: A reactive system can provide immediate service and can omit unnecessary transmissions because it has knowledge about active recipients of ongoing transmissions. Additionally, a reactive system can provide a much higher type of interactivity (up to virtual reality, see section 2.4.2). Accounting (e. g. pay-per-view), encryption (e. g. using keys which are only announced to active recipients) and error correction (e. g. using a NACK-based mechanism) benefit from a back-channel, too.

But pro-active systems have the advantage that they provide a better scalability in huge systems: As a pro-active system does not get any feedback from the receiver systems, their operation is not influenced by the number of active receiver systems. Esp., pro-active systems always provide the same quality of service: The playback delay will always be determined by the provider when scheduling the media stream transmission, it will never be the case that the playback delay increases because the media transport system is overloaded from too many active recipients. Last but not least, pro-active systems do not require a back-channel, so they can even be used if the transport system only provides a unidirectional service.

To benefit from several of the advantages of both types, reactive-pro-active hybrid systems combine mechanisms from both sides, so they can provide immediate service, a better scalability and a bandwidth reduction at the same time.

Topology of Media Transmission Systems

The media content in a media transmission system can travel on different ways, but in total it is transferred from a single or limited number of sender systems to a huge number of receiver systems. As stated in section 2.3, several topologies are possible, e. g. stars, trees or any type of meshes.

The advantage of using complex topologies instead of a simple star topology is to limit the number of playbacks of each sender system so the load and bandwidth use can be lowered. Another advantage is increased robustness to failures of sender systems if other sender systems can take

over the tasks of failed systems. The disadvantage of using multiple, possibly distributed sender systems are higher installation and maintenance costs.

Besides of utilization and failover reasons, the use of complex topologies depends as well on the type of the network: For example, if the media stream is transmitted via satellite with all receiver systems in the satellite footprint, complex topologies do not make any sense as all receiver systems share the same data link between sender and receiver systems, but for large media-on-demand systems using point-to-point connections or routing multicast connections, hierarchies or meshes are very beneficial.

In some cases, even both ways are possible: For example for cable TV networks, the bandwidth of the network will be most probably the limiting factor which determines the number of concurrently active media streams, so if only a few, popular media streams should be transmitted, no hierarchies are needed. At the other extreme, if the media assortment is very large and does not contain many popular media streams, placing one intermediate sender system at each cable TV network segment feed allows to transmit the same number of concurrently active media streams at each segment as before in the whole setup.

Another problem of retrieving a media stream through a chain of sender systems (e. g. along the path from the master sender system through intermediate sender systems to the receiver system in case of a tree topology) occurs at the intermediate sender systems: When media streams are transmitted to intermediate sender systems as a result of a request from a receiver system, the intermediate sender system has no access to the whole media stream during the transmission but only to the yet received parts. This is a situation similar to the live streaming problem (see section 4.17) where the sender system cannot access future parts of the media stream for transmission. Fortunately, one advantage over live streaming exists: If the intermediate sender system uses the same transmission scheme as its parent system, the received data can simply be forwarded without caring for live streaming issues because the parent sender system already guarantees that all data will be available in due time.

Bandwidth Requirements

The bandwidth requirements at the sender side and at the receiver side depend on many parameters. For the media-on-demand sender system, the following parameters may influence the bandwidth requirements:

- The number of transmitted media streams,
- the number and join times of receiver systems receiving a media stream,
- the number and bandwidth of the channels used for the transmissions (which may in turn depend on the transmission parameters, e. g. the playback delay),

- the amount of bandwidth which is needed for preloading (see section 3.5.4 and 4.6).

For the receiver system, only the following parameters are of interest:

- the number and bandwidth of the channels used for the transmissions,
- the amount of bandwidth which is needed for preloading.

To describe the bandwidth requirements of a transmission scheme, it is sensible to describe the sender and receiver bandwidth requirements in multiples of the media stream bit rates and to mention if it is possible to adapt the transmission scheme to a specific receiver system bandwidth (probably with impacts to other parameters, e. g. the sender system bandwidth requirements or the playback delay).

Storage System Requirements

The size requirements of the storage system at the sender side are very obvious: The media-on-demand sender system must be able to keep all media streams of the offered media assortment into its media database. The throughput of this storage system equals the sender system bandwidth (ignoring the fact that parts of the media stream may reside in the file system cache if they are transmitted twice within short time).

Unfortunately, size and throughput are not the only requirements to the storage system: If the sender system is sending a lot of media streams (or a lot of segments of a single media stream) at the same time, the storage system must be able to perform this many direct accesses to the media streams. For systems using hard disks, this requires a lot of seek operations which slows down the throughput of the storage system. If we assume a fixed amount of memory for read-ahead for a whole media stream transmission, the only value which influences the seek rate is the number of media stream positions from which data is read in parallel during the transmission:

One position per channel: The media stream data is read from one position for each transmitted channel.

One position per slot: The media stream data is read from several positions for each transmitted channel, but only from one position for each slot on the channel.

Multiple positions per slot: The media stream data is read from several positions for each transmitted channel, even for each slot on the channel.

The size requirements of the storage system at the receiver side depend on several parameters:

- The size of the received media streams,
- the amount of the media stream which must be buffered (which depends on the used transmission scheme),

- the time the received media streams should be available after reception,
- the amount of storage which is needed by preloading.

For the classification of the receiver system storage requirements, we use the following classes:

No media stream buffering: The receiver system does not store any media stream data except for jitter compensation.

Partial media stream buffering: The receiver system must buffer part of the media stream.

Configurable media stream buffering: The receiver system must buffer part of the media stream, the amount can be configured freely (but may influence other parameters, e. g. the sender bandwidth requirements)

Full media stream buffering: The receiver system must buffer the whole media stream.

Partial media stream buffering and partial preloading: The receiver system must buffer part of the media stream and additionally the beginning of several/all media streams of the media assortment.

Configurable media stream buffering and partial preloading: The receiver system must buffer part of the media stream, the amount can be configured freely; additionally the beginning of several/all media streams of the media assortment must be stored.

Full preloading: The receiver system must buffer all media streams of the media assortment.

The throughput of the receiver storage system equals the receiver system bandwidth plus the bit rate of the played media streams (again ignoring the file system cache).

Similar to the sender system, direct access operations are required for storing all received segments of each media stream (either when receiving them or when assembling them for playback). The seek rate of the receiver system can then be classified as:

One position per channel: The media stream data is written to one position for each received channel.

One position per slot: The media stream data is written to several positions for each received channel, but only from one position for each slot on the channel.

Multiple positions per slot: The media stream data is written to several positions for each received channel, even for each slot on the channel.

Some transmission schemes do not require any data to be stored, but on the other side, they require a high sender system bandwidth. Other transmission schemes concentrate on lowering the sender system requirements but have high receiver system requirements and storage requirements. In general, storage and bandwidth requirements of sender and receiver are contrary parameters of a

media-on-demand transmission scheme and it is not possible to keep them all low at the same time. An ideal media-on-demand transmission scheme should therefore be adjustable in the parameters to find the right compromise between these parameters.

2.4.2 Functional Classification for Media Transmission Systems

In this section, several classifications are introduced which can be used to grade a media transmission system. While the technical classifications of the last section are almost only relevant for providers, most of the functional classifications are interesting for the consumer, too, as these classifications specify the capabilities of the media transmission system.

Media Scheduling

Requesting a media playback can be as simple as tuning into an ongoing transmission, but often a more complex procedure is required, e. g. sending a request to a remote system and negotiating the playback. Throughout this thesis, the following terms are used for describing the different classes for media scheduling:

Prearranged: The media transmissions are scheduled to fixed dates. No request processing from client systems is possible; the media transmission has to be joined at the scheduled transmission time to be received.

Pro-active media-on-demand: The media transmission is performed permanently. No request processing from client system is required; the media transmission can be joined at any time (probably with a reasonable playback delay).

Reactive media-on-demand: The media transmission is performed on demand. The media transmission can be joined only after a request has been issued (possibly with a reasonable playback delay).

Combinations: For example, the media transmission can be joined at any time when a long playback delay is accepted, but by sending a request, an additional transmission is scheduled. Other combinations (e. g. combining prearranged with reactive media-on-demand) are also possible.

Interactivity Levels of Media Transmission Systems

In [GKSW91], the authors classified media-on-demand and interactive TV services by their level of interactivity into the following five categories:

Broadcast (No-VoD): *No interactive control.*

Pay-per-view (PPV): *Viewer signs up for specific programming.*

Quasi video-on-demand (Q-VoD): *Users are grouped based on a threshold of interest, users can switch to a different group.*

Near video-on-demand (N-VoD): *Limited interactive control, e. g. rewind and fast forward in steps of 5 minutes.*

True video-on-demand (T-VoD): *Full control over playback.*

Although this classification allows a simple classification of media-on-demand and interactive TV services of the early nineties, it brings several drawbacks and limitations when classifying today's media-on-demand services:

- Different types of classification are mixed together, e. g. communication type (broadcast), accounting type (pay-per-view), media scheduling (on-demand) and playback control (rewind, fast forward).
- The selected classes only represented the available types of media-on-demand/interactive TV services of the late eighties/early nineties (e. g. quasi video-on-demand describes Batching transmissions (see 3.3.2) and near video-on-demand describes channel changes of Staggered transmissions (see 3.4.2)). Later developments cannot be assigned to any of these classes (e. g. it is not possible to find a class for a transmission scheme which delivers a media on request with a short delay but allows nearly no other type of interactivity which is common for pro-active transmission schemes).
- The classes do not allow a precise specification of the possible types of interactivity. For example, a class where the user can pause or rewind the playback but cannot use fast forward is not given.
- The used terms unnecessarily cause the impression that the transmission system can only be used for one type of media content (video).

For a better classification of the media-on-demand schemes, a more fine-grained classification of interactivity is beneficial. Because of the huge breadth of different types of interactivity, the interactivity classification is split into two distinct classes[5]:

Playback Control: Playback control means the navigation on the media stream in a VCR-like manner, e. g. pause, fast forward, rewind, slow motion, etc., but also jumping to a scene in the past or future of the media stream. To specify a playback control class completely, one normally would have to enumerate all types of possible playback control. Fortunately, the support for different types of playback controls are not unrelated to each other in media-on-demand transmission schemes, so only the following few classes are of interest:

[5]The other aspects of the quoted classification — communication type, back-channel support and accounting type — are presented separately.

No playback control: No playback control is supported.

Fixed jump playback control: No playback control is supported except for jumping forward or backward for a large, fixed amount and pausing for a long, fixed time.

Playback control within received part: All types of playback control which navigate within the transmitted part of the media are supported. This includes pause, slow motion and rewind, but also fast forward up to the point of the initial transmission. Jumping to specific locations in the media (either specified by time or chapter) and search forward and backward are possible, too (with the same restriction as for fast forward). This is the maximum possible level of playback control for a live transmission and typically the maximum possible level of playback control for pro-active and prearranged transmissions. (Today's digital video recorders often support this kind of interactivity as part of the time shifting features.)

Full playback control: All playback controls can be used, including jumping to specific locations (e. g. using an index of scenes or selecting a specific chapter by number or time) in past and future parts of the media stream.

The level of playback control can always be raised up to rewind interactivity if the receiver system has the ability to store the whole received part of the media stream while it is received at single playback rate.

Content Navigation: Content navigation can be distinguished from playback control by the type of interaction: While playback control only influences the speed of the playback and repetitions of it, content navigation changes the displayed content of the media stream itself. For example, playback control requests can be used to see the ending of a movie at the beginning of the playback, but content navigation requests can be used to select an alternative ending for the movie. In this sense one can regard playback control as a way to navigate on continuous media streams while content navigation is used for non-continuous navigation (see also section 2.5.1 for a detailed description of the media content structure). Within this thesis, the following classification for content navigation is used:

No non-continuous navigation: No content navigation is supported.

Branch navigation: If the media stream contains branches (e. g. alternative scenes, view angles), the consumer is able to choose which fork she wants to follow.

Multi-branch navigation: If the media stream contains branches, the consumer is able to switch between the branches at any time or display several branches at once ("Picture-in-Picture").

Full content navigation: Any type of content navigation is possible (e. g. virtual reality).

Many other types of content navigation are conceivable (e. g. accessing extra information, change of language, interactive shopping, interactive advertisements, navigating between different media streams [LV93]) but they can be reduced to the above given list of interactivities (e. g. access to extra information can be viewed as a branch in the media), a combination from the reception of several media streams (e. g. if one audio stream per language and one video stream are transmitted in parallel, a change of the language is simply a change of the audio stream) or to activities which do not influence the design of a media-on-demand transmission schemes (e. g. if payment information have to be transmitted for interactive shopping).

Dynamic Schedule Changes

For pro-active transmissions, the provider has to decide which media streams she wants to transmit and divide the available bandwidth to these transmissions. Therefore, the provider has to configure a bunch of properties which influence the bandwidth of a transmission, e. g. the media stream encoding and bit rate, the transmission scheme to use, the maximum tolerated playback delay and the amount of added redundancy for error correction.

For many of these decisions, the provider does not want to tie himself down to fixed values. For example, the popularity of media streams may depend on the daytime, the day of week, special events or even the weather. The flexibility and dynamic adaptability of the used transmission scheme is therefore described by the dynamic scheduling classification:

No dynamic change support: None of the parameters of a transmission can be changed dynamically.

Dynamic bandwidth and playback delay change support: The bandwidth and the playback delay can be changed dynamically. Typically, bandwidth and playback delay are oppositional parameters, i. e. the bandwidth increases when the playback delay is decreased and vice versa.

Dynamic stream duration prolongation: For live transmissions, the total media stream duration may be unknown in advance, thus it may be necessary to keep the media stream duration dynamic.

Dynamic content support: For some transmissions, it may be useful to exchange the content of the transmission dynamically: For example in a news transmission, single contributions may be replaced by newer ones.

Depending on the transmission scheme, these changes may come into effect immediately or after some delay. Furthermore the impact of dynamic changes may differ as some transmission schemes perform less efficient if dynamic changes are used or even made possible.

Variable-Bit-Rate Media Support

When a media stream is to be encoded, the producer has great latitude how to encode the media stream: Besides of selecting a media format (see section 2.5.3), it is often possible to select the encoding quality of the resulting media stream. Typically, the encoding quality is determined by configuring a target bit rate.

Many media formats use lossy encodings which remove some of the highest details when the target bit rate is too low. These media formats often allow two different configurations:

Constant-bit-rate (CBR): The media stream has a (nearly) constant bit rate.

Variable-bit-rate (VBR): The bit rate of the media stream varies over time.

Although the variable-bit-rate encoding provides the higher quality of these two (for the same average bit rate), it is not supported by every transmission scheme. In this case **smoothing** to constant-bit-rate (see section 4.8) can be utilized as a bypass.

Live Streaming Capability

Live streaming means the capability to transmit a media stream which is just captured and encoded. The reason why live streaming is mentioned for classification is that live transmissions impose several problems to the transmission scheme: Many of the transmission schemes assume that the media stream is completely known and available at media startup, but for live transmissions this is not true:

- The duration of a live transmission may be unknown.
- The media content is not known in advance, only the yet captured and encoded part of the media stream is accessible, which disallows the transmission of later parts of the media stream.
- Any additional information about the media stream is unknown for the remaining part, e. g. the time, duration and strength of bandwidth variations of a variable-bit-rate media stream are unpredictable.

While the first item objects the dynamic stream duration prolongation capability which has already been mentioned above, the latter ones require a further classification:

Not live streaming capable: The transmission scheme does not support live streaming at all.

CBR live streaming capable: The transmission scheme supports only live streaming of constant-bit-rate media streams.

Full live streaming capable: The transmission scheme supports live streaming of CBR and VBR media streams.

Similarly to live streaming, near-live streaming means the capability to transmit a media stream with a short delay after capture. As near-live streaming does not lower the requirements of live streaming to the media transmission system significantly no further classification is introduced for near-live streaming. The main advantage of near-live streaming to real live streaming is an improved result for smoothing of bandwidth variations.

Accounting Type

Financing of media-on-demand services is possible in different ways:

Free access: The service may be accessible for no charge. This may apply if the service is state-subsidized or financed through commercials.

Pay-per-view: The consumer has to pay for each requested media stream transmission.

Pay-per-subscription: The consumer pays regular fees for the media streams of a media stream set.

Combinations: Any type of combination is possible, too: For example, a pay-per-view system with additional regular fees or a service where the consumer can select between advertising sponsored transmissions and paid, ad-free ones.

Besides of the requirement of a back-channel for pay-per-view and of access restrictions (e. g. using encryption) for non-free access, these accounting types have no effects on media-on-demand transmissions and are not examined in this thesis.

2.5 Media Content and Format

The most important value for media-on-demand systems is the media itself. The media content may be as simple as a continuous video stream but also as complex as any graph-structured, piece-wise continuous, real-time data collection, containing alternative endings, different view angles, selectable languages and subtitles, breaks for commercials, dynamically updated parts, etc. The first subsection of this section gives an overview of the different structures for media content of media-on-demand systems.

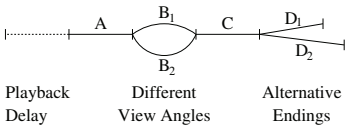
Media streams can carry different content types of media, e. g. movies, news, or radio plays, and can use a manifold of different encodings. In subsections 2.5.2 and 2.5.3, several media types and encodings are presented, resp., and their structure is analyzed. Additionally, properties like variable-bit-rate and fault tolerance are reviewed.

2.5.1 Structure of Media Content

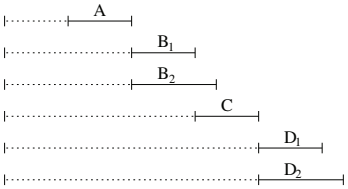
The basic structure for the content of a media-on-demand system is a **real-time data stream**. The term "real-time" herein means that a timing constraint exists: If the media content is split into small pieces, it is possible to assign time stamps to each of them, which describe the time offset relative to the beginning of the playback when the respective pieces are needed at the receiver system[6]. The term "data stream" indicates that the media stream must have a continuous structure, i. e. that the time stamps of the pieces are monotonously increasing.

Although the continuous structure is typical for media-on-demand content, this requirement can be loosened: A reactive system which handles each receiver system separately can support any type of media structure (including virtual reality). But even for pro-active systems, it is possible to loosen the continuity requirement:

To support piecewise continuous real-time data streams, every continuous piece can be thought of as a separate transmission, using a playback delay equal to the first possible playback time of the piece[7] after the media playback has been started (see figure 2.5 for an example).



(a) Example of a piecewise continuous media stream



(b) Decomposition into several continuous media stream transmissions

Figure 2.5: Example for decomposition of a piecewise continuous media stream

Allowing piecewise continuous real-time data streams, it is possible to support all real-time media data where the continuous pieces can be arranged as edges of a directed graph. This allows handling the following media structures:

- Branches
- Joins
- Alternatives
- Insertions

[6]Exactly, it is possible to state that every data transmission has a real-time requirement which matches this definition by stating that the whole file is needed at some time after the download has been started. Consequently, a good media-on-demand transmission scheme would find the same method for transmitting files as used in content distribution systems: Depending on the type of connectivity either point-to-point transmission (via one-to-one connections) or a round-robin broadcast (via one-to-many connections). This way a media-on-demand system can be regarded as a generalization of content distribution systems.

[7]Depending on the transmission protocol it may be possible to combine these separate transmissions into a single, comprehensive one.

Even loops are possible this way (but very uncommon).

Beneath the static structure of the media stream, the content may also change dynamically: The simplest change is the replacement of one media stream by another (e. g. substitution of a news transmission by an updated recording), but replacing only parts of a media are possible, too (e. g. updating individual contributions in a news transmission).

2.5.2 Content of Media Streams

When regarding the media stream content instead of its structure, several commonly used types can be found:

- Video streams (movies, news, ...): The mostly used media content type for media-on-demand systems will be video streams[8]. The video stream may be as simple as a continuous stream (as it is contained on video cassettes) but piecewise continuous stream features that are known from DVDs can be supported, too, like
  - alternative endings (branches)
  - alternative beginnings (joins)
  - different view angles (alternatives)
  - director's comments, deleted scenes (insertions)
- Audio streams (music, radio plays, news, ...): Audio streams behave very similar to video streams, but have a much lower bandwidth; therefore the bandwidth requirements are much lower for audio-on-demand than for video-on-demand.
- Text streams (books, tickers, subtitles): Except for subtitles to a video stream, text streams are very uncommon for media-on-demand transmissions: Although they are normally continuous streams, a real-time requirement is very artificial (due to different and varying speed of readers). Another issue is that pure text is very compact (a typical reader will read at most some tens of characters per second) and only few large texts will be read without interruption, so downloading the text entirely (using either pull or push technology) probably gives nearly the same result with much less effort.

2.5.3 Media Encodings

Because of the high bit rates of typical media streams (esp. of video streams), media-on-demand systems utilize **media compression** to reduce bandwidth requirements. Media compression can

[8] According to the common usage, the term "video stream" hereby means an "audio-visual stream".

work in two ways, utilizing **spatial** and **temporal redundancies**, e. g. in case of video compression, similarities within a frame (spatial) or between frames (temporal) are registered and encoded. Although removal of redundancies allows creating **lossless** media compression schemes, nearly all media compression schemes generally work **lossy**, i. e. some information is lost through compression and cannot be retained from the compressed media stream at decompression time. Fortunately, human senses are too imprecise and inert to recognize many of these changes as long as only details are affected. Many media compression schemes are even specialized to human recognition to improve the compression ratio without lowering the recognizable quality.

From the point of cost-effectiveness, the media encoding has to be selected very carefully: A higher compression ratio reduces the required bandwidth at the media-on-demand sender system and therefore reduces the costs for transmissions or increases the possible number of concurrently active playbacks. A higher compression ratio also lowers the bandwidth requirements of the receiver system, reducing the hardware and connectivity requirements for media-on-demand receiver systems. On the other side, high compression rates require more complex encoding hardware at the sender system (which may get more relevance in case of live transmissions) and decoding hardware at the receiver system, which conversely increases the costs for media-on-demand systems.

Today, MPEG-2 [Itu00] decoder chips are very cheap (as they are produced in huge amounts for DVB receiver systems and DVD players) and MPEG-4/Part 2 [Itu04] and H.264/MPEG-4 AVC [Itu05b] decoder chips are available for reasonable prices today, too[9].

But not only is the total bit rate of the media stream of interest: One other important issue is the varying of the bit rate. Some media encoders produce a media stream at constant-bit-rate, which can be handled by all media-on-demand transmission schemes, but if a variable-bit-rate encoding is used, a higher quality can be obtained for the same average bit rate. Unfortunately, not all transmission schemes support VBR media streams, so it is necessary to smooth the media stream bit rate variations in these cases, loosing a lot of efficiency (see section 4.8).

Another problem is fault tolerance of the media encoding: Even if some type of error correction is used, it is not possible to prevent losses completely (without blocking the playback for an indefinite time). As a consequence only encodings should be used which can handle losses within the media stream by re-synchronizing at the next logical boundary (e. g. frame or group-of-pictures in case of video data).

[9] For example, Sigma Designs offers a chip which supports VC-1, WMV9, H.264/MPEG-4 AVC besides MPEG-2 and MPEG-4/Part 2 as well as a wide range of audio formats (including Dolby Digital, WMA, WMA Pro, AAC and MP3), encryption (including AES, DES, Triple-DES, RC4, CSS, DVB-CSA and Multi-2) and many interfaces (Ethernet 10/100, USB 2.0, IDE, PCI) [Sig06].

2.6 Summary

This chapter provided a deep insight look into the general functioning of media-on-demand systems: Firstly, many terms related to media-on-demand have been defined and explained, the structure of media-on-demand sender and receiver systems has been shown and commented on by decomposing them into functional units and the creation of hierarchical media-on-demand systems has been proposed as a first, structural enhancement.

Secondly, several classifications have been proposed for media-on-demand systems. Many of them, esp. the technical ones, determine if a specific type of media-on-demand system can be used in a prevailing setup or which changes are required, e. g. the communication type, while the functional classifications can be used to differentiate the provided service and utilizability of a media-on-demand system.

Finally, the structure and content of media streams have been analyzed and properties of encodings have been outlined.

In the next chapter, the functioning of different types of media-on-demand systems will be analyzed and compared to each other, using the classifications introduced in this chapter to characterize the functional and technical attributes of each system.

Chapter 3

Media Transmission Schemes

IN the previous chapter, an introduction to media transmission systems has been presented, covering the structure and operation of sender and receiver systems, an analysis of content types and encodings of transmitted media streams and an examination of different network types which can be used for media transmissions. This chapter focuses on one single (but important) part of a media transmission system: the **media transmission scheme**.

The media transmission scheme of a media transmission system describes how a media stream is transferred from a sender system to receiver systems. Currently, a wide range of transmission schemes have been proposed, and the goal of this chapter is to provide an overview about their functioning and basic approaches, their differences and commonalities and their advantages and drawbacks.

The main goal thereby is not to present a detailed and complete description of every transmission scheme but to show the basic idea behind each of them. This understanding of the differences between the schemes is particularly important to comprehend the evaluation and comparison process of the transmission schemes which concludes each of the sections describing transmission schemes.

For a further description, the classes which have been introduced in the section 2.4 are used to classify each presented transmission scheme. Therefore, a classification table follows each presented transmission scheme, containing the following entries:

- Technical classifications (see section 2.4.1 for a detailed description):

  Communication type: The communication type which is used by the transmission scheme.

  Back-channel: Indication if a back-channel is required by the transmission scheme.

  Media-on-demand topology: Topology of a media-on-demand system using this transmission scheme.

Sender bandwidth: Amount of bandwidth needed at the sender system when sending a media stream using this transmission scheme.

Receiver bandwidth: Amount of bandwidth needed at the receiver system when receiving a media stream using this transmission scheme.

Receiver storage size: Amount of storage needed at the receiver system when receiving a media stream using this transmission scheme.

Sender storage seeks: Amount of storage seeks needed at the sender system when sending a media stream using this transmission scheme.

Receiver storage seeks: Amount of storage seeks needed at the receiver system when receiving a media stream using this transmission scheme.

- Functional classifications (see section 2.4.2 for a detailed description):

Media scheduling: Type of scheduling of media streams for this transmission scheme.

Playback control: Types of playback control natively supported by this transmission scheme.

Content navigation: Types of content navigation natively supported by this transmission scheme.

Dynamic schedule changes: Indication if changes to the schedule for ongoing transmissions are possible.

Dynamic content changes: Indication if changes to the content for ongoing transmissions are possible.

Variable-bit-rate media support: Indication if the transmission scheme supports variable-bit-rate media streams without smoothing to constant-bit-rate.

Live streaming capability: Indication if the transmission scheme allows live (or near-live) transmissions.

Important entries (signalling restrictions or extensions) are marked with an exclamation mark (**!**) at the left, entries with mark severe restrictions with three exclamation marks (**!!!**).

Besides the textual (and sometimes mathematically supported) descriptions, diagrams are commonly added for better understanding, showing the internal structure of the transmission scheme or presenting an example transmission in a graphical notation. The used diagram types are introduced in the subsections where they are used for the first time, in particular

- channel-oriented diagrams in section 3.3.1 on page 58,
- stream-oriented diagrams in section 3.3.3 on page 63,
- enumerating transmission schedules in section 3.7.2 on page 114,

- tabular transmission schedules in section 3.7.2 on page 116,
- rectangular transmission schedules in section 3.7.2 on page 115,
- tree transmission schedules in section 3.7.2 on page 116.

Before starting with the presentation of existing transmission schemes, the following section 3.1 provides the mathematical background for an efficiency analysis of the required bandwidth of the schemes which is used to rate the schemes. In the six sections following this one, many different schemes are presented:

Starting with some traditional media transmission schemes in section 3.2, this chapter continues with a group of reactive media-on-demand transmission schemes in section 3.3 and strides ahead to the — most interesting — pro-active media-on-demand transmission schemes, which are presented in sections 3.4 to 3.7. Finally, section 3.8 presents the last group of transmission schemes which combine reactive and pro-active approaches.

In the last section of this chapter, a short summary and overall comparison of the presented transmission schemes and enhancements is given.

3.1 Efficiency of Media Transmission Schemes

Each of the following sections concludes with a comparison of the proposed transmission schemes. The most important issue for a comparison of different schemes is the **transmission efficiency**, i. e. the bandwidth consumption for a specific setup (e. g. the required sender bandwidth for a given maximum playback delay in case of pro-active schemes or for a given maximum playback delay and for a specific client request pattern in case of reactive schemes) compared to the theoretical lower limit of bandwidth which is needed to serve the same setup.

Bandwidth requirements of a transmission scheme depend very much on the prevailing conditions: While the media bit rate influences the transmission bandwidth requirements linearly and thus can be ignored in an analysis of the transmission scheme, the efficiency of a particular transmission scheme is mainly influenced by parameters like playback delay and the request pattern. For these reasons, typical request patterns for media-on-demand transmissions are examined in the next subsection.

For a comparison of transmission schemes, an efficiency value is used which can be visualized for different preconditions in charts at the end of each following section. This efficiency value η is calculated by

$$\eta = \frac{B^-}{B} \tag{3.1}$$

where B is the average bandwidth consumption of the transmission scheme and B^- is the minimum required bandwidth for the same setup.

As $B \geq B^- > 0$ is always fulfilled, the resulting value range of the efficiency is $0 < \eta \leq 1$, where $\eta = 1$ means a perfect transmission scheme in respect to its average bandwidth consumption.

This comparison introduces a question which has first been formulated in [EVZ99a] for reactive transmission schemes and in [ES01] for pro-active schemes: *What is the minimum required bandwidth to serve all requests of a given media-on-demand setup?* In subsections 3.1.2 and 3.1.3, this question is discussed and solutions for different preconditions and types of transmission schemes are presented.

3.1.1 Request Pattern for Media-on-Demand Transmissions

The minimum required bandwidth of reactive media-on-demand transmissions is strongly influenced by the request pattern for the media streams. As requests for media-on-demand transmissions are in mostly any case not predictable, random processes have to be used when evaluating transmission schemes mathematically. When request patterns for media streams are analyzed, typically two probability distributions have to be examined:

The first distribution describes how customers' requests are spread over the provider's media assortment. This distribution depends very much from the topicality and popularity of the media streams as well as from advertisements or sales discounts, but also from time, date, season, weather or competitive offers [LV93].

Interestingly, if the probabilities are not artificially impaired too much by the provider, the distribution is nearly identical in mostly any case: Media streams in media-on-demand systems are requested approximately according to the Zipf distribution with a parameter $s \approx 0.729$ [DPK94, DSS94] where the Zipf distribution is defined through the probability function [KRB06a, Wik08f]

$$P_s[X = n] = \frac{n^{-s}}{\sum_{i=1}^{N} i^{-s}} \tag{3.2}$$

In this formula, X is the variable of the random process, N is the total number of media streams in the media assortment, $1 \leq n \leq N$ is the rank of the examined media stream and s is the parameter of the distribution. The resulting distribution is shown in figure 3.1 for different numbers of media streams.

In the context of media-on-demand, the most important fact about this distribution is that most requests are performed for a very low number of media streams: For example in case of an assortment consisting of 1 000 media streams, about 36.2 % of all requests are sent for the most popular 5 % of the streams. Table 3.1 gives further examples for this distribution.

The other distribution of relevance describes how requests for a single media stream are distributed along time. Although the request rate for a single media stream varies over time (which

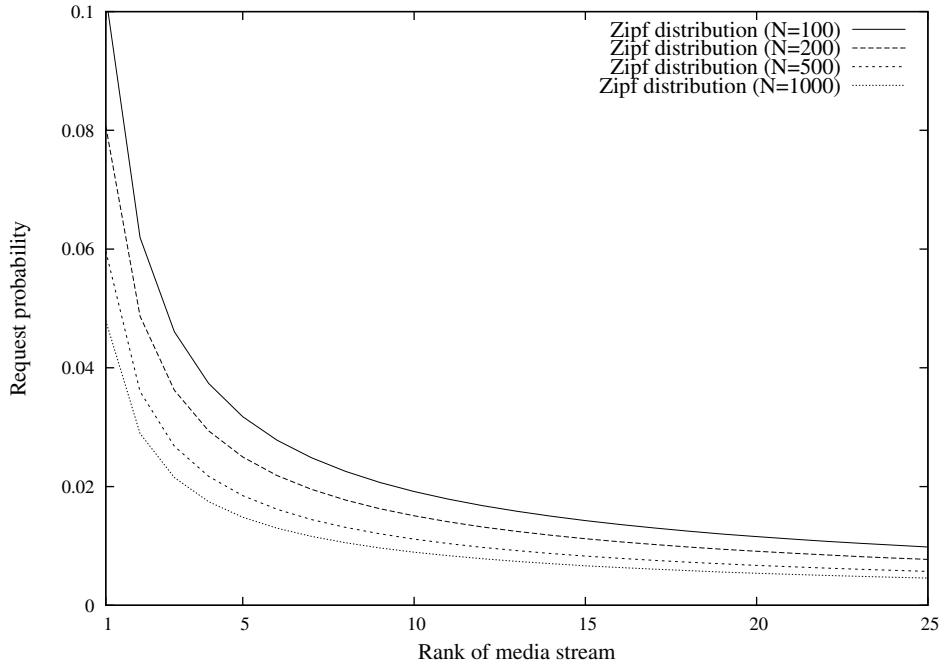


Figure 3.1: Zipf distribution with parameter 0.729 for media stream popularity

Number of requested media streams	Fraction of requested media streams	Fraction of requests
1	0.1 %	4.8 %
2	0.2 %	7.6 %
5	0.5 %	13.1 %
10	1.0 %	18.4 %
20	2.0 %	25.1 %
50	5.0 %	36.2 %
100	10.0 %	46.7 %
200	20.0 %	59.4 %
500	50.0 %	80.3 %

Table 3.1: Request frequencies for the most popular parts of the media stream assortment for a assortment of 1 000 media streams

may even change the popularity ranking of the media stream and shows up in the Zipf distribution), short term observations show that requests are Poisson distributed, a result which shows a high independence of request times of different customers[1].

The Poisson distribution [KRB06b, Wik08d] is parameterized by a single parameter λ and is

[1] Other distributions may occur if the transmission is time-bounded, e. g. most people would join the eight o'clock news at the same time for custom reasons or join a football transmission with curiosity nearly without delay.

defined by the following probability function

$$P_\lambda[X = n] = \frac{\lambda^n \cdot e^{-\lambda}}{n!} \tag{3.3}$$

where X is again the variable of the random process, n is the number of requests during a predefined period and e is the base of the natural logarithm ($e \approx 2.718\,281\,828$). Figure 3.2 shows some examples for Poisson distributions using different parameters.

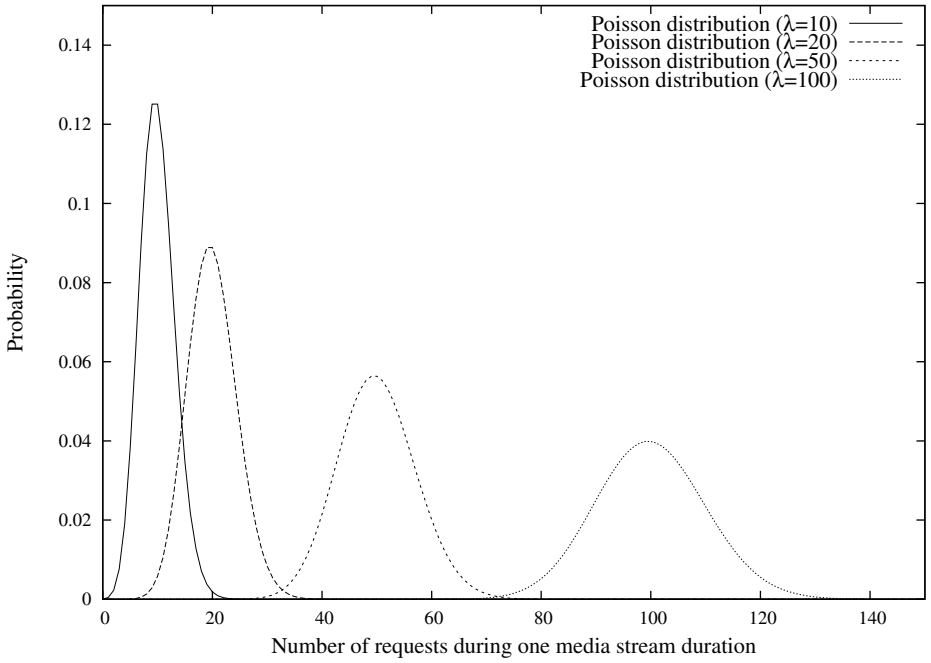


Figure 3.2: Poisson distribution for media stream requests

3.1.2 Minimum Required Bandwidth for Reactive Transmission Schemes

For reactive transmission schemes and a limited receiver bandwidth of R times the (constant[2]) media bit rate b, the authors of [EVZ99a, EVZ01] presented the following formula assuming Poisson distributed requests of parameter λ:

$$B^- \approx b \cdot \eta_R \cdot \ln\left(1 + \frac{\lambda \cdot d}{\eta_R}\right) \tag{3.4}$$

[2]Bandwidth variations are typically ignored when calculating the bandwidth requirements of a particular transmission scheme. See section 4.8 for impacts of variable-bit-rate media streams to transmission schemes and the efficiency loss which has to be expected when the transmission scheme does not support variable-bit-rate media streams natively.

where η_R is the positive real constant which satisfies the following equation:

$$\eta_R \cdot \left(1 - \left(\frac{\eta_R}{\eta_R + 1}\right)^R\right) = 1 \tag{3.5}$$

This formula has been obtained by assuming that infinitely small segments are transmitted on channels at the media bit rate and calculating the transmission frequency of each segment from probabilities. Table 3.2 shows the value of η_R and $\frac{B^-}{b}$ for some values of R and λ.

R	η_R	$\frac{B^-}{b}$ for $\lambda = 10$	$\frac{B^-}{b}$ for $\lambda = 100$	$\frac{B^-}{b}$ for $\lambda = 1\,000$
2.0	1.618	3.190	6.699	10.401
3.0	1.191	2.669	5.292	8.023
4.0	1.078	2.512	4.895	7.367
5.0	1.035	2.450	4.742	7.116
∞	1.000	2.398	4.615	6.909

Table 3.2: Value of η_R and the bandwidth requirements of the sender system $\frac{B^-}{b}$ for different values of receiver bandwidth R and request frequencies λ

If the segments are transmitted on channels at a bandwidth below the media bit rate instead, more channels are needed, but the total required bandwidth at the sender system is even lower. The authors of the above equation provided in [EVZ01] the result if the channel bandwidth is reduced to $0 < C < 1$ times the media bit rate:

$$B^- \approx b \cdot \eta_{R,C} \cdot \ln\left(1 + \frac{\lambda \cdot d}{\eta_{R,C}}\right) \tag{3.6}$$

where $\eta_{R,C}$ is the positive real constant which satisfies the following equation:

$$\eta_{R,C} \cdot \left(1 - \left(\frac{\eta_{R,C}}{\eta_{R,C} + C}\right)^{\frac{R}{C}}\right) = 1 \tag{3.7}$$

If C tends towards zero, the bandwidth at the sender system is minimized. The value of $\eta_{R,C}$ converges in this case to the constant $\eta_{R,\epsilon}$ which satisfies the following equation:

$$\eta_{R,\epsilon} \cdot \left(1 - e^{-\frac{R}{\eta_{R,\epsilon}}}\right) = 1 \tag{3.8}$$

Table 3.3 shows the value of $\eta_{R,\epsilon}$ and $\frac{B^-}{b}$ for some values of R.

R	$\eta_{R,\epsilon}$	$\frac{B^-}{b}$ for $\lambda = 10$	$\frac{B^-}{b}$ for $\lambda = 100$	$\frac{B^-}{b}$ for $\lambda = 1\,000$
1.0	∞	10.000	100.000	1 000.000
1.2	3.188	4.527	11.085	18.335
1.5	1.716	3.296	7.004	10.929
2.0	1.255	2.753	5.510	8.386
2.5	1.120	2.571	5.044	7.613
3.0	1.063	2.491	4.843	7.281
4.0	1.020	2.428	4.688	7.028
5.0	1.007	2.408	4.641	6.950
∞	1.000	2.398	4.615	6.909

Table 3.3: Value of $\eta_{R,\epsilon}$ and the bandwidth requirement $\frac{B^-}{b}$ of the sender system for different values of receiver bandwidth R and request frequencies λ

3.1.3 Minimum Required Bandwidth for Pro-Active Transmission Schemes

For pro-active transmission schemes, the minimum required bandwidth has been examined several times if no restriction is put on the recipient bandwidth, e. g. by the authors of [ES02] which provided the following formula for the lower bound of required bandwidth:

$$B^- \approx b \cdot \left(\ln\left(1 + \frac{d}{W^+}\right) + O\left(\frac{\delta}{W^+}\right)\right) \tag{3.9}$$

where B^- is the lower bound for the required bandwidth for the transmission, b is the bit rate of the media stream, W^+ is the (maximum) playback waiting time, d is the duration of the media stream and δ is the duration of each segment of the media stream.

In the more detailed article [ES01] of the same authors, this lower bound has been calculated exactly as:

$$B^- = b \cdot \left(\Psi_0\left(\frac{d + W^+}{\delta}\right) - \Psi_0\left(\frac{W^+}{\delta}\right)\right) \tag{3.10}$$

where Ψ_0 is the digamma function [Wal01a, Wei06] which fulfills $\Psi_0(x) = \sum_{i=1}^{x-1} \frac{1}{i} - \gamma$ where $\gamma \approx 0.577\,215\,665$ is the Euler-Mascheroni constant [Wal01b]. As this formula is based on information theoretic arguments, the bandwidth of no pro-active on-demand transmission scheme can fall below it. But as shown later (see subsections 3.6.4 and 3.7.1) it is possible for a pro-active transmission scheme to reach this lower bound.

For huge values of x, Ψ_0 can be approximated by $\Psi_0(x) \approx \ln x$ which gives the result of equation (3.9). But the above formula also allowed the authors to derive a more exact approximation

using a better approximation for Ψ_0:

$$B^- \approx b \cdot \left(\ln\left(1 + \frac{d}{W^+}\right) + \frac{1}{2\frac{W^+}{\delta}\left(1 + \frac{W^+}{d}\right)} + O\left(\frac{\delta^2}{W^{+2}}\right) \right) \tag{3.11}$$

In this thesis, always the exact formula of equation (3.10) is used to compute the efficiency of pro-active transmission schemes. Only when estimating the effects of some enhancements, the approximations are used.

Unfortunately, no lower limit has been given yet in literature when the receiver system is not able to receive all transmission channels at once. Therefore, the following model is used to calculate the lower limit:

- The media stream is split into equally sized segments.
- Each segment is transmitted round-robin on its own channel.
- All segments are transmitted using the lowest possible bandwidth. The bandwidth of each channel has to reflect the playback time of the segment and the joining delay of receiver systems for this channel.
- The receiver system joins the channels starting with the first one, up to the bandwidth limit of the receiver system.
- When the receiver has got all data of a channel, the channel is released and the regained bandwidth is used to continue joining of further channels.
- To find the bandwidth limit, the number of segments into which the media stream is split is increased infinitely.

The limit where this algorithm converges to can also be retrieved by moving to continuous functions. This yields to the following definitions:

1. Calculate the position in the media stream until which the receiver system can join the reception immediately:
$$t_0 = W^+ \cdot (e^R - 1) \tag{3.12}$$

2. Let $F(t)$ be the bandwidth characteristics of the media stream transmission at time t. Then for $t \leq t_0$, let $F(t)$ be
$$F(t) = \frac{b}{W^+ + t} \tag{3.13}$$

3. Let $\mathcal{F}(t)$ be an antiderivative of $F(t)$. Therefrom follows for $t \leq t_0$ that $\mathcal{F}(t)$ equals
$$\mathcal{F}(t) = b \cdot \ln(W^+ + t) + C \tag{3.14}$$

with some constant C.

4. For $t > t_0$, let $F(t)$ and its antiderivative $\mathcal{F}(t)$ be the solution of the following differential equation:

$$\mathcal{F}(t) = b \cdot R + \mathcal{F}\left(t - \frac{R}{F(t)}\right) \tag{3.15}$$

Then the required bandwidth for the transmission of the media stream from time offset t_1 to t_2 is

$$B^{-} = \int_{t_1}^{t_2} F(t)\,dt = \mathcal{F}(t_2) - \mathcal{F}(t_1) \tag{3.16}$$

Unfortunately, there is no closed formula known for equation (3.15) so it must be solved numerically.

Although no prove for the optimality of this model is given here, these approaches (which converge to the same results) can be seen as a straightforward extension of the Polyharmonic Broadcasting scheme which has been proven to be optimal (see section 3.6.4).

Figure 3.3 shows the so gained minimum bandwidth requirements at the sender system for different receiver bandwidths. The most interesting fact about the resulting bandwidth requirement is that the influence of the receiver bandwidth is much minor than expected: Even for a very short playback delay of $\frac{1}{7200}$ of the media stream duration (which equals 1 second for a two hour media stream), the bandwidth requirements for a transmission which considers a three channel receiver bandwidth capacity is less than 4 % higher than the bandwidth requirement without receiver bandwidth restriction. Because of this result, lowering the receiver system bandwidth requirements is studied in more detail for the Generalized Greedy Broadcasting scheme in section 4.18.

3.1.4 Efficiency Graphs

The media-on-demand transmission schemes which are proposed in this chapter react very differently on changes of parameters, and esp. the efficiency of some transmission schemes varies strongly for different sets of parameters.

To make the transmission schemes comparable, their efficiency is drawn into a graph at the end of each section. For reactive transmission schemes, the efficiency is drawn for request rates ranging from 1 to 1 000 per media stream duration, for pro-active transmission schemes, the efficiency is shown for maximum playback delays between $\frac{1}{7200}$ and $\frac{1}{5}$ of the media stream duration (which equals a delay between 1 second and 24 minutes in case of a two-hour media stream). This allows comparing the efficiency for different request rates or maximum playback delays as well as to observe the stability of the efficiency in case of parameter changes.

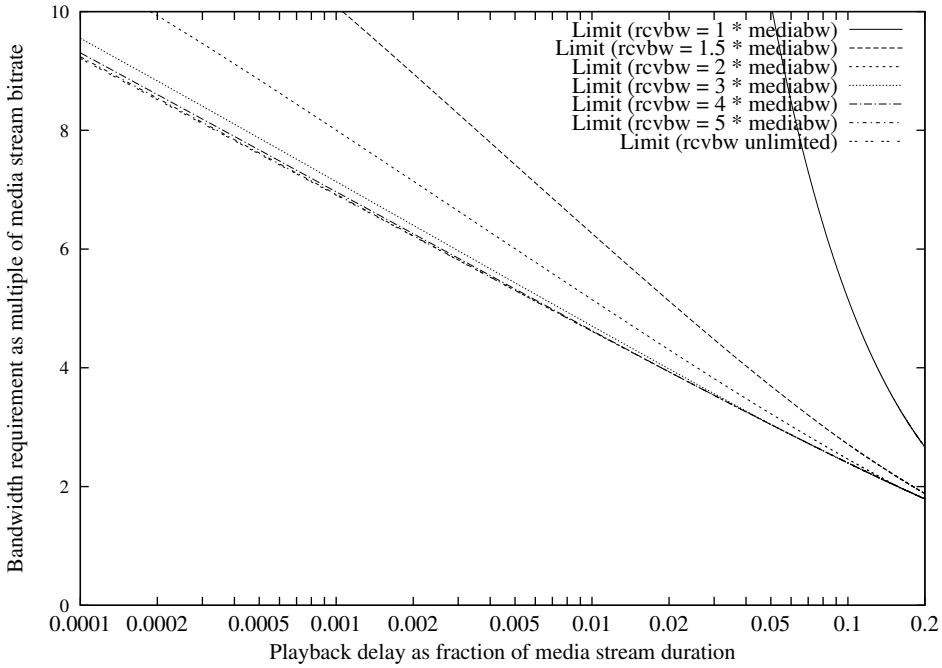


Figure 3.3: Minimum required bandwidth at the sender system for different receiver bandwidth capacities

3.2 Traditional Media Transmission Schemes

Before introducing "advanced" media-on-demand transmission schemes, this subsection presents some traditional media transmission schemes. The term "traditional" thereby is used to indicate that these schemes have been invented long time ago and provide simple and (partially) trivial solutions for media stream transmissions. Some of these systems do not even offer "on-demand" support, and others provide on-demand services using non-technical solutions. But for completeness and as these schemes are the origins of the media-on-demand idea, a short overview of these transmission schemes is given in this section.

3.2.1 Single Broadcast

Since the invention of radio and television broadcasting (in 1893 and 1925, respectively [Wik08c, Wik08a]), broadcasting organizations established and started to transmit media streams (traditionally news, music and movies) regularly. The schedule which is used by the broadcasting companies is usually very irregular: For a viewer it is mostly impossible to predict the schedule of a specific transmission. Nevertheless, single broadcasts are very popular today and are used broadly.

To make the transmissions more predictable for viewers, several solutions have been established:

- A TV/radio guide lists the schedule for the next week(s).
- Some transmissions are broadcast at regular times (e. g. many radio stations transmit news every full hour, and episodes of TV series are transmitted daily or weekly at the same time).
- Special events (e. g. sport events, concerts) are transmitted without delay (live) or with minor delay (near-live).
- For request programs, viewers can vote for their favorite movie which is then transmitted at a preallocated slot in the schedule. Similarly, many radio stations allow their hearers to send their music wishes by telephone, mail or Internet.

But howsoever, for a viewer it is nearly impossible to determine when a specific movie will be transmitted the next time, so this transmission scheme cannot be classified as “on-demand”.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Single media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Single media stream bit rate **Receiver storage size:** No media stream buffering **Receiver storage seeks:** —	**Media scheduling:** **!** Prearranged **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** Full dynamic content support **Variable-bit-rate media support:** Full dynamic content support **Live streaming capability:** Full live streaming capable

Table 3.4: Classification for Single Broadcast transmissions

3.2.2 Personal Tape Archive

Keeping a personal tape archive is a very simple form of traditional media-on-demand: It allows requesting a media from a given assortment within a relative short time (depending only on the cataloging and organization of the archive). Besides of the space consumption of a personal video assortment, the main disadvantage of this type of media-on-demand lies in the availability of media streams in the archive: Each media stream has to be bought or recorded from another source (e. g.

a previous broadcast transmission) before it can be viewed, so the assortment typically does not contain the newest media streams and is acquainted within short time.

Technical classification	Functional classification
Communication type: None **Back-channel:** None **Media-on-demand topology:** None **Sender bandwidth:** None **Sender storage seeks:** None **Receiver bandwidth:** None **Receiver storage size:** Full media stream **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** Full playback interactivity **Content navigation:** Multi-branch navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: **!** Based on manually administered archive	

Table 3.5: Classification for Personal Tape Archives

3.2.3 Video Rental Stores and Media Libraries

An alternative to personal tape archives are commercial or public ones: Video rental stores or media libraries. In contrast to personal tape archives, their assortment does not require any space at home, is usually up-to-date and contains many new movies. The main disadvantages of these solutions are that it takes some time and (sometimes) self conquest to fetch a movie and some luck that the wanted movie is available. Additionally, the movie has to be returned afterwards.

3.2.4 Complete Preloading of Media Streams

Complete Preloading of media streams (also known as “push video-on-demand”) means that streams are transmitted to the receiver system in advance (i. e. before a playback request is received) and stored on the receiver system until they are requested. Although this media transmission scheme has the best efficiency conceivable, it does not provide a media-on-demand access in the strict sense as the media reception is independent from the media request and the media stream is not received concurrently to the media playback.

Complete preloading has a lot of disadvantages: Firstly, the receiver system must have enough storage to save a whole copy of all media streams at once which imposes high costs for receiver

Technical classification	Functional classification
Communication type: None **Back-channel:** None **Media-on-demand topology:** None **Sender bandwidth:** None **Sender storage seeks:** None **Receiver bandwidth:** None **Receiver storage size:** Full media stream **Receiver storage seeks:** One position	**Media scheduling:** Reactive media-on-demand **Playback control:** Full playback interactivity **Content navigation:** Multi-branch navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: ! Based on physical delivery	

Table 3.6: Classification for Video Rental Stores and Media Libraries

systems (or limits the size of the media assortment badly). Secondly, the receiver system has to stay tuned on to the preloading transmission permanently to receive and store all pushed streams. Thirdly, the media streams on the systems have to be protected from being viewed or copied from the storage, esp. if payment is on a per-view basis.

Instead of preloading all media streams, it is also possible that the customer performs a selection some time before the playback is requested [WMS06]. The receiver system then can tune to exactly the selected media streams and store them until the playback is requested. While this reduces the storage requirements of the receiver system, it does not allow any spontaneous requests for media streams other than the selected ones.

3.2.5 Summary of Traditional Media Transmission Schemes

Traditional media transmission schemes do not provide a media-on-demand service in the strict sense: The transmission schemes do not provide the media stream on request, use physical delivery, require huge storage devices or a preselection of the transmitted media streams. Ignoring these problems, Complete Preloading provides the best approximation to media-on-demand.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Any (even lower than single media stream bit rate) **Sender storage seeks:** One position per channel **Receiver bandwidth:** Equal to sender system bandwidth **Receiver storage size:** Full preloading **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** Full playback interactivity **Content navigation:** Multi-branch navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: **!** Huge storage requirement for large media assortments	

Table 3.7: Classification for Complete Preloading

3.3 Reactive Media-on-Demand Transmission Schemes

After the presentation of these very simple, traditional media transmission schemes in the last section, this section advances to more modern solutions which provide real media-on-demand service.

In this section, several reactive media-on-demand transmission schemes are described. The first two of them are still very simple and transmit the media streams as a whole, but the following four schemes use partial transmissions to lower the bandwidth requirements. The transmission schemes presented in the last two subsections of this section differ from the above ones completely in the type of media distribution because the media stream is forwarded between recipient systems.

3.3.1 Point-to-Point Transmissions

For Point-to-Point transmissions (which are sometimes referred to as **streaming** [Wik08b] in the context of media transmissions), the media stream is transmitted to the receiver system just-in-time on request (probably with a delay if not enough resources are available to handle the request), using a one-to-one (Point-to-Point) transport connection. Point-to-Point transmissions allow the highest level of interactivity, as any request from the receiver system can be handled at the sender system without influencing other receivers. But on the other side, using one dedicated channel per active receiver system causes high bandwidth requirements for the sender system as the sender system must be able to attend all active recipients at the same time. Besides of the high bandwidth

requirements, the playback delay of this transmission scheme depends very much on the load of the sender system: If enough resources are available, the media stream can be served immediately, but otherwise the transmission has to be delayed until resources from another transmission are released.

For managing the available sender system resources, a simple scheduling strategy can be used, e. g. incoming requests may be appended to a first-in-first-out (FIFO) queue and removed from the queue whenever resources are available, but also more complex scheduling strategies are possible (e. g. priorizing premium customers).

To visualize this type of transmission scheme, a **channel-based diagram** is used: In channel-based diagrams, the transmitted part of the media stream of all channels are shown using bars, which are arranged in rows, one below the other. If the media stream has been cut into several segments, the bars may be labeled with indexes to name the segments they carry (see figure 3.20 for an example). If channels of different or varying bandwidth are used, the thickness of the bars is used to indicate the bandwidth (see figure 3.32 for an example).

Sometimes it is desirable to show a playback together with the transmission. In this case the received parts of the transmission are shaded (using different types of shading if several receptions are shown) and an additional bar (per visualized playback) below the time axis shows the time of playback. The time when the playback request has been issued by the receiver system is shown with a small triangle in front of the playback bar[3].

Figure 3.4 shows a channel-based diagram for three Point-to-Point transmissions, together with the playbacks of the recipients.

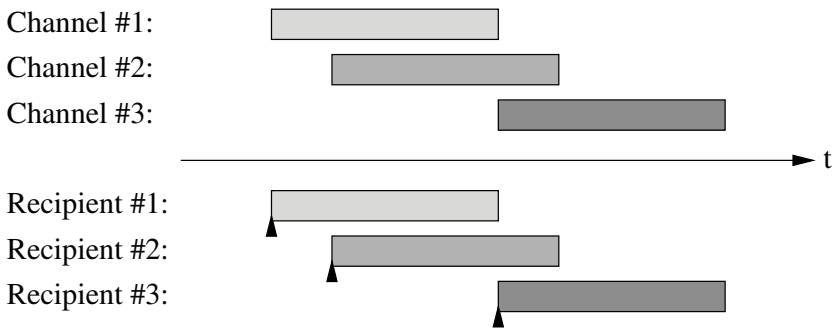


Figure 3.4: Diagram of a Point-to-Point transmission.

[3]Delays from network transmissions and for jitter compensation are not shown in these figures for better comprehensibility.

Technical classification	Functional classification
Communication type: ! One-to-one **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Single media stream bit rate **Receiver storage size:** No media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** Full playback interactivity **Content navigation:** Full content navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** Full dynamic content support **Variable-bit-rate media support:** Full dynamic content support **Live streaming capability:** Full live streaming capable
Comment: ! High bandwidth requirements at sender system for large setups	

Table 3.8: Classification for Point-to-Point transmissions

3.3.2 Batching

Batching is a very old and intuitive way to lower the bandwidth requirements of the Point-to-Point transmission scheme by using one-to-many transmissions. In Batching systems, several requests from receiver systems for the same media stream are served at once in a batch, using a single transmission.

As it is a seldom coincidence that several receiver systems ask simultaneously for the same media stream, receiver requests have to be collected for some time, delaying the playback for the first receivers. For making the decision when to start a transmission, several different strategies exist, always trying to make a compromise between fairness and the number of served recipients. To find a balance between fairness and the number of served recipients is very important for Batching schemes [GH90, BCMM00, CT99, PCL98a]: If the system weights fairness to high, it serves requests too inefficient and therefore decreases the number of served recipients. On the other side, if media streams with many outstanding requests are preferred to media streams with few requests, the consumers of the latter streams have to wait too long and may reject the service at all.

The most popular Batching algorithms for media-on-demand are:

- **First-come-first-served (FCFS)** [DSS94]: Requests are inserted into one waiting queue. Whenever a channel becomes available, the longest waiting request is removed from the queue and served (together with all other requests for the same media stream). This Batch

algorithm is the fairest one as the maximum playback delay is independent from the requested media stream.

- **Delayed First-come-first-served (Delayed FCFS)**: Same basic algorithm as in FCFS, but requests are never served immediately but always after a short delay, so the available bandwidth at the sender system is not exhausted as fast as when using FCFS.
- **Maximum-queue-length (MQL)** [DSS94]: For each media stream, a separate queue is maintained. When a channel becomes available, the queue with the most requests is searched and the media stream of this queue is sent. This algorithm has a much lower average service time, but handles requests for seldom media streams very unfair as these customers may have to wait a long time for their request being served. In case of high utilization of the sender system, requests for seldom media streams may even never get served.
- **Maximum-factored-queue-length (MFQL)** [AWY01]: Whenever a channel becomes available, the media stream with the highest factored queue length is scheduled where the factored queue length is the product of the queue length and of the reciprocal of the square root of the request frequency of the media stream. This strategy depicts a compromise between MQL (which does not consider that viewers may renege because of too long waiting times) and FCFS (which does not take queue lengths into account).
- **Prioritized Queuing**: FCFS-like Batching, but requests can additionally be prioritized to pass other requests with low priority. For example, requests of premium customers can be assigned a higher priority to provide lower waiting times to them.
- **Combinations** of a Batching scheme with *n* Staggered Broadcasting transmissions (e. g. FCFS-*n*, MQL-*n* [DSS94] or MFQL-*n*): The hottest *n* media streams are transmitted using Staggered Broadcasting (see section 3.4.2) and the remaining media streams are served using a Batching algorithm. As a consequence, the *n* hottest media streams have a fixed maximum playback delay while the remaining media streams can be handled as desired.
- **Combinations** of a Batching scheme with Single Broadcasting: As an alternative to queuing requests which cannot be fulfilled immediately, it is possible to allow the consumer switching onto the last requested transmission of the media stream. If the consumer chooses this option, she misses the beginning of the media stream but she is not required to wait until bandwidth becomes available for the next transmission of this media stream.

Besides of these ones, many schemes from other Batching systems may also be used for media-on-demand systems as well as other combinations or minor modifications of the above presented schemes.

Batching schemes can lower the bandwidth requirements a lot but they also reduce the interactivity which can be granted to each consumer compared to Point-to-Point transmissions as

any navigation would affect all consumers of the whole batch. This decoupling of media-on-demand sender and receiver systems is typical for media-on-demand systems which serve several consumers with a single transmission and is indispensable if the interacting user is not isolated from the remaining ones (or moved to another group [LLG97]) and if the interactivity cannot be performed solely on the media-on-demand receiver system (e. g. backward navigation can be provided by the receiver system if it has enough storage to save one copy of the media stream).

Figure 3.5 shows a transmission diagram for a Batching transmission (e. g. Delayed FCFS), together with the request times and the playback of three recipient.

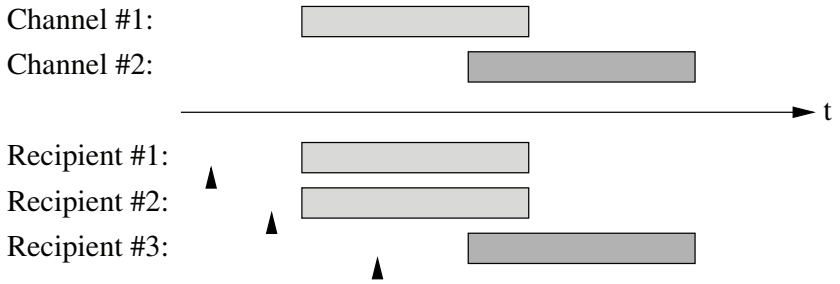


Figure 3.5: Diagram of a Batching transmission.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Single media stream bit rate **Receiver storage size:** No media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** Full dynamic content support **Variable-bit-rate media support:** Full dynamic content support **Live streaming capability:** Full live streaming capable
Comment: ! Playback delay depends on system load	

Table 3.9: Classification for Batching transmissions

3.3.3 Adaptive Piggy-Backing

While Batching benefits from one-to-many transmissions by allowing several recipients to listen to one media stream, the Adaptive Piggy-Backing transmission scheme [GLM95, GLM96, AWY96a] introduced a completely new approach to reduce the bandwidth in media-on-demand transmissions: The basic idea of the Adaptive Piggy-Backing transmission scheme is that a recipient can *benefit from other ongoing transmissions* of the same media stream.

The Adaptive Piggy-Backing transmission scheme accomplishes this by joining a new and an ongoing transmission at some time. To bring two media stream transmissions to the same time of transmission without interrupting the old one or skipping part of the transmission in the new one, the display frame rate of the transmissions is modified at the receiver system for the Adaptive Piggy-Backing transmission scheme. This allows delaying the ahead-running transmission and to speed-up the later transmission until both transmissions reach the same point in the media stream. A bandwidth gain is then achieved because the channel of one of the transmissions can be terminated and used for the next request when the transmissions have been joined.

The authors of the article describing the Adaptive Piggy-Backing transmission scheme also stated that

> *[. . .] the video editing industry have assured [them] that alterations of the actual display rate in the 2-3 % range or expansion and contraction (through interpolation) in the 8 % range can be accomplished without being noticeable to the viewer.*

But even if one media stream is expanded by 8 % and the next following one contracted by 8 %, two requests can only be merged if the time between their arrivals is shorter than 16 % of the media duration. For example for a two hour media stream, the requests can only be merged if the second request arrives within 19.2 minutes after the first one. And the bandwidth reduction is very small, even when the offset of the second request is as near as 9.6 minutes, only the half media stream transmission can be left out.

Another drawback of this approach is that this creates additional requirements to the display system: Assuming a transmission which includes a visual stream, the display system must be able to adapt its frame rate to the display frame rate which has been determined by the Adaptive Piggy-Backing transmission scheme. If this is not supported by the display system, interpolated frames have to be inserted into or surplus frames have to be deleted from the transmitted stream for presentation to adjust the display frame rate. These changes typically cause irritating judders and thereby reduce the media quality significantly during presentation.

In [GLM95, GLM96], four different algorithms for the selection of the streams for which to adjust the display frame rate have been proposed:

- **Odd-Even Reduction Policy:** The media stream for the first request is adjusted to the

lowest possible bit rate, the next request (within a given window) uses the maximum bit rate until the streams merge. If the next request arrives very late (e. g. at a time so it cannot be merged) or if a third request arrives, the algorithm is restarted, serving the request again using the lowest possible bit rate.

- **Simple Merging:** For the first stream, the lowest bit rate is used, for all further streams within a fixed window the maximum bit rate is used.
- **Greedy Policy:** Same as odd-even policy, but after a merging or when reaching the window when no other stream can merge to it, the algorithm is applied again to this stream. This way, merged streams can be merged to other merged ones.
- **Limited Merging:** Merging is restricted to a predefined first part of the media, so the whole media needs not to be available using several bit rates on the sender system.

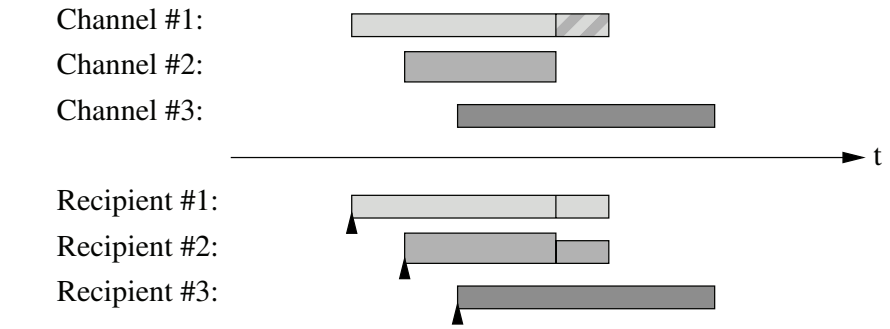
For the Adaptive Piggy-Backing transmission scheme, a new type of diagram is needed as the merging process is not clearly visible in the channel-based type of diagram which has been used to illustrate the previously described transmission schemes. In **stream-oriented diagrams**, each stream transmission is represented by a diagonal line. The bit rate of the transmission is shown by the slope of the line, and a premature termination of a channel can be identified by the fact that the line does not reach the height of the end stream marker at the left, vertical axis. Horizontal, dashed lines are sometimes added at the premature termination of a stream to show which stream is used to continue the playback. (This only applies to transmission schemes which buffer data. For Adaptive Piggy-Backing, the continuation stream meets the terminating stream, so this line is not necessary.)

Figure 3.6 shows a transmission diagram for the Adaptive Piggy-Backing transmission scheme, illustrating the playback of three recipients. The transmission of the second recipient could be merged to the first one in this example.

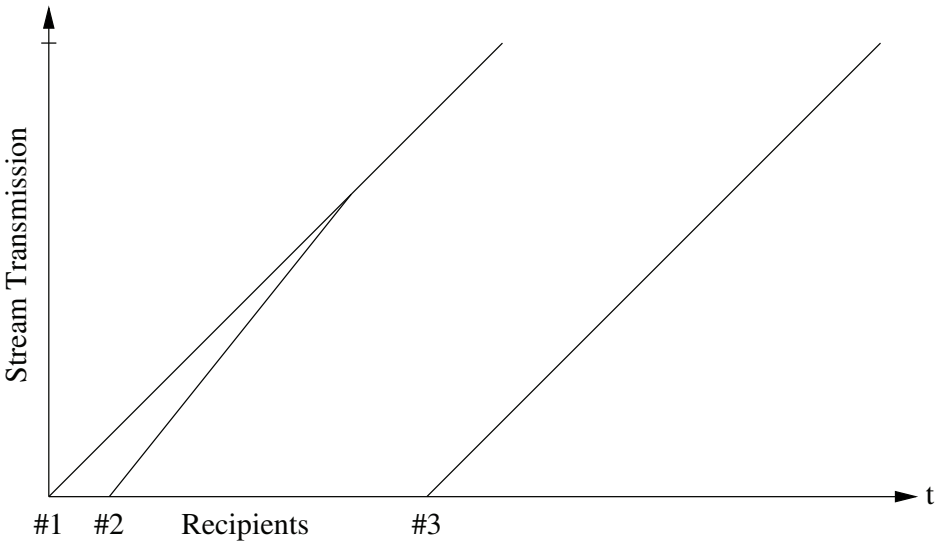
3.3.4 Patching

Patching [HCS98b] uses the same idea as Adaptive Piggy-Backing to reduce the bandwidth of the sender system: Recipients benefit from other ongoing transmissions of the same media stream. But instead of modifying the display frame rate of transmissions, Patching requires that the clients receive data from another transmission concurrently to the reception and playback of the actual transmission.

Therefore, two types of transmissions are possible for the Patching scheme: Firstly, the sender system can transmit a media stream in its whole, as it has been done in the Batching scheme. The first request for a media stream is always served using such a **full transmission**. But secondly, if a request arrives a short time after a full transmission has been started, the sender system can



(a) Adaptive Piggy-Backing in a channel-based diagram



(b) Adaptive Piggy-Backing in a transmission-based diagram

Figure 3.6: Diagram of an Adaptive Piggy-Backing transmission.

decide to send only the "missed" part to the recipient. The recipient has to listen to both the new **patch transmission** and the ongoing full transmission in this case, buffering the data of the full transmission and "patching" the parts together when playing.

While full transmissions must be sent using one-to-many transmissions, the patch transmission can use either one-to-one transmissions (if the recipient request is to be served immediate), or one-to-many transmissions (if several recipients are served with the patch transmission, e. g. if Batching is used to collect several requests).

As already mentioned above, the Patching scheme introduces two new requirements to the receiver system compared to the Batching scheme:

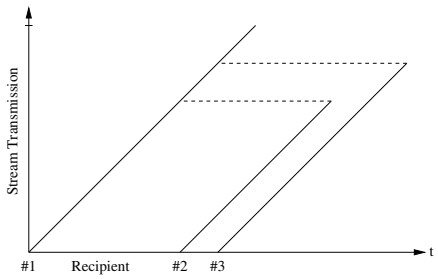
- The receiver system must be able two receive two transmissions at the same time, so the bandwidth requirement of the receiver system is twice the (maximum) media stream bit rate.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Single media stream bit rate **Receiver storage size:** No media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Full live streaming capable
Comment: ! Display frame rate depends on merging strategies	

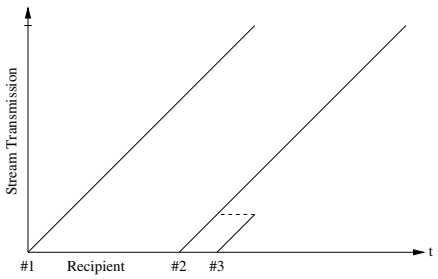
Table 3.10: Classification for Adaptive Piggy-Backing transmissions

- The receiver system must have a buffer where the data of the full transmission is stored while the patch stream is played.

The efficiency of the Patching scheme depends very much on the decision when a full transmission is to be started for an incoming request and when a patch stream is used instead. Unfortunately, using patch transmissions whenever possible does not always result in the lowest bandwidth consumption as illustrated in figure 3.7: In figure 3.7a, both transmissions #2 and #3 merge to transmission #1, resulting in a total of 2.5 media stream transmissions, whereas in figure 3.7b only transmission #3 merges to the previous full transmission #2 and the needed bandwidth totals to $\frac{13}{6} \approx 2.17$ media stream transmissions.



(a) Both streams #2 and #3 patch to the full transmission #1



(b) Only the last stream #3 patch to the preceding full transmission #2

Figure 3.7: Two different ways for the transmission of three media streams using patching.

Another problem arises if the buffer size of the recipient system is not large enough to hold half of the media stream. (It is not possible that the receiver system has to buffer more than half of the media stream as only two transmissions are received and one of them is played immediately.) In [HCS98b, CHV99], some algorithms for the merge decision have been proposed for this case:

- **Greedy Patching**: If the client buffer is too small for the missed part, the last part of the full transmission is buffered and the patch has to transmit the remainder.
- **Grace Patching**: If the client buffer is too small for the missed part, a new full transmission is scheduled.
- **Patching with Fixed Window**: If the client request arrives within a fixed window after a full transmission, only the patch is transmitted, otherwise a new full transmission is started. For this algorithm, the client buffer must be large enough to store a part of the media stream of the size of the fixed window.
- **Optimal Patching** [CHV99, GT01, GT99]: Same as patching with fixed window, but the window size is calculated using the request arrival rate of the recipients. Assuming a Poisson distribution of the client requests, this algorithm calculates the window size so that the average bandwidth is minimized.

Figure 3.8 shows the four different approaches how a receiver request is handled if it arrives so late that the missed part cannot be cached by the client. To emphasize the differences, a very small buffer has been chosen.

The authors of [BNL04] provided an algorithm of complexity $O(n^2)$ for calculating the optimal merge decisions for a set of request times. Although this algorithm cannot be used in a practical environment (as the request times are normally not known in advance), it is possible to use this algorithm to examine the theoretical limit which can be reached by the Patching transmission scheme at all.

3.3.5 Tapping

Tapping [CL97, SGRT99] can be seen as an improved version of Patching[4]. While the Patching scheme only differentiated two types of transmissions — full transmissions and patch transmissions — the Tapping transmission scheme uses three classes: **original streams** for full transmissions, **full tap streams** for patch transmissions and **partial tap streams** as new class of transmission.

Partial tap streams are used when a stream should be merged to another transmission but the buffer of the receiver system is too small to save the missed part of the transmission. For partial

[4]Strictly speaking, Tapping has been published before Patching, so Patching has to be looked at as a simplified version of Tapping. But for better understanding, Patching has been presented before Tapping in this thesis.

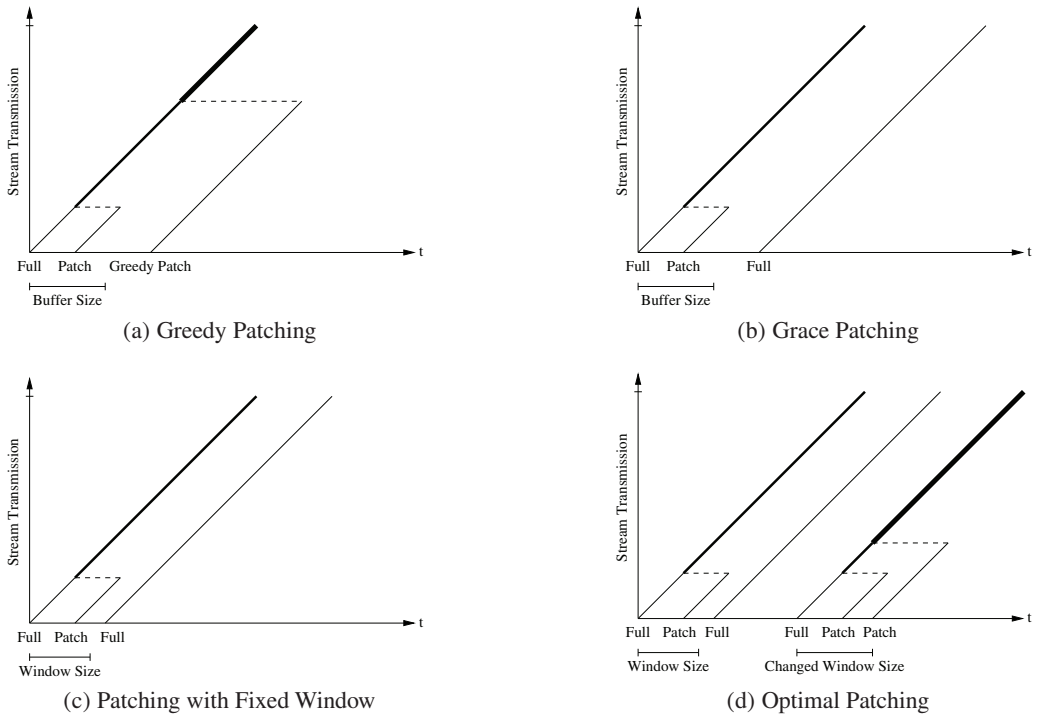


Figure 3.8: Four different Patch transmission strategies.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** **!** Twice the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Full live streaming capable

Table 3.11: Classification for Patching transmissions

tap streams, the buffer of the receiver system is first filled with the data of the ongoing original transmission until it is full. When the data of the tap stream reaches the buffered position, the transmission on the tap stream can be interrupted and the receiver system can play the buffered data. While playing data from the buffer and therefore emptying it, the receiver system again

listens for new data from the original transmission and saves it into the buffer. When the data of the buffer has been played, the stream is again continued at the tap stream, until the position in the buffer is reached and the procedure is repeated. Figure 3.9 shows this in an example.

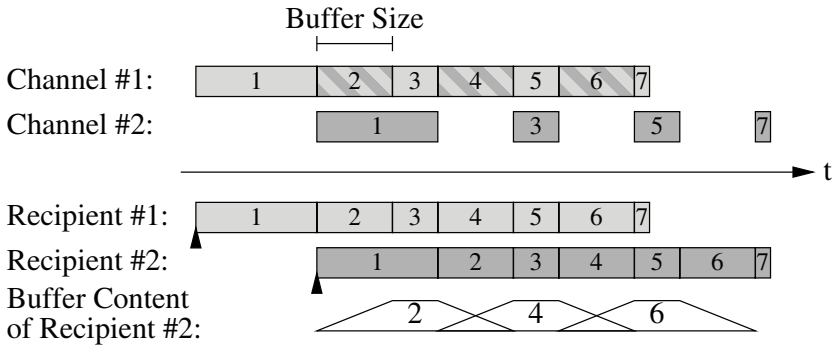


Figure 3.9: Example of a full transmission and a partial tap stream transmission together with the buffering

Figure 3.10 shows a (greedy) Patching transmission compared to an original Tapping transmission, a Tapping transmission using extra tapping and a Tapping transmission using stream stacking.

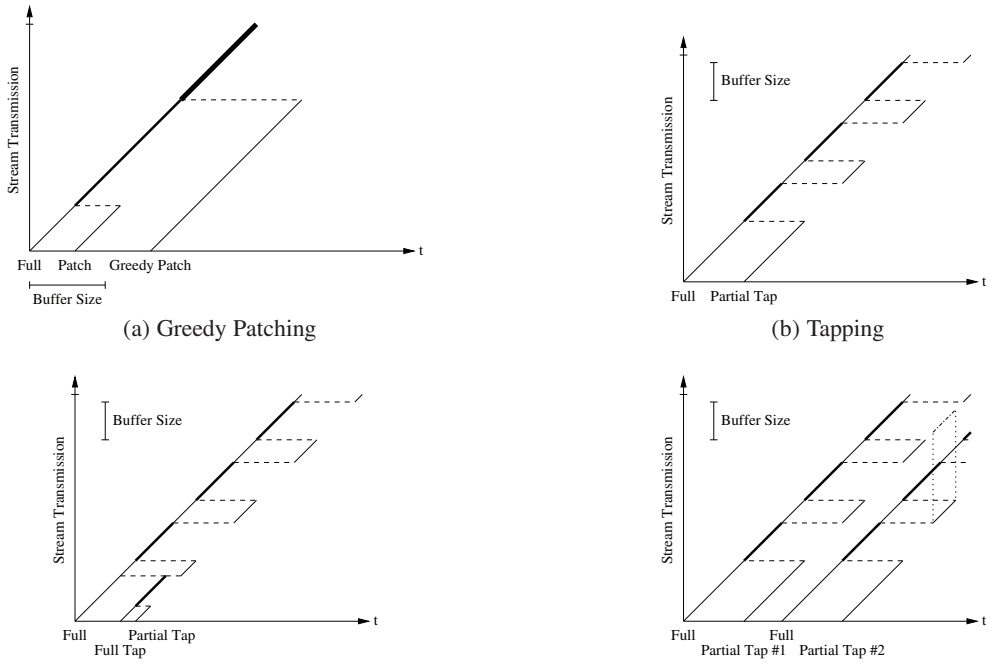


Figure 3.10: Comparison of Patching and Tapping.

The idea of Tapping has also been published from other authors under the terms **Generalized**

Buffer Reuse Patching or **Optimized Patching** [SGRT99]. Additionally, in [Pâr01c], the authors improved the performance of Tapping by preloading a part of the media stream to the receiver system some time before the stream is requested. The effects of partial preloading are examined in detail in section 4.6.

If the receiver system can receive more than two streams at once, extended versions of Tapping [CL99, CLP00] can be used:

- **Extra tapping**: The receiver system can tap to any ongoing stream transmission, not only to the original transmission.
- **Stream stacking**: When the receiver system has space left in its buffer and another channel is not used at the moment (e. g. a partial tap stream), the order of the transmissions may be rearranged, using the suspended other channel for the transmission. This does not reduce the total bandwidth to transmit but allows releasing channels earlier.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** **!** Twice the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Full live streaming capable

Table 3.12: Classification for Tapping transmissions

3.3.6 Merging

Merging [EVZ99b, EVZ99a, EVZ01] is the most generalized version of Patching and Tapping: The Merging scheme allows receiver systems to be merged to any stream, not only to full transmissions.

This improves the performance, but at the same time it increases the complexity at the sender side: The sender system must not only decide if a full or patch transmission should be used for a request, all merge decisions have to be evaluated.

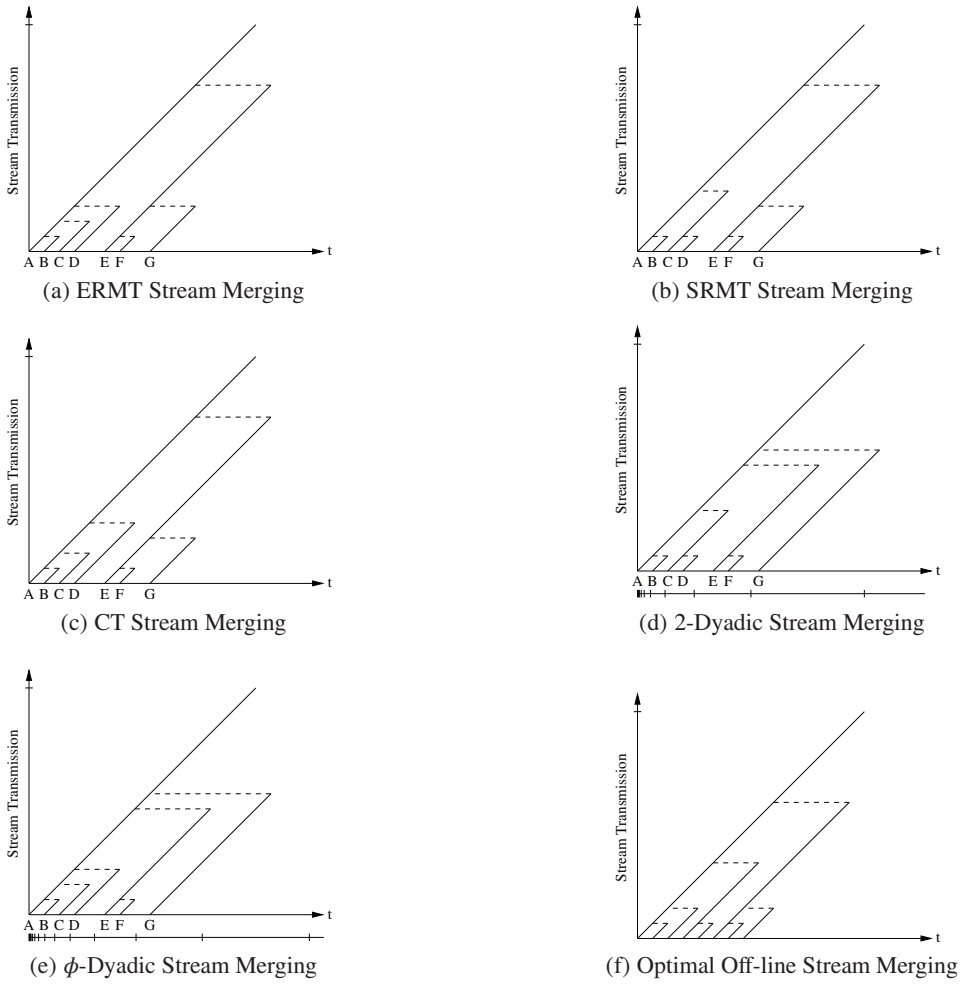
Merging is a very efficient transmission scheme for media-on-demand, and several strategies which streams to merge with which other stream have been proposed:

- **Earliest Reachable Merge Target (ERMT)** [EVZ99b, EVZ99c]: For the Earliest Reachable Merge Target strategy, the first possible merge target is selected when the merging process of all (yet) existing streams is simulated.
- **Simple Reachable Merge Target (SRMT)** [EVZ99b, EVZ99c]: As the decision for Earliest Reachable Merge Target is very complicated to calculate in large systems, this strategy uses a more simple one: A merge target is only selected if the merge time of the merge target with its immediately preceding stream lies behind the start time of the examined stream.
- **Closest Target (CT)** [EVZ99b, EVZ99c]: Another simplification of the Earliest Reachable Merge Target strategy is to always merge to the closest preceding stream which still exists and restart the search when the chosen stream terminates as it merged to another stream.
- **Dyadic Stream Merging (2-Dyadic)** [CJM02]: For the Dyadic Stream Merging strategy, the interval of a full playback is divided into dyadic intervals of parameter 2 (i. e. into intervals which are obtained by splitting the interval into two halves and recursively continuing splitting the first half interval again and again). The first request of the second half always starts a new full transmission in this scheme, therefore no requests of the second half have to be considered here. For requests of the first half, the first request in each of the dyadic intervals is merged directly to the full transmission. For the remaining request, the intervals, starting at these requests until the end of their dyadic interval are again split into dyadic intervals and requests in them are merged in the same manner, continuing until all requests have been processed.
- **Dyadic Stream Merging with golden ratio (ϕ-Dyadic)** [BNGLT02]: This strategy uses the same merging algorithm as the Dyadic Stream Merging strategy, but instead of halving intervals, they are split according to the golden ratio $\frac{1+\sqrt{5}}{2}$ [BS91]. The authors of the Dyadic Stream Merging strategy have proven that this is the optimal merging strategy for Poisson distributed request arrivals if the requests are not known in advance.
- **Optimal stream forest calculation** [LLG98, BNL04]: For a given list of request times it is possible to calculate the optimal merging combinations in $O(n^2)$ using dynamic programming. This approach cannot be used in real environments as the list of request times is not known in advance.
- **Optimal off-line merging trees** [BNGL03]: For a pro-active transmission scheme (sometimes called off-line transmission scheme), it is possible to compute merging trees in advance. The optimal merging trees in this case are Fibonacci trees, i. e. trees which are

defined by a recursive, Fibonacci-like function:

$$S(i) = \begin{cases} \text{the single node } i & \text{if } i \leq 2 \\ \text{the tree obtained by adding } S(i-2) \text{ as} \\ \text{last child to the root node of } S(i-1) & \text{if } i > 3 \end{cases} \tag{3.17}$$

Figure 3.11 shows an example how one request pattern is handled differently by the ERMT, SRMT, CT, 2-Dyadic, ϕ-Dyadic and optimal off-line merging strategies.



(a) ERMT Stream Merging

(b) SRMT Stream Merging

(c) CT Stream Merging

(d) 2-Dyadic Stream Merging

(e) ϕ-Dyadic Stream Merging

(f) Optimal Off-line Stream Merging

Figure 3.11: Comparison of different Stream Merging strategies.

3.3.7 Virtual Batching

Virtual Batching (sometimes also referred to as Chaining) [SHH97, SHT97] differs from the above transmission schemes by requiring that each receiver system can send a media stream to another receiver system. Virtual Batching uses this capability to create chains of receivers, where each re-

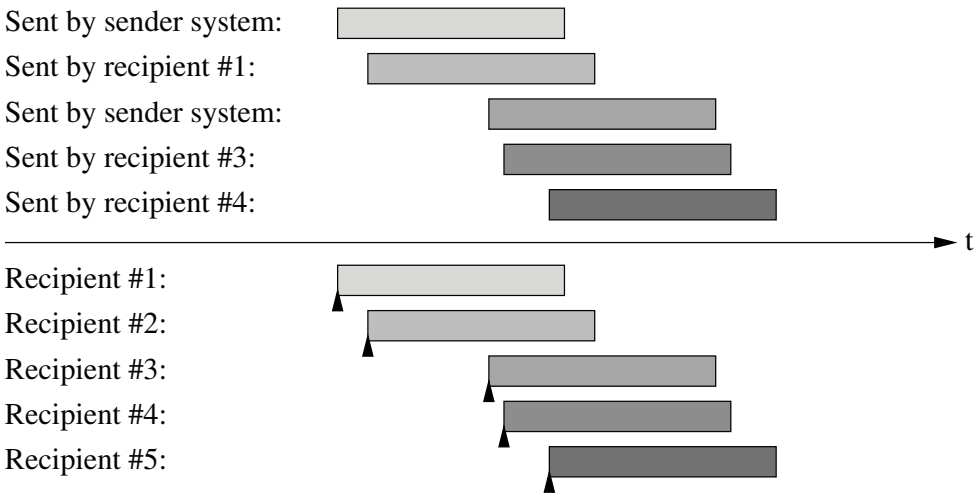
Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Full live streaming capable

Table 3.13: Classification for Merging transmissions

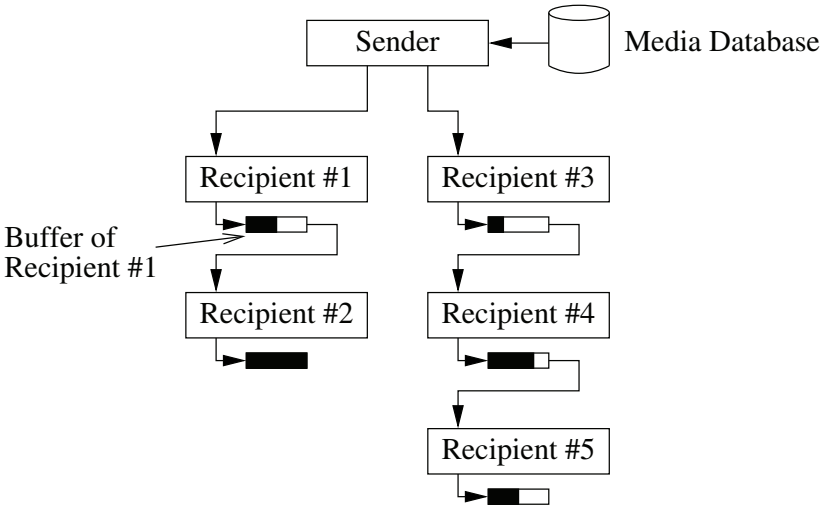
ceiver passes the media stream data to another receiver with a delay (which is realized by buffering the data before passing it). Whenever a new recipient requests the media stream before the buffer of the last recipient in the chain was exhausted, the new recipient can simply be added to the end of the chain, otherwise the receiver system becomes the first system in a new chain. Figure 3.12 illustrates this with five receiver systems in two chains, showing the used buffer size of the receiver systems as bars below them.

Using receiver systems as further senders reduces the bandwidth requirements of the sender system a lot, but induces several problems:

- At first, the receiver systems must be able to send the media stream to another receiver system. This is probably the biggest problem as the upstream bandwidth of the receiver system often does not provide enough bandwidth (if available anyhow).
- The receiver system must provide means for resource reservations: It has to provide buffer space for delaying the media stream in the chain and it has to reserve bandwidth for sending the media stream to the next system in the chain.
- The media transmission system is more error-prone as it requires proper working of receiver systems. For example, switching a receiver system off or a power failure at one of the receiver system breaks the chain and causes an abortion of the stream transmission at all systems in the tail of the chain.
- All receiver systems in the chain must be trusted that they send the data unmodified. Probably, the media stream has to be signed using an asymmetrical algorithm, so the recipient systems can check if the media stream has not been altered.



(a) Virtual Batching Transmissions



(b) Chains in Virtual Batching Transmission

Figure 3.12: Example of a Virtual Batching transmission

- It may be undesirable that a receiver system can detect its successor system in the chain for privacy reasons.

3.3.8 Peer-to-Peer Networks

A generalization of Virtual Batching is provided by Peer-to-Peer Networks: In a Peer-to-Peer Network, the sender and receiver roles are merged to a single peer role, so every node can consume data and provide data at the same time.

Similar to some advanced media-on-demand transmission schemes, today's Peer-to-Peer Networks split files into several small parts which are retrieved separately and finally concatenated. In

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required, supporting media bit rate to another recipient **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** ! Single media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Full live streaming capable
Comment: ! Receiver systems must forward media streams	

Table 3.14: Classification for Virtual Batching transmissions

contrast to advanced media-on-demand transmission schemes where the splitting of media streams into segments is used to lower the total bandwidth consumption for a media stream transmission to several recipients, the reason for splitting them in Peer-to-Peer Networks only serves the purpose to lower the send bandwidth requirements of the peers[5]. The total bandwidth consumption for media stream transmissions using Peer-to-Peer Networks will still be the same as for Point-to-Point Transmissions because no multicasting transmissions are used.

Similar to Virtual Batching, Peer-to-Peer Networks have several drawbacks if used for media-on-demand (i. e. if the media stream is transmitted on request and consumed concurrently to its reception):

- The peer must be able to send the media stream to another peer. In contrast to Virtual Batching, the peer must not support sending the media stream at full media bit rate if several peers send the media stream together to a receiving peer.
- The peer must provide means for resource reservations: As for Virtual Batching, buffer space and bandwidth for sending media stream segments must be reserved.
- The Peer-to-Peer Network will fail to work if the total media bandwidth of all active requests exceeds the total transmission bandwidth of all peers. As peers in Peer-to-Peer Networks have often only a low upstream bandwidth (e. g. using ADSL), the upstream bandwidth of

[5]The peers of Peer-to-Peer Networks have often only a low upstream bandwidth (e. g. using ADSL), so several peers are combined in a Peer-to-Peer Network to transmit the file in an acceptable time.

several peers is required for the transmission of one media stream to a requesting peer. This disallows using Peer-to-Peer Networks for media-on-demand when it is possible that a high ratio of the peers is requesting a media stream concurrently.

- The peer must provide real-time semantics so each segment is available in due time when needed for playback. Peer-to-peer software which is currently used for file distribution does not provide any timing constraints for transmissions. This makes it currently unusable for real-time transmissions like media-on-demand stream transmissions.
- The last three bullets of the list of drawbacks of Virtual Batching (more error-prone, security considerations, privacy concerns) apply to Peer-to-Peer Networks, too.

Technical classification	**Functional classification**
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Mesh **Sender bandwidth:** Any **Sender storage seeks:** One position per channel **Receiver bandwidth:** Single media stream bit rate **Receiver storage size:** Full media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Reactive media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: **!** Receiver systems must forward partial media streams	

Table 3.15: Classification for Peer-to-Peer transmissions

3.3.9 Comparison of Reactive Transmission Schemes

A comparison of the above reactive transmission schemes is very difficult as the schemes differ in many parameters, e. g.:

- average needed bandwidth of the sender system,
- maximum of needed bandwidth of the sender system,
- disposable bandwidth of the sender system,
- maximum bandwidth capacity of the receiver system,

- maximum storage requirements of the receiver system,
- playback request pattern,
- average playback delay,
- maximum playback delay,

To get an overview of the efficiency, the following assumptions are made for the forthcoming comparison:

- Playback requests are Poisson distributed with parameter λ.
- Receiver systems have enough storage to buffer any necessary amount of media data.
- The transmission of a single media stream is examined.

The first of these assumptions can be accepted as long as requests from different receiver systems are independent from each other (see section 3.1.1). The second assumption can be validated by observing the steadily falling costs of memory and hard disk storage: Memory is nowadays cheaper than EUR 20 per gigabyte, and hard disks are available for EUR 0.20 per gigabyte, making tight restrictions on storage requirements nearly obsolete. The restriction to a single media stream is valid as long as all media transmissions are independent from each other.

Using these assumptions, the following schemes can be omitted in the comparison:

- FCFS Batching, MQL Batching and MFQL Batching only differ in the way how they handle requests for different media streams. Because of restriction three, these Batching schemes give equal results in this comparison.
- Prioritized Queuing Batching assumes different classes of recipients. This favors some requests by penalizing other ones, but the overall performance is lowered.
- The Adaptive Piggy-Backing transmission scheme modifies the display rate of the media stream to accomplish merges. As this violates the real-time constraint of media-on-demand transmission schemes (see section 1.1), the Adaptive Piggy-Backing scheme is excluded here.
- The different Patching schemes assume limited receiver storage. Due to restriction two, only Patching with sufficient large storage space and an optimal patching window (calculated according to [EVZ99a, EVZ01] from the average request rate) is used.
- The Tapping schemes are equivalent to Patching if no limit on the storage size of the recipient systems is imposed.
- Virtual Batching and Peer-to-Peer Networks are omitted as they use recipient systems as additional senders.

To compare the remaining transmission schemes, they are evaluated using simulated, Poisson distributed client requests of different request rates. Each simulation is run 50 times and the observed interval is chosen each time to contain 5 000 requests to get meaningful results.

For transmission schemes which provide immediate service, the efficiency of the transmission scheme is then retrieved by calculating the minimum required bandwidth using the formula given in section 3.1.2 and dividing this value by the average bandwidth of the simulations as described in section 3.1. Figure 3.13 shows the results for request rates between 1 and 1 000 requests per media stream duration.

The results of the Optimal Merge Decision Patching algorithm (see section 3.3.4) have been included in this graph although it requires knowledge of the request times in advance. As this knowledge gives the algorithm an advantage to transmission schemes which have to react on random request patterns, it has a higher efficiency. For low request rates, its "efficiency" is even higher than 100 % as the efficiency calculation assumes a random Poisson process for the distribution of requests without knowledge of future requests.

For the Point-to-Point transmission scheme, a receiver bandwidth equal to the media bit rate has to be asserted. Under this precondition, the Point-to-Point transmission scheme has an efficiency of 100 % as it is the optimal (and apart from channel permutations the only) solution which provides immediate service if the receiver systems cannot receive more than the media bit rate. To prove that the Point-to-Point transmission scheme looses its attractiveness if the receiver bandwidth is not limited to the media stream bit rate, the efficiency of the Point-to-Point transmission scheme has been added a second time to the graph for the case that the receiver system is able to receive twice the media bit rate.

Besides of this special case for the Point-to-Point transmission scheme, it is clearly visible that the ERMT Merging algorithm has the highest efficiency of all reactive transmission schemes. Another important point about reactive transmission schemes is that they are less efficient for high media request rates.

For the Batching transmission scheme which serves requests delayed, the bandwidth requirements are measured for a maximum playback delay of $\frac{1}{120}$ of the media stream duration (which corresponds to a playback delay of one minute for a two hour media stream) and compared to the theoretical minimum to calculate the efficiency of this scheme. Figure 3.14 shows the result of this analysis. As for the Point-to-Point transmission scheme, the Batching transmission scheme requires only a receiver bandwidth equal to the media bit rate and has an efficiency of nearly 100 % under this precondition. As above, the efficiency of the Batching scheme for a receiver bandwidth of twice the media bit rate has been added to the graph to make it comparable with other transmission schemes if this precondition is not necessary.

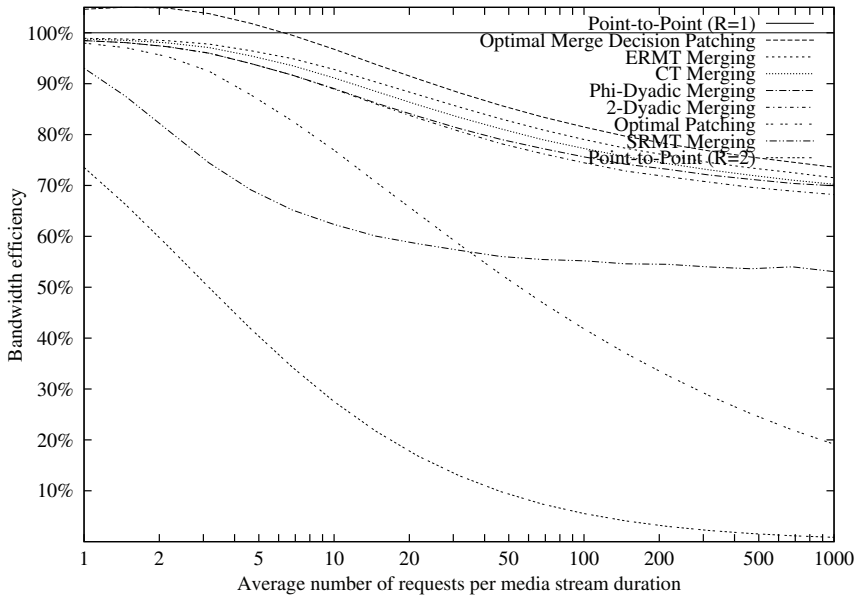


Figure 3.13: Comparison of efficiency of reactive transmission schemes with immediate service

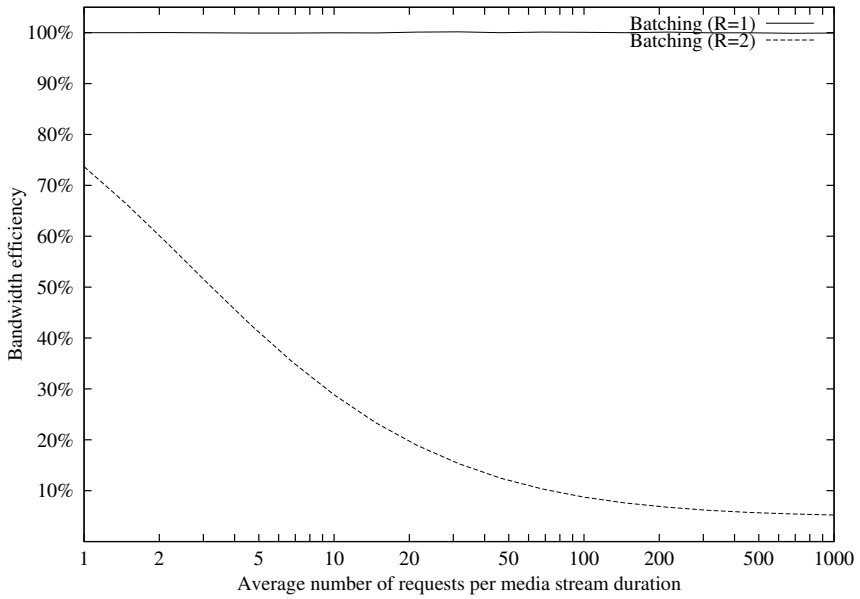


Figure 3.14: Efficiency of reactive transmission schemes with delayed service

3.4 Non-Segmenting Pro-Active Transmission Schemes

In the last section, reactive transmission schemes have been examined. Starting with this section and extending up to section 3.7, the large group of pro-active transmission schemes is examined.

In contrast to reactive schemes which transfer media streams (immediately or with some delay) as a result of individual request from consumers, pro-active transmission schemes transmit media streams continuously, independent of any requests. This has several advantages over reactive schemes:

- Playback requests must not be forwarded from the receiver system to the sender system, it is actually possible to drop any communication from the receiver system towards the sender system.
- Resource requirements (bandwidth, memory) and service guarantees (playback delay) can be calculated independent of the number of concurrent transmissions and the pattern of client requests.
- The needed server bandwidth for highly requested media streams is lower if a pro-active scheme is used.
- The transmission schedule has not to be created dynamically but can be computed in advance which lowers the real-time requirements of the sender system and removes some complexity.

The main disadvantage of a pro-active transmission scheme is that it cannot save bandwidth when only few or even no recipients listen to the transmission. Solutions to this problem are addressed in more detail in sections 3.8 and 4.20.1.

The remainder of this section describes two very old and simple pro-active transmission schemes. These transmission schemes always transmit media streams as a whole and are generally only interesting because of their simplicity. Several more advanced schemes are presented in the sections after this one in three groups, sorted by their functioning.

3.4.1 Round-Robin Broadcasting

A simple type of media-on-demand is to transmit one (or several) media streams per transmission channel permanently in a round-robin manner. This type of media-on-demand is very popular in small, ancient media-on-demand systems and can sometimes still be found in hotels. Its main advantages are low system requirements: Besides of accounting, the media-on-demand receiver system has no other requirements than a normal TV system, and at the sender side, usual video cassette recorders with automatic rewind or endless tapes can be used. The main disadvantage of Round-Robin Broadcasting is the high playback delay: In worst case, a consumer has to wait nearly one whole media playback duration until the media stream is started again.

Figure 3.15 shows a transmission diagram for a Round-Robin Broadcasting transmission, together with the media playback by one recipient.

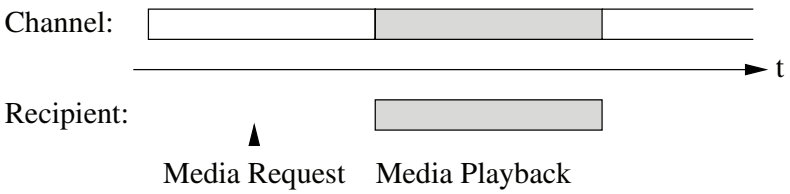


Figure 3.15: Diagram of a Round-Robin Broadcasting transmission.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Single media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Single media stream bit rate **Receiver storage size:** No media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** Full dynamic content support **Variable-bit-rate media support:** Full dynamic content support **Live streaming capability:** Full live streaming capable
Comment: ! Maximum playback delay equals media stream duration	

Table 3.16: Classification for Round-Robin Broadcasting transmissions

3.4.2 Staggered Broadcasting

An improvement to the Round-Robin Broadcasting scheme presented above is the Staggered Broadcasting scheme (also known as Baseline transmissions) [PLE03]: For Staggered Broadcasting transmissions, each media stream is transmitted several times in parallel, using different phase shifts for each transmission. As a result, the playback delay is reduces to a fraction equivalent to the reciprocal of the number of used channels.

Another improvement of this transmission scheme is the possibility to navigate within the media stream: By changing to another transmission of the same media stream, the receiver system can jump within the media stream in a limited way (fixed jump playback control).

Although the improvement over Round-Robin Broadcasting is very high, this transmission scheme requires a lot of sender bandwidth, so the number of transmitted media streams is typically low.

Figure 3.16 shows a transmission diagram for a transmission using three channels for the Staggered Broadcasting scheme, together with the playback by one recipient.

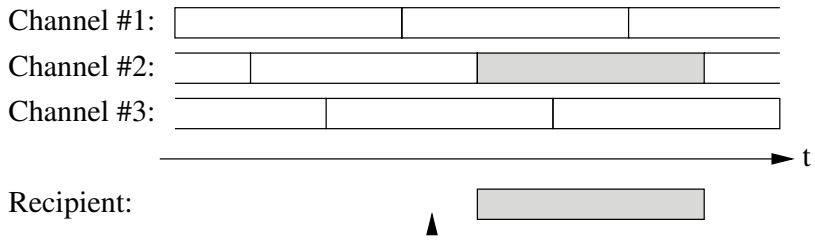


Figure 3.16: Diagram of a staggered transmission.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** ! Single media stream bit rate **Receiver storage size:** No media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** Fixed jump playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Full live streaming capable
Comment: ! High server bandwidth requirements if short playback delays should be provided	

Table 3.17: Classification for Staggered Broadcasting transmissions

3.4.3 Comparison of Non-Segmenting Pro-Active Transmission Schemes

Round-Robin Broadcasting transmissions and the Staggered Broadcasting transmission scheme are two very simple but widely used versions for pro-active media-on-demand. Their main advantages lay in the low requirements for the receiver systems and their simplicity for both sender and receiver systems. But for short playback delays, they are practically of no use:

- For the Round-Robin Broadcasting scheme, no modification of the playback delay is possible, the worst case playback delay is always equal to the media stream duration.

- For the Staggered Broadcasting scheme, the required bandwidth is inversely proportional to the playback delay, i. e. the bandwidth at the sender system increases very fast if the playback delay becomes small.

The efficiency of the Round-Robin Broadcasting scheme is 100 % for the single set of supported parameters (playback delay equals stream duration), but as no modification of the playback delay is supported, the use of this transmission scheme is very limited.

Calculating the efficiency for the Staggered Broadcasting scheme depends very much on the prerequisites: Assuming that the receiver system can only receive one channel at media stream bit rate, it is not possible to create a pro-active transmission scheme with lower bandwidth requirements than the Staggered Broadcasting scheme provides, i. e. the efficiency η is 100 %.

But if the receiver systems can receive more than a single channel at media stream bit rate, the efficiency decreases very fast for short playback delays. For example, if a playback delay of $\frac{1}{120}$ of the media stream duration is desired (which equals a playback delay of one minute for a two hour media stream) and the receiver system is able to receive six times the media bit rate (which is the bandwidth needed by one of the advanced transmission scheme presented later for the same setup), the efficiency of the Staggered Broadcasting scheme is only[6]

$$\eta = \frac{\Psi_0\left(\frac{1+\frac{1}{120}}{\frac{1}{7200}}\right) - \Psi_0\left(\frac{\frac{1}{120}}{\frac{1}{7200}}\right)}{120} = \frac{\Psi_0(7260) - \Psi_0(60)}{120} \approx 4.0\,\% \tag{3.18}$$

3.5 Size-Based Segmenting Transmission Schemes

The last section introduced the idea of pro-active transmission schemes with two very inefficient approaches. Starting with this section and continued up to section 3.7, a group of advanced, much more efficient, pro-active media-on-demand transmission schemes is examined.

One common fact for this group of schemes is that the media stream is not transmitted sequentially any more. Instead, the media stream is split into several segments, which are broadcast according to a fixed or calculated **transmission schedule** by the sender system. On the receiver system side, all received segments have to be saved for these schemes and must be reassembled when they are needed for playback.

The main advantage of this segmentation procedure is that bandwidth can be saved: As receiver systems do not need the later segments of a media stream until all preceding segments have been played, these segments can be transmitted less frequently.

Three different ways how segmented transmissions can be performed have evolved:

[6] This calculation is based on equation (3.10) using a slot interval of $\delta = \frac{d}{7200}$ which equals segments of one second in case of a two hour media stream.

- **Size-based approach:** All segments are transmitted in parallel at the same time using the same bandwidth, but the segment sizes are increasing. This way, the size of a segment determines the frequency for the transmission of the data of the segment, and increasing the size for later segments of a stream therefore reduces the transmission frequency of these segments.
- **Bandwidth-based approach:** The media stream is split into segments of equal size, which are transmitted in parallel, using decreasing bandwidths. Using this approach, the bandwidth determines the frequency for the segment transmissions. As the bandwidth for later segments of a media stream is decreased, their transmission frequency is lowered.
- **Frequency-based approach:** The media stream is split into segments of equal size, which are transmitted using the same bandwidth for all segments but only at increasing periodic intervals. As the segment period determines the frequency for the segment transmission in this approach, the frequency for the transmission of later segments of a media stream is decreased by increasing the periods of these segments.

Figure 3.17 illustrates the the differences between these three approaches.

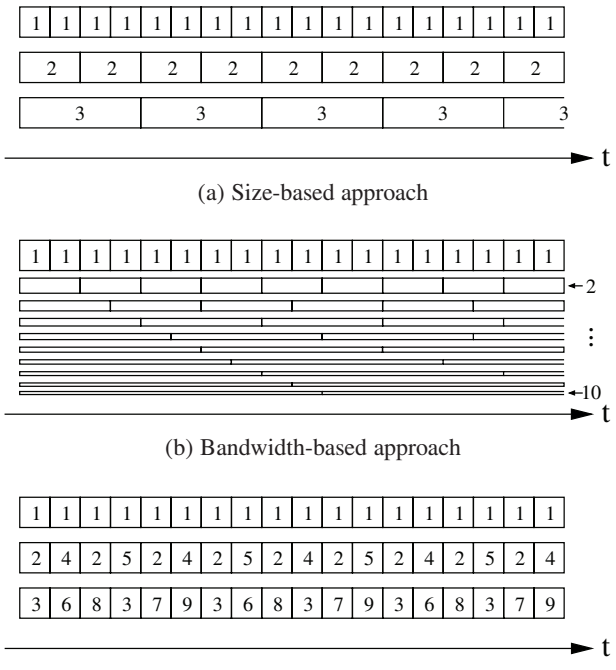


Figure 3.17: Three different approaches for segmented media-on-demand transmissions

The remainder of this section describes transmission schemes which use the size-based approach, while transmission schemes of bandwidth-based and frequency-based approaches are described in the next two sections 3.6 and 3.7, resp.

3.5.1 Pyramid Broadcasting

The Pyramid Broadcasting scheme [VI96] was the first published transmission scheme which used the segmenting approach. Although its origins lay rather in TV network based transmissions, it can be used on any other multicast or broadcast capable network.

For the Pyramid Broadcasting scheme, each media stream is split into segments of geometrically increasing size, i. e. the size of the i-th segment is α^{i-1} times the size of the first segment. The parameter α thereby is chosen by the Pyramid Broadcasting Scheme in such a way that the maximum playback delay is minimized. As the maximum playback delay W^+ can be calculated from α and the number of channels N (which is the same as the number of segments n for the Pyramid Broadcasting scheme) by

$$W^+ = d \cdot \frac{\alpha - 1}{\alpha \cdot (\alpha^N - 1)} \tag{3.19}$$

it is only possible to calculate the optimum values for α by applying a numerical minimum search. Table 3.18 shows the results of this calculation for two to eight channels.

Number of channels	α
2	1.7105
3	2.0556
4	2.2397
5	2.3485
6	2.4183
7	2.4664
8	2.5012
∞	$e \approx 2.7182$

Table 3.18: Optimal values for the parameter α of the Pyramid Broadcasting scheme

For the Pyramid Broadcasting scheme, all media streams are transmitted on the same set of channels, thus the total available bandwidth of the sender system is divided into as many channels as segments are used in each media stream (which assumes that all media streams have the same duration).

To receive a media stream, the receiver system has to join the first channel and has to wait for the next transmission of the first segment of the desired media stream. The segment data is then received into a buffer and the playback is started. While playing the segment, the receiver system joins the next channel, waits for the beginning of the next segment and appends it to the buffer. This procedure is continued until all segments have been received and played.

Figure 3.18 shows a media transmission using the Pyramid Broadcasting scheme for three channels and two media streams (one consisting of segments 1 to 3, the other of segments A to C).

Note that the channel bandwidth is much higher (at least α times the sum of all media stream bit rates) than the media bit rate for the Pyramid Broadcasting scheme to work.

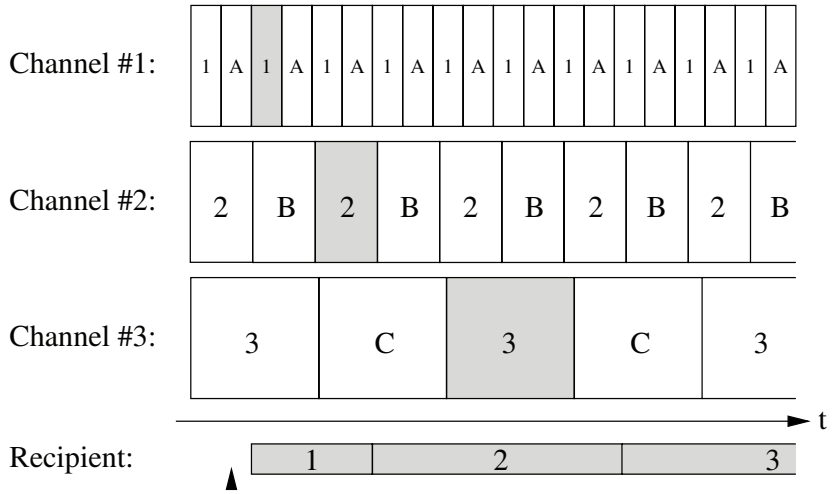


Figure 3.18: Pyramid Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** **!** Multiple of several media stream bit rates **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.19: Classification for Pyramid Broadcasting transmissions

3.5.2 Permutation-Based Pyramid Broadcasting

The Permutation-based Pyramid Broadcasting scheme [AWY96b] is a direct successor of the Pyramid Broadcasting scheme. The goal of the authors of the Permutation-based Pyramid Broadcasting scheme was to reduce the high bandwidth and disk throughput requirements of the Pyramid Broadcasting scheme at the receiver systems.

Its basic architecture is the same as for Pyramid Broadcasting, but instead of transmitting the segments as single pieces at high bandwidth, they are split into subsegments and interleaved with subsegments of other media streams. By this means, the authors of the Permutation-based Pyramid Broadcasting scheme achieved that the receiver systems do not have to receive each segment as a whole any more but only subsegments at periodic intervals.

Another difference between Permutation-based Pyramid Broadcasting and Pyramid Broadcasting are the formulas which are used to calculate the parameter α of the transmission scheme, i. e. the growth factor of the segments. But as the authors of [HS97] have proven, this modification degrades the performance of Permutation-based Pyramid Broadcasting below that of Pyramid Broadcasting.

Figure 3.19 shows a media transmission using the Permutation-based Pyramid Broadcasting scheme for three channels and two media streams (one consisting of segments 1 to 3, the other of segments A to C) where each segment is split into two subsegments (indexed with 1 and 2).

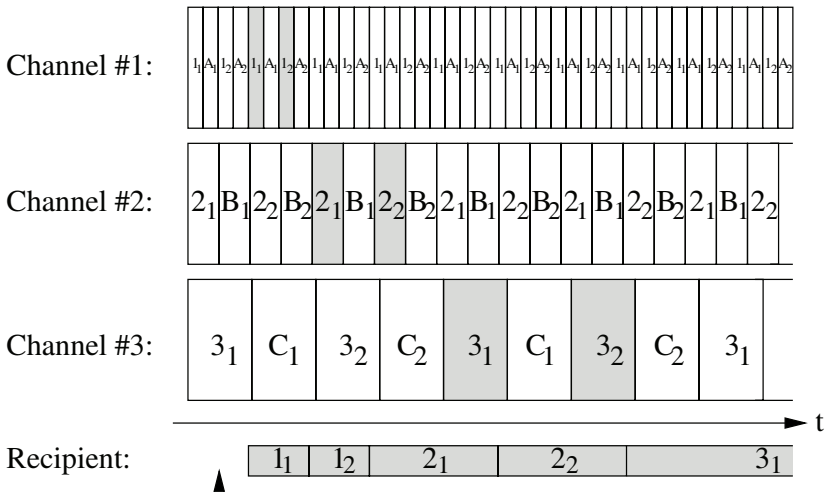


Figure 3.19: Pyramid Broadcasting

3.5.3 Skyscraper Broadcasting

The Skyscraper Broadcasting scheme [HS97, HS00] uses the same size-based approach as Pyramid Broadcasting, but introduces several improvements. Firstly, only one media stream is considered for each transmission, i. e. different media streams are sent independently on different sets of channels. This not only removes some unnecessary complexity from the transmission scheme, it also reduces the bandwidth requirements of the receiver systems.

Another progress is that the authors of the Skyscraper Broadcasting scheme reduced the analysis of size-based transmission schemes to the examination and comparison of different **broadcast-**

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** **!** Multiple of several media stream bit rates **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.20: Classification for Permutation-based Pyramid Broadcasting transmissions

ing series, where each broadcasting series simply describes the (relative) sizes of the segments of the examined transmission scheme. For example, the broadcasting series of Pyramid Broadcasting is the geometric series $1, \alpha, \alpha^2, \alpha^3, \ldots$ because the size of the i-th segment is α^{i-1} times the size of the first segment for this transmission scheme.

For Skyscraper Broadcasting, a new broadcasting series has been introduced which is recursively defined by the following function:

$$S(i) = \begin{cases} 1 & \text{if } i = 1, \\ 2 & \text{if } i = 2 \text{ or } i = 3, \\ 2 \cdot S(i-1) + 1 & \text{if } i \geq 4, i \bmod 4 = 0, \\ S(i-1) & \text{if } i \geq 4, i \bmod 4 = 1, \\ 2 \cdot S(i-1) + 2 & \text{if } i \geq 4, i \bmod 4 = 2, \\ S(i-1) & \text{if } i \geq 4, i \bmod 4 = 3 \end{cases} \tag{3.20}$$

The broadcasting series generated by this function starts with

$$1, 2, 2, 5, 5, 12, 12, 25, 25, 52, 52, 105, 105, 212, 212, 425, \ldots \tag{3.21}$$

and results in shorter playback delays than the geometric series of Pyramid Broadcasting.

Another advantage of Skyscraper Broadcasting is that the broadcasting series can be adjusted to the receiver systems storage size: If an upper bound is applied to the elements of the broadcasting series, the required storage of the receiver system can be limited (at the cost of a longer playback

delay or higher sender bandwidth requirements). In section 4.19, this approach is revived in the context of a frequency-based transmission scheme.

Skyscraper Broadcasting uses channels which have the same bandwidth as the media stream to transmit[7] instead of high bandwidth channels which are required for Pyramid Broadcasting and Permutation-based Pyramid Broadcasting. Of these channels, the receiver systems must be able to receive two channels simultaneously for Skyscraper Broadcasting transmissions, but as the channel bandwidth is much lower than for the Pyramid Broadcasting scheme, the bandwidth requirements have been much relaxed for the receiver systems.

Figure 3.20 shows a media transmission using the Skyscraper Broadcasting scheme for five channels.

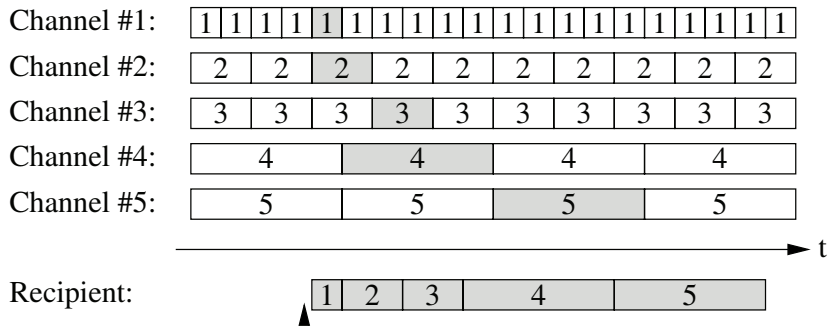


Figure 3.20: Skyscraper Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** ! Twice the media stream bit rate **Receiver storage size:** ! Configurable media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Full live streaming capable

Table 3.21: Classification for Skyscraper Broadcasting transmissions

[7] Skyscraper Broadcasting does not support media streams which use a variable-bit-rate encoding.

3.5.4 Mayan Temple Broadcasting

The Mayan Temple Broadcasting scheme [PLM99] generalizes the Skyscraper Broadcasting scheme in mainly three points: Firstly, it requires that the receiver system receives the data from all channels at once. Although this increases the requirements for the receiver systems, the needed sender bandwidth is reduced this way.

Secondly, it does not use a fixed broadcasting series but a function which calculates the size of the segments from the cumulative bandwidth function of the media stream[8]. The major advantage of this approach is that the Mayan-Temple Broadcasting scheme can support variable-bit-rate media streams. As a consequence, the channel bandwidth can be selected independently from the media stream bit rate, even for constant-bit-rate media streams.

Lastly, the Mayan Temple Broadcasting scheme supports a mechanism called **partial preloading**. Partial preloading means that a part (sensibly the beginning) of the media stream is transmitted to the recipient system before the client requests the media stream. By partial preloading, the Mayan Temple Broadcasting scheme provides support for immediate playback. The disadvantage of partial preloading is, that the beginnings of all media streams have to be transmitted periodically on an additional channel and have to be saved by the recipient systems for later playback. The advantages and drawbacks of partial preloading will be reviewed in section 4.6.

The formula for the duration δ_i of segment $i \geq 2$ of the Mayan Temple Broadcasting scheme is given as:

$$\delta_i = f^{-1}\left(f\left(W^+ + \sum_{j=2}^{i-1}\delta_j\right) + \beta \cdot \left(W^+ + \sum_{j=2}^{i-1}\delta_j\right)\right) - W^+ + \sum_{j=2}^{i-1}\delta_j \tag{3.22}$$

where f is the cumulative bandwidth function (see section 4.1.2), f^{-1} is the inverse cumulative bandwidth function (see section 4.7.3) and β is the channel bandwidth. This formula can be reduced to

$$\delta_i = W^+ \cdot \sum_{j=1}^{i-1}\left(\frac{\beta}{\beta_1}\right)^j \cdot \binom{i-2}{j-1} \tag{3.23}$$

if all segments have the same average bit rate β_1 (which is esp. true for constant-bit-rate media streams)[9].

For a constant-bit-rate media stream and channels at media stream bit rate, this formula generates the following broadcasting series:

$$(1), 1, 2, 4, 8, 16, 32, 64, 128, 256, 512, 1\,024, 2\,048, 4\,096, 8\,192, 16\,384, \ldots \tag{3.24}$$

[8]The cumulative bandwidth function describes the characteristics of the bit rate variations of a variable-bit-rate media stream. A definition of the cumulative bandwidth function is presented in section 4.1.2.

[9]Please take care when reading the article about the Mayan Temple Broadcasting scheme [PLM99] as in their formula the binomial coefficient is accidentally missing.

The first segment from this series is typically transmitted in advance (i. e. preloaded) for this transmission scheme (denoted by the parenthesis in the broadcasting series) but may also be transmitted on a separate channel if preloading is not used.

Figure 3.21 shows a media transmission using the Mayan Temple Broadcasting scheme for four channels and a preloading of segment #1.

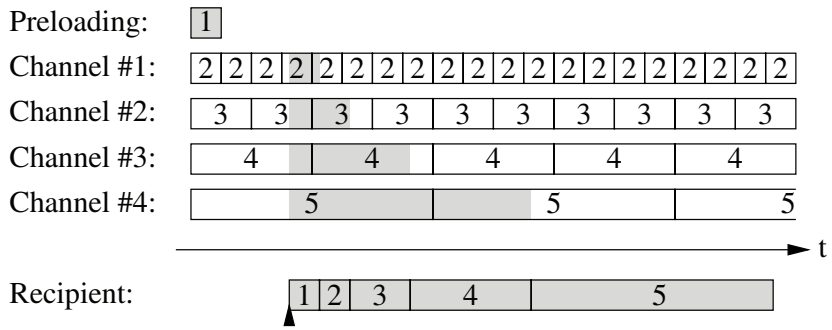


Figure 3.21: Mayan Temple Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** ! Configurable partial preloading **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Full live streaming capable

Table 3.22: Classification for Mayan Temple Broadcasting transmissions

3.5.5 Client-Centric Approach

The Client-Centric Approach transmission scheme [HCS98a] has its origins at the Skyscraper Broadcasting scheme, but it allows specifying the number of channels R which a receiver system can join simultaneously. It uses another broadcasting series, which is defined by the following

function[10]:

$$S_R(i) = \begin{cases} 1 & \text{if } i = 1, \\ 2 \cdot S(i-1) & \text{if } i \geq 2, i \bmod R \neq 1, \\ S(i-1) & \text{if } i \geq 2, i \bmod R = 1 \end{cases} \\ = 2^{i - \lceil \frac{i}{R} \rceil} \tag{3.25}$$

For $R = 2, \ldots, 5$, the broadcasting series of this scheme are

$$\begin{aligned} R = 2: &\quad 1, 2, 2, 4, 4, 8, 8, 16, 16, 32, 32, 64, 64, 128, 128, 256, \ldots \\ R = 3: &\quad 1, 2, 4, 4, 8, 16, 16, 32, 64, 64, 128, 256, 256, 512, 1\,024, 1\,024, \ldots \\ R = 4: &\quad 1, 2, 4, 8, 8, 16, 32, 64, 64, 128, 256, 512, 512, 1\,024, 2\,048, 4\,096, \ldots \\ R = 5: &\quad 1, 2, 4, 8, 16, 16, 32, 64, 128, 256, 256, 512, 1\,024, 2\,048, 4\,096, 4\,096, \ldots \end{aligned} \tag{3.26}$$

Figure 3.22 shows an example for $R = 2$ and three recipients.

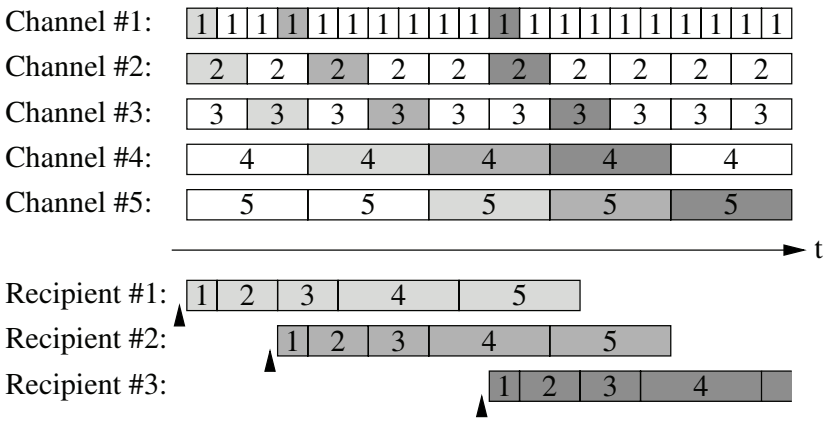


Figure 3.22: Client-Centric Approach

3.5.6 Greedy Disk-Conserving Broadcasting

The Greedy Disk-Conserving Broadcasting scheme [GKT02] is another successor of the Skyscraper Broadcasting scheme. Similar to the Client-Centric Approach, it allows creating a broadcasting series for receiver systems which can receive a specific number of R channels simul-

[10]The recursive version of the formula in [HCS98a] is unfortunately wrong, but the closed form is correct.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** ! Configurable multiple of the media stream bit rate **Receiver storage size:** ! Configurable media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Full live streaming capable

Table 3.23: Classification for Client-Centric Approach transmissions

taneously. The used broadcasting series is defined by the following function for $R = 2$:

$$S_2(i) = \begin{cases} 2^{i-1} & \text{if } i \leq 3, \\ 4 & \text{if } i = 4, \\ 10 & \text{if } i = 5 \text{ or } i = 6, \\ 24 & \text{if } i = 7 \text{ or } i = 8, \\ 5 \cdot S_2(i-4) & \text{if } i \geq 9 \end{cases} \tag{3.27}$$

and the following one for $R \geq 3$:

$$S_R(i) = \begin{cases} 2^{i-1} & \text{if } i \leq R+1, \\ \left\lfloor \dfrac{\sum_{k=i-R}^{i-1} S_R(k)}{S_R(i-R)} \right\rfloor \cdot S_R(i-R) & \text{if } i > R+1 \end{cases} \tag{3.28}$$

For $R = 2, \ldots, 5$, this gives the following broadcasting series:

$$\begin{aligned} R = 2: &\quad 1, 2, 4, 4, 10, 10, 24, 24, 50, 50, 120, 120, 250, 250, 600, 600, \ldots \\ R = 3: &\quad 1, 2, 4, 8, 14, 24, 40, 70, 120, 200, 350, 600, 1\,000, 1\,750, 3\,000, 5\,000, \ldots \\ R = 4: &\quad 1, 2, 4, 8, 16, 30, 56, 104, 192, 360, 672, 1\,248, 2\,304, 4\,320, 8\,064, 14\,976, \ldots \\ R = 5: &\quad 1, 2, 4, 8, 16, 32, 62, 120, 232, 448, 864, 1\,674, 3\,240, 6\,264, 12\,096, 23\,328, \ldots \end{aligned} \tag{3.29}$$

Figure 3.23 shows an example for $R = 2$ and two recipients.

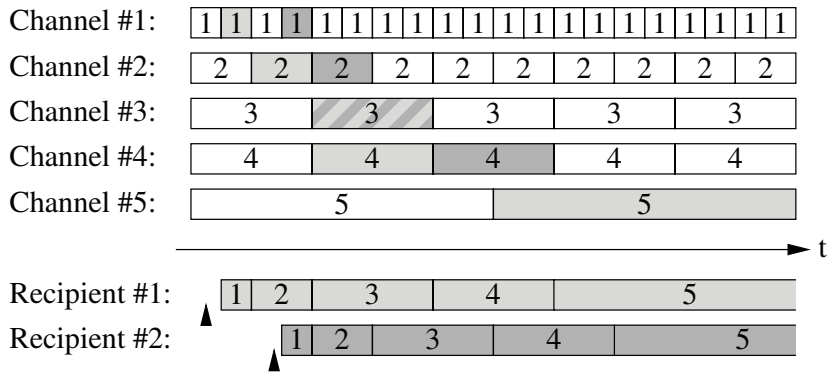


Figure 3.23: Greedy Disk-Conserving Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** ! Configurable multiple of the media stream bit rate **Receiver storage size:** ! Configurable media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Full live streaming capable

Table 3.24: Classification for Greedy Disk-Conserving Broadcasting transmissions

3.5.7 Greedy Equal Bandwidth Broadcasting

With the Greedy Equal Bandwidth Broadcasting scheme [HNvB99], the authors wanted to get the most efficient transmission scheme of the size-based, segmenting group. Therefore, the authors of this scheme proved that their scheme uses the best bandwidth to channel division (which is an equal bandwidth division) and best segment sizes for the size-based class of broadcasting schemes. To achieve this maximum of efficiency, the Greedy Equal Bandwidth Broadcasting requires that the receiver systems can join all used transmission channels at once.

One major advantage of the Greedy Equal Bandwidth Broadcasting scheme to the previously

presented schemes is that the playback delay can be specified independently of the segment sizes. This way, the number of segments can be increased without changing the playback delay, only the channel bandwidth, the number of channels and the size of the segments change when the number of segments is modified. This idea is reviewed again in section 4.5.

Another change to the above described transmission schemes is the starting point of data reception: Instead of waiting for the beginning of the next segment after joining a channel, the receiver systems have to receive the media stream transmission immediately when joining the channel. By starting the reception immediately (an idea which has been proposed earlier for the Polyharmonic Broadcasting scheme which is presented in section 3.6.4), the broadcasting scheme becomes more efficient because segments can be played at the receiver systems immediately after their transmission period has elapsed, even if the channel bandwidth is lower than the media bit rate. (If the receiver system awaits the beginning of the segment instead, the segment cannot be played if it is received concurrently to its playback and if the channel bandwidth is lower than the media bit rate because the receiver system would run out of data.) Again, this idea is analyzed in more detail later in this thesis in section 4.9.

The broadcasting series of the Greedy Equal Bandwidth Broadcasting scheme is defined by the following function:

$$S(i) = \left(\frac{d}{W^+} + 1\right)^{\frac{i-1}{n}} \tag{3.30}$$

where d is the total length of the media stream and W^+ is the desired maximum playback delay.

If the playback delay is set to the playback duration of the first segment (for better comparison with the previous schemes), this formula gives the following broadcasting series:

$$1, 2, 4, 8, 16, 32, 64, 128, 256, 512, 1\,024, 2\,048, 4\,096, 8\,192, 16\,384, 32\,768, \ldots \tag{3.31}$$

Figure 3.24 shows an example for two recipients. While it would be possible for recipient #1 to receive only two channels at once in this example, recipient #2 has to receive all four channels.

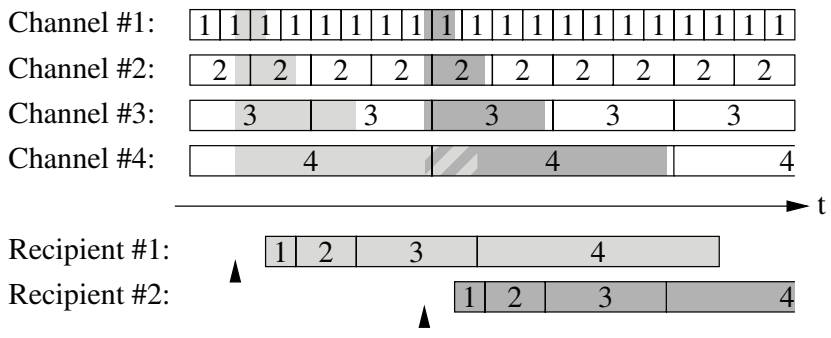


Figure 3.24: Greedy Equal Bandwidth Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** Multiple of the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** **!** Full live streaming capable

Table 3.25: Classification for Greedy Equal Bandwidth Broadcasting transmissions

3.5.8 Fibonacci Broadcasting

The Fibonacci Broadcasting scheme [YK03] is a transmission scheme which uses the Fibonacci series (leaving out the first element) as broadcasting series, so its broadcasting series is defined by the following function:

$$S(i) = \begin{cases} 1 & \text{if } i = 1, \\ 2 & \text{if } i = 2, \\ S(i-2) + S(i-1) & \text{if } i \geq 3 \end{cases} \tag{3.32}$$

which gives the following broadcasting series:

$$1, 2, 3, 5, 8, 13, 21, 34, 55, 89, 144, 233, 377, 610, 987, 1\,597, \ldots \tag{3.33}$$

Using Fibonacci Broadcasting, the receiver systems only have to receive two channels at a time.

Similar to the Greedy Equal Bandwidth Broadcasting scheme, this transmission scheme uses the efficiency advantage that is gained if the receiver systems receive data from the channels immediately after joining them instead of waiting for the beginning of the segments after joining.

Figure 3.25 shows an example transmission using the Fibonacci Broadcasting scheme.

3.5.9 Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting

The Reliable Periodic Broadcasting scheme [MEVSS01], which has also been published independently by the authors of the Fibonacci Broadcasting scheme some time later as Generalized

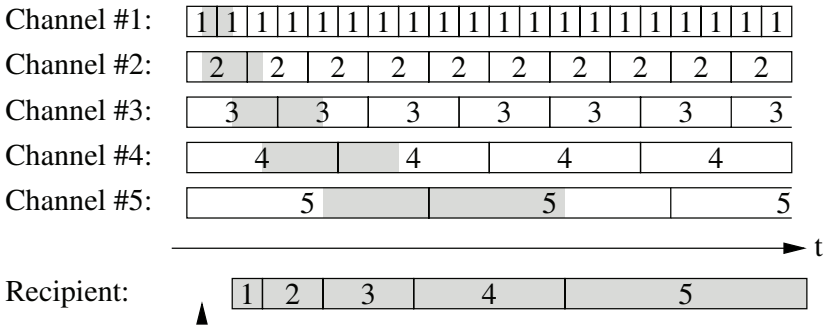


Figure 3.25: Fibonacci Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** ! Twice the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Full live streaming capable

Table 3.26: Classification for Fibonacci Broadcasting transmissions

Fibonacci Broadcasting scheme [YK03], can be seen as a successor of the Fibonacci Broadcasting scheme. Compared to the Fibonacci Broadcasting scheme, it has been generalized in such a way that the channel bandwidth and the number of channels a receiver system is able to receive simultaneously can be configured.

The broadcasting series of the Reliable Periodic Broadcasting scheme/Generalized Fibonacci Broadcasting scheme can be determined from the following formula:

$$S(i) = \begin{cases} 1 + \dfrac{\sum_{k=1}^{i-1} S(k)}{g} & \text{if } i \leq R, \\ \dfrac{\sum_{k=i-R}^{i-1} S(k)}{g} & \text{if } i > R \end{cases} \tag{3.34}$$

where g is a parameter which specifies the bandwidth of the transmission channels, i. e. the bandwidth of the channels is $\frac{1}{g}$ times the media bit rate.

For $g = 1$ (for better comparison), this gives the following broadcasting series:

$$\begin{aligned}
R = 2: &\quad 1, 2, 3, 5, 8, 13, 21, 34, 55, 89, 144, 233, 377, 610, 987, 1\,597, 2\,584, \ldots \\
R = 3: &\quad 1, 2, 4, 7, 13, 24, 44, 81, 149, 274, 504, 927, 1\,705, 3\,136, 5\,768, 10\,609, \ldots \\
R = 4: &\quad 1, 2, 4, 8, 15, 29, 56, 108, 208, 401, 773, 1\,490, 2\,872, 5\,536, 10\,671, 20\,569, \ldots \\
R = 5: &\quad 1, 2, 4, 8, 16, 31, 61, 120, 236, 464, 912, 1\,793, 3\,525, 6\,930, 13\,624, 26\,784, \ldots
\end{aligned} \tag{3.35}$$

As one can see, $g = 1$ and $R = 2$ gives the broadcasting series of the Fibonacci Broadcasting scheme and $g = 1$ and $R = \infty$ gives the broadcasting series of the Greedy Equal Bandwidth Broadcasting scheme.

Again, this transmission scheme requires that the receiver systems receive data from the channels immediately after joining them.

Figure 3.26 shows an example transmission using the Reliable Periodic Broadcasting scheme/Generalized Fibonacci Broadcasting scheme using channels of half the media bit rate and a receiver bandwidth of three channels ($g = 2$, $R = 3$).

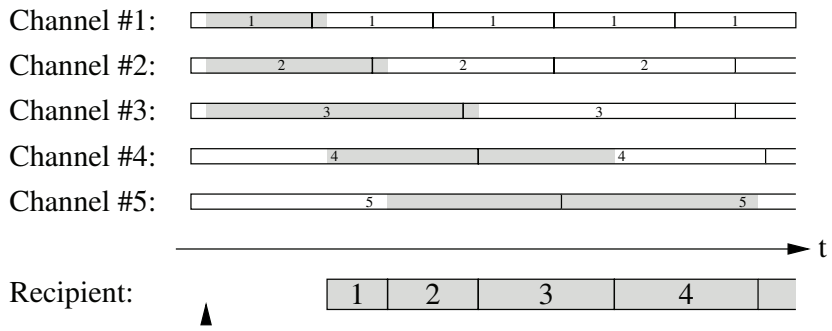


Figure 3.26: Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting

3.5.10 Comparison of Size-Based Segmenting Transmission Schemes

Comparing all size-based transmission schemes with each other is very difficult as the transmission schemes differ in several points:

- The number of channels which have to be received simultaneously by the client systems are not the same for all transmission schemes: For Pyramid Broadcasting and Permutation-based Pyramid Broadcasting, just one channel has to be received at once, for Skyscraper Broadcasting and Fibonacci Broadcasting, two channels must be received in parallel, for Mayan Temple Broadcasting and Greedy Equal Bandwidth Broadcasting, all channels have

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** One position per channel **Receiver bandwidth:** ! Configurable multiple of the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** One position per channel	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Full live streaming capable

Table 3.27: Classification for Reliable Periodic Broadcasting/Generalized Fibonaccy Broadcasting transmissions

to be received simultaneously and using the Client-Centric Approach, the Greedy Disk-Conserving Broadcasting or the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting, it is possible to specify the number of receiver channels.

- For the Pyramid Broadcasting and the Permutation-based Pyramid Broadcasting scheme, the bandwidth of the channels is higher than the media bit rate while all other transmission schemes allow using a channel bandwidth equal to the media bit rate. The Mayan Temple Broadcasting scheme and the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting scheme even allow specifying a channel bandwidth independent of the media stream bit rate.
- The Mayan Temple Broadcasting scheme supports variable-bit-rate media streams, all other transmission schemes only support constant-bit-rate media streams.
- The Skyscraper Broadcasting scheme and the Mayan Temple Broadcasting scheme allow specifying a limit for the required storage of the receiver system.
- The Mayan Temple Broadcasting scheme supports partial preloading.
- The Greedy Equal Bandwidth Broadcasting scheme allows defining the playback delay independent of the size of the first segment.

To get an overview of the efficiency of all size-based transmission schemes, most of these features have to be ignored and only a few setups are examined: In a first setup, the receiver system can only receive two channels equal to the media bit rate, while in a second setup, no limit is imposed on the receiver bandwidth. To include Pyramid Broadcasting and Permutation-based Pyramid

Broadcasting in this comparison, a special setup is examined where the receiver system is allowed to receive a channel of a bandwidth higher than the media bit rate[11]. In all setups, the receiver storage is pretended to be sufficiently large and neither partial preloading nor variable-bit-rate media streams are used in any of the setups. Additionally, a minimum segment size of $\frac{1}{7200}$ of the media stream size is assumed to limit splitting of the media stream.

Figures 3.27 to 3.29 show the results for each of the three setups, resp.[12] For the two-channel setup (see figure 3.27), the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting scheme performs best if it is used with high parameter g. The reason for this is that the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting scheme allows increasing the number of segments independent from the sender and receiver bandwidth which allows approximating the transmission frequency of the transmitted parts more exactly near to the theoretically needed frequency. For $g = 1$, the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting scheme is identical to the Fibonacci Broadcasting scheme.

If the receiver system is able to receive the complete transmission at once (see figure 3.28), the Greedy Equal Bandwidth Broadcasting scheme and the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting scheme produce the same results (if the channel bandwidth of the Greedy Equal Bandwidth Broadcasting scheme is set to $\frac{1}{g}$ of the media stream bit rate where g is the parameter of the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting scheme). The Greedy Disk Conserving Bandwidth scheme and the Client Centric Approach deliver the same results as the Greedy Equal Bandwidth Broadcasting scheme and the Reliable Periodic Broadcasting/Generalized Fibonacci Broadcasting scheme with $g = 1$.

From figure 3.29 it is clearly visible that the Pyramid Broadcasting scheme and the Permutation-based Pyramid Broadcasting scheme perform much worse than the other size-based schemes. Please note that these schemes use integer arithmetic to calculate some of their parameters which results in several lines in the efficiency graph. As the parameters are not a direct result of the playback delay, several parameters are even valid for some playback delay values.

3.6 Bandwidth-Based Segmenting Transmission Schemes

In the previous section, the size-based transmission schemes improved the efficiency of the media-on-demand transmissions by increasing the size of later segments and therefore reducing the transmission frequency of these segments. The bandwidth-based transmission schemes use another

[11] The channel bandwidth of Pyramid Broadcasting and Permutation-based Pyramid Broadcasting varies depending on the total bandwidth and the calculated α-parameter. As the efficiency values of both the Pyramid Broadcasting scheme and the Permutation-based Pyramid Broadcasting scheme are very poor, this is not examined in more detail in this thesis.

[12] Please note that some of the graphs have been truncated because the segments became smaller than $\frac{1}{7200}$ of the media stream duration.

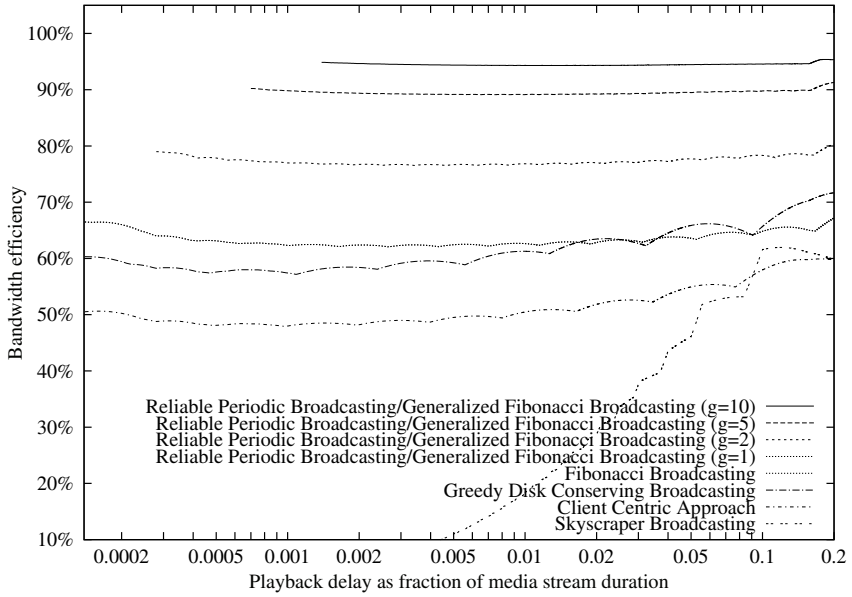


Figure 3.27: Efficiency of size-based schemes when only twice the media bit rate can be received

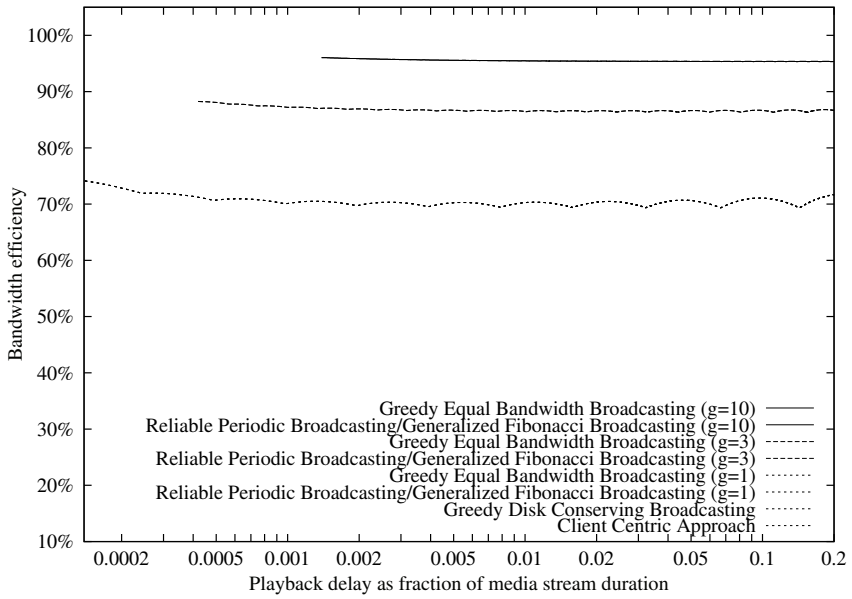


Figure 3.28: Efficiency of size-based schemes when the whole transmission can be received at once

approach to achieve a similar effect: Similar to the size-based transmission schemes, they use one channel for the transmission of each segment, but instead of increasing the segment sizes for later segments, segments of equal size are used and the bandwidth of the channels is decreased. The

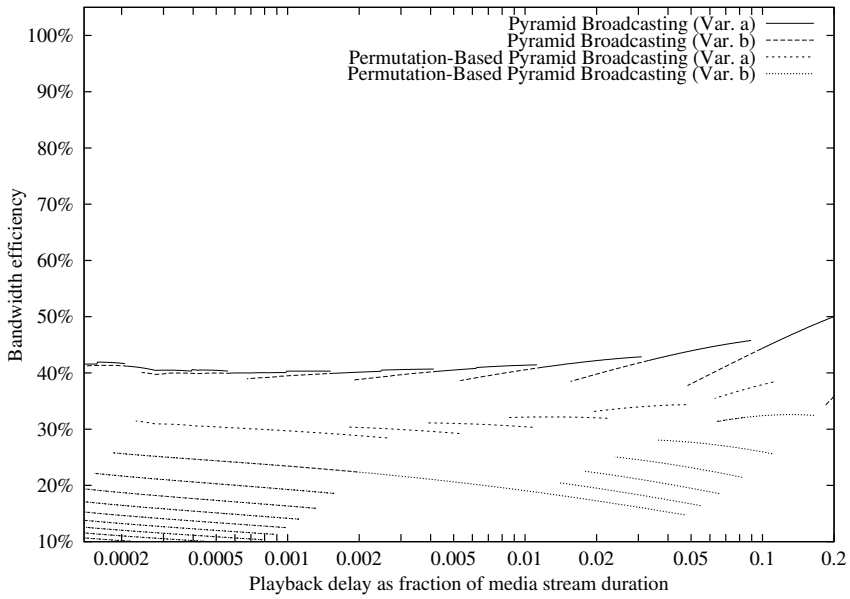


Figure 3.29: Efficiency of size-based schemes where the receiver has to receive two high-bandwidth channels

result of this bandwidth modification is that later segments take longer to be transmitted, so their transmission frequency is lowered this way.

The remainder of this section describes transmission schemes which benefit from this approach. The first published scheme of this class is especially interesting as it supplies some mathematical background for the calculation of a lower bound for on-demand transmissions, although the scheme is not functional (as proven in [PCL98b]). It is followed by a series of transmission schemes which correct this problem in different ways or focus on some other issues.

3.6.1 Harmonic Broadcasting

The Harmonic Broadcasting scheme [JT97a] uses segments of equal size and a very simple formula for the bandwidth of the channels on which the segments are transmitted:

$$B_i = \frac{b}{i} \tag{3.36}$$

Figure 3.30 shows an example transmission using the Harmonic Broadcasting scheme.

The idea behind this simple formula is that segment i can be transmitted at one i-th of the media bit rate because the receiver system has i times the duration of the first segment to receive segment #i before it is needed for playback.

The above formula assures that the segment start is received when the playback time of the

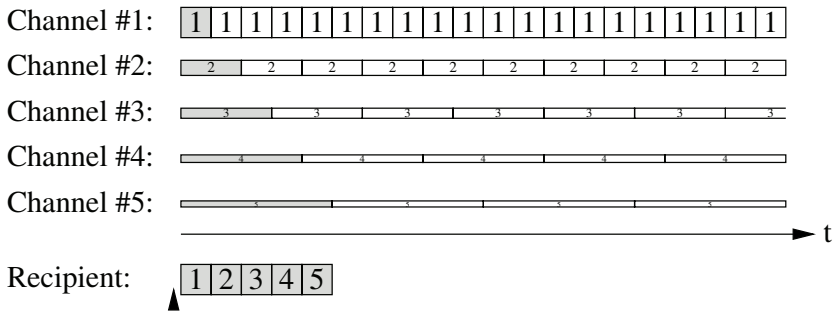


Figure 3.30: Harmonic Broadcasting

segment is reached and that the segment has been completely received when its playback time ends. But unfortunately, this does not ensure that the segment can be played as is illustrated in figure 3.31, a problem which results from the fact that the channel bandwidth is lower than the media bit rate. A simple way to correct this problem is to double the playback waiting time at the receiver side, but this decreases the efficiency of the transmission scheme a lot. In the following subsections, other solutions are proposed.

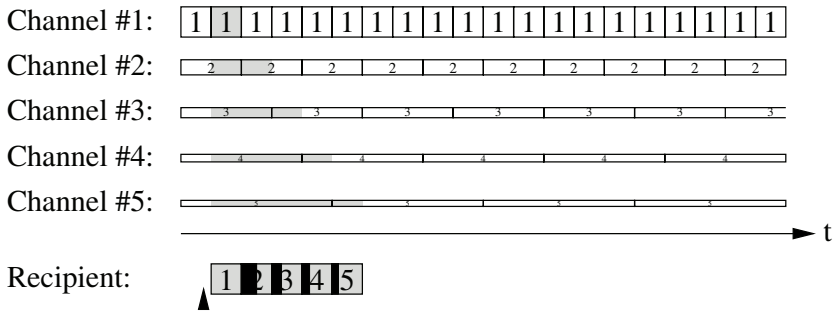


Figure 3.31: Example for failure of Harmonic Broadcasting: The blackened parts at the recipient are missing when the media stream is played

Although this means that the transmission scheme is non-functional, it marks a very important step in the evolution of transmission schemes as it introduced a new and very efficient class of transmission schemes.

3.6.2 Cautious Harmonic Broadcasting

The authors of [PCL98b] not only discovered that the Harmonic Broadcasting Protocol is not working in the proposed version, they also provided three broadcasting schemes as solutions which are presented in this and the following two subsections.

The Cautious Harmonic Broadcasting scheme [PCL98b] is one of their solutions. For this

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** **!** Multiple positions per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** **!** Multiple positions per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: **!!!** This transmission scheme is non-functional	

Table 3.28: Classification for Harmonic Broadcasting transmissions

transmission scheme, the media stream is cut into segments of equal size, but the channel assignment is changed slightly if compared to the Harmonic Broadcasting scheme: On the first channel, the first segment is transmitted at full bandwidth. The second channel alternately transmits the second and third segment at full bandwidth (which is — strictly speaking — against the rules of bandwidth based transmission schemes but these two segments form the only exception). Channels three to $n-1$ are used to transmit the remaining segments four to n at decreasing bandwidth, using one $(i-1)$-th of the media stream bit rate as bandwidth for the transmission of the $(i+1)$-th segment on the i-th channel.

Figure 3.32 shows an example for a Cautious Harmonic Broadcasting transmission.

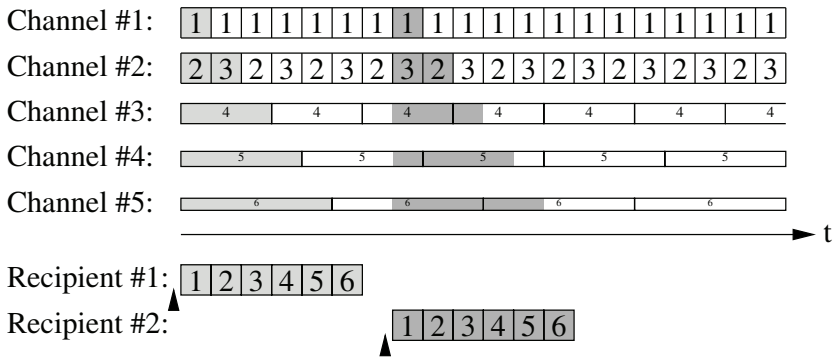


Figure 3.32: Cautious Harmonic Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** ! Multiple positions per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** ! Multiple positions per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.29: Classification for Cautious Harmonic Broadcasting transmissions

3.6.3 Quasi-Harmonic Broadcasting

The Quasi-Harmonic Broadcasting scheme [PCL98b] is another solution to solve the problems of the Harmonic Broadcasting Protocol.

For the Quasi-Harmonic Broadcasting scheme, the media stream is not only cut into segments of equal size, all segments except the first one are further cut into smaller subsegments and transmitted using a complex scheme:

The first segment is simply transmitted on the first channel at the full media stream bit rate. But channel $i \geq 2$ transmits all subsegments of the segment i at $\frac{m}{i \cdot m - 1}$ times the media stream bit rate in the order specified by the series

$$\left(\begin{cases} i \cdot (t \bmod m) + \left\lfloor \frac{t \bmod im}{m} \right\rfloor & \text{if } t \bmod m \neq m-1, \\ (t \bmod (i-1)) + 1 & \text{if } t \bmod m = m-1 \end{cases} \right)_{t \in \{0,1,2,\ldots,i \cdot m - 1\}} \tag{3.37}$$

where $m \geq 2$ is a parameter. The advantage of the Quasi-Harmonic Broadcasting scheme is that it converges to the same bandwidth requirements as the non-functional Harmonic Broadcasting scheme if m tends towards infinity. Figure 3.33 shows an example transmission for the Quasi-Harmonic Broadcasting scheme.

3.6.4 Polyharmonic Broadcasting

The Polyharmonic Broadcasting scheme [PCL98c] uses another trick to get around the design error of Harmonic Broadcasting: Instead of starting the playback when the beginning of the first

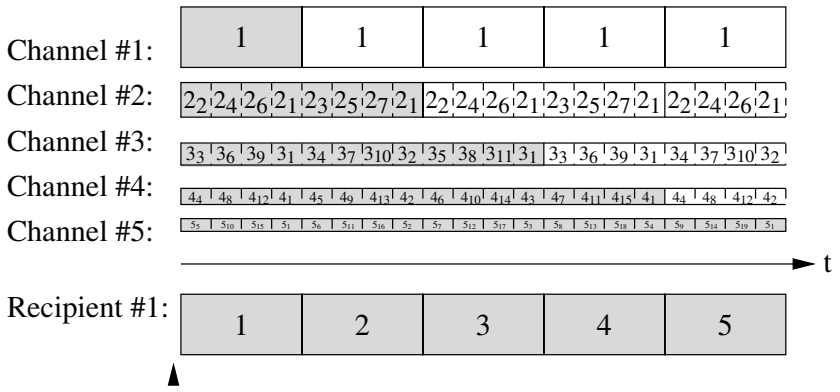


Figure 3.33: Quasi-Harmonic Broadcasting with parameter $m = 4$

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** ! Multiple positions per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** ! Multiple positions per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.30: Classification for Quasi-Harmonic Broadcasting transmissions

segment is received (and therefore the next slot interval boundary is reached), Polyharmonic Broadcasting uses the fixed wait policy which already has been presented in section 3.5.7 for the Greedy Equal Bandwidth Broadcasting scheme, i. e. segments are received immediately after joining the channel (instead of waiting for the next segment boundary when joining) and the playback is started a fixed time after the receiver system started reception (and not at the next segment boundary). This way, Polyharmonic Broadcasting assures that each segment has been completely received when its playback is started.

As the bandwidth division and segment mapping are the same as of Harmonic Broadcasting, Polyharmonic Broadcasting reaches the bandwidth promises of Harmonic Broadcasting. The only difference is that the minimum playback delay equals the maximum playback delay for the Poly-

harmonic Broadcasting scheme due to the fixed wait policy, most other broadcasting schemes (without fixed wait policy) have a minimum playback delay of zero.

Another extension of the Polyharmonic Broadcasting scheme is that the playback delay can be selected as an arbitrary multiple of the segment size (shown as parameter *m* in comparison graph in figure 3.39 at the end of this section). This allows reducing the segment size while keeping the playback delay constant and thereby reducing the total bandwidth requirement. The consequences of this idea are examined in more detail in sections 4.4 and 4.5.

The authors of [ES01, ES02] proved that Polyharmonic Broadcasting has the lowest possible bandwidth requirements for a given segment size, so its efficiency equals 100 % for a given segment size. Figure 3.34 shows an example transmission for the Polyharmonic Broadcasting scheme.

Polyharmonic Broadcasting is also able to transmit variable-bit-rate media streams [Hu01]: In this case, the duration for the transmission of a segment is determined from its playback time (or decode time if this is lower) and the bandwidth of the channel of the segment is calculated according to the size of the segment.

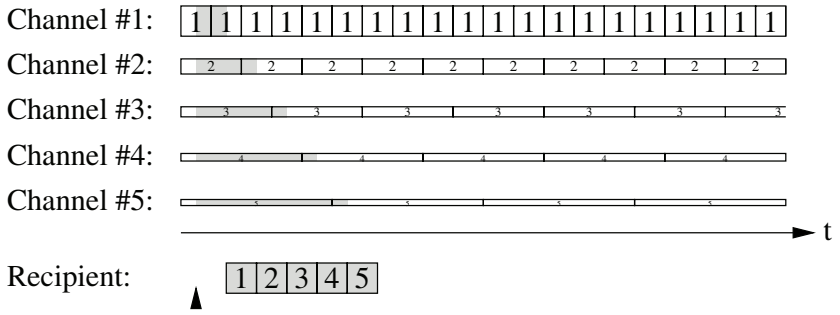


Figure 3.34: Polyharmonic Broadcasting

3.6.5 Tailored Transmissions

The Tailored Transmission scheme [BM99] can be seen as an extension of the Polyharmonic Broadcasting scheme. It uses the same basic idea to circumvent the problems of Harmonic Broadcasting and allows decoupling the playback delay from the slot interval but introduced more flexibility at the same time:

- The transmission scheme supports variable-bit-rate media streams more efficiently. The underlying idea of the Tailored Transmission scheme for supporting variable-bit-rate media streams is to use different bandwidths for each segment which are chosen in such a way that the segments are transmitted at the lowest possible period. This idea can indeed be used for any of the transmission schemes of the bandwidth-based class although the Tailored

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** **!** Multiple positions per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** **!** Multiple positions per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.31: Classification for Polyharmonic Broadcasting transmissions

Transmission scheme is the only one which describes this approach. Further information on variable-bit-rate transmissions is given in section 4.8.

- The transmission scheme allows selecting all but one of the design parameters media length, playback delay, receiver bandwidth, transmission bandwidth and receiver storage requirements and throughput to create a matching transmission scheme.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** **!** Multiple positions per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** **!** Multiple positions per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.32: Classification for Tailored Broadcasting transmissions

3.6.6 Staircase Broadcasting

The Staircase Broadcasting scheme [JT97b] is another transmission scheme of the class of bandwidth-based transmission schemes. In contrary to the above ones, it is based on a binary splitting of the bandwidth, i. e. segments $\#2^k$ to $\#2^{k+1}-1$ are transmitted at $\frac{1}{2^k}$ of the media bit rate ($k \geq 0$). As a result, the channels of the transmission can be aggregated into channels at media bit rate.

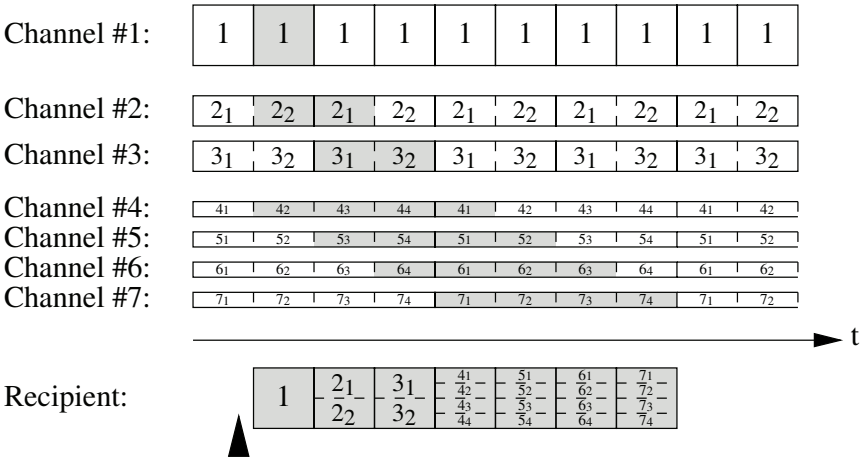
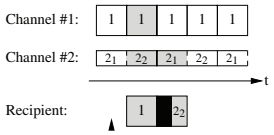


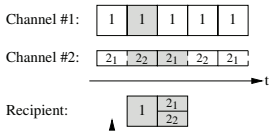
Figure 3.35: Staircase Broadcasting

To circumvent the problems of Harmonic Broadcasting, the authors of the Staircase Broadcasting scheme used another type of subsegment splitting: Instead of splitting a segment into subsegments of same duration to transmit them on low bandwidth channels (**vertical partitioning**), they split the segment transmission into streams of same bandwidth (**horizontal partitioning**). This allows playing a segment even when the first subsegment is just received — which was the problem of Harmonic Broadcasting. Figure 3.35 shows an example for a Staircase Broadcasting transmission and figure 3.36 shows the difference between vertical and horizontal partitioning for a two segment transmission.

Although this splitting technique seems to be a completely new idea, it has already been proposed similarly in a previous transmission scheme. To understand this, it is important to realize that transmitting a segment in streams of same bandwidth has to be performed by splitting it into small parts which are interleaved for transmission. But this introduces an additional, small playback delay as the first part of the first subsegment has to be played at full speed while it is received with lower bandwidth. This problem can be solved if the first parts of the first subsegments of the segments are transmitted more often. Exactly this approach is used in the Quasi-Harmonic Broadcasting scheme, where the parameter m describes in how many parts a subsegment is split.



(a) Subsegments of same duration produce problems at Harmonic Broadcasting



(b) Subsegments of same bandwidth can be used for instead

Figure 3.36: Comparison of different sub-segmentation schemes

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** ! Multiple positions per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** ! Multiple positions per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.33: Classification for Staircase Broadcasting transmissions

3.6.7 Seamless Staircase Broadcasting

The Seamless Staircase Broadcasting scheme [TCS04] has a very similar structure as the Staircase Broadcasting scheme which has been proposed in the previous subsection. Similar to the Staircase Broadcasting scheme, the segments are transmitted for this transmission scheme in such a way that the channels can be aggregated into channels at media bit rate.

The main difference between the Seamless Staircase Broadcasting scheme and the Staircase Broadcasting scheme is that the Seamless Staircase Broadcasting scheme does not use the binary horizontal partitioning scheme for the first three channels. Instead, the following schedule is used:

- The first segment is transmitted at media bit rate.
- The second and third segment are transmitted at media bit rate alternately.
- Segment #$3 \cdot 2^k + 1$ to #$3 \cdot 2^{k+1}$ are transmitted at $\frac{1}{3 \cdot 2^k}$ of the media bit rate ($k \geq 0$).

Figure 3.37 shows an example for a Seamless Staircase Broadcasting transmission.

Although this transmission scheme is less efficient than the original Staircase Broadcasting

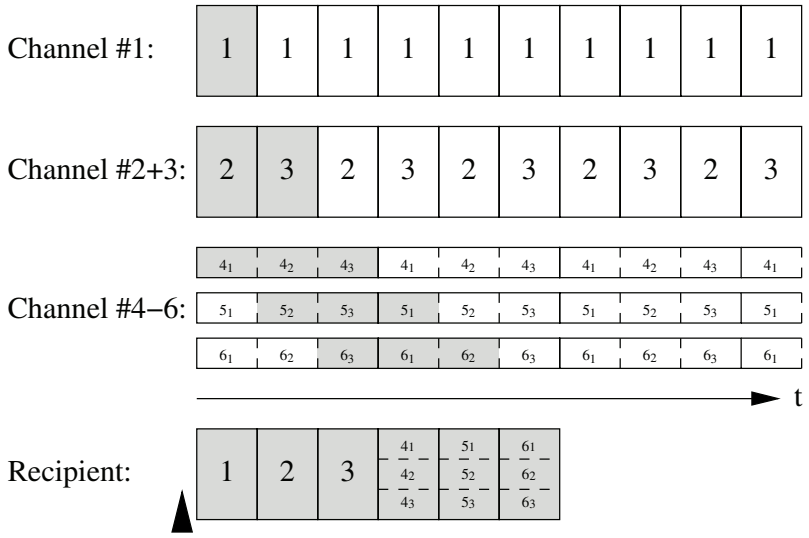


Figure 3.37: Seamless Staircase Broadcasting

scheme, the Seamless Staircase Broadcasting scheme can be regarded as an enhanced version of the former one as it supports a **seamless bandwidth change**: Often the media-on-demand providers want be able to change the bandwidth assignments to media streams to increase or decrease the playback delay dynamically. As ongoing transmissions should not be disrupted, the only chance to perform these changes for most transmission schemes is to start the transmission with the new parameters in parallel to the ongoing one and "fading out" the old transmission. The disadvantage of this procedure is that both transmissions consume bandwidth during the transition time.

With the Seamless Staircase Broadcasting scheme, the bandwidth can be increased dynamically (in steps of the media bit rate): Each time the bandwidth is increased by the media bit rate, the playback delay halves a short time thereafter[13]. It is even possible to double the playback delay at a later time up to the initial value, thereby releasing bandwidth again after the transition finished.

Figure 3.38 shows an example where bandwidth is dynamically added, including an ongoing playback (recipient #1 in the figure) which continues uninterrupted after the bandwidth change and a playback (recipient #2) which is served after the bandwidth has been changed. Seamless bandwidth changes are re-examined in section 4.16 as an extension to the newly proposed Generalized Greedy Broadcasting scheme.

[13]Unfortunately, the Seamless Staircase Broadcasting scheme cannot serve playback requests while this transition takes place, a fact which has not been mentioned by the authors of the transmission scheme.

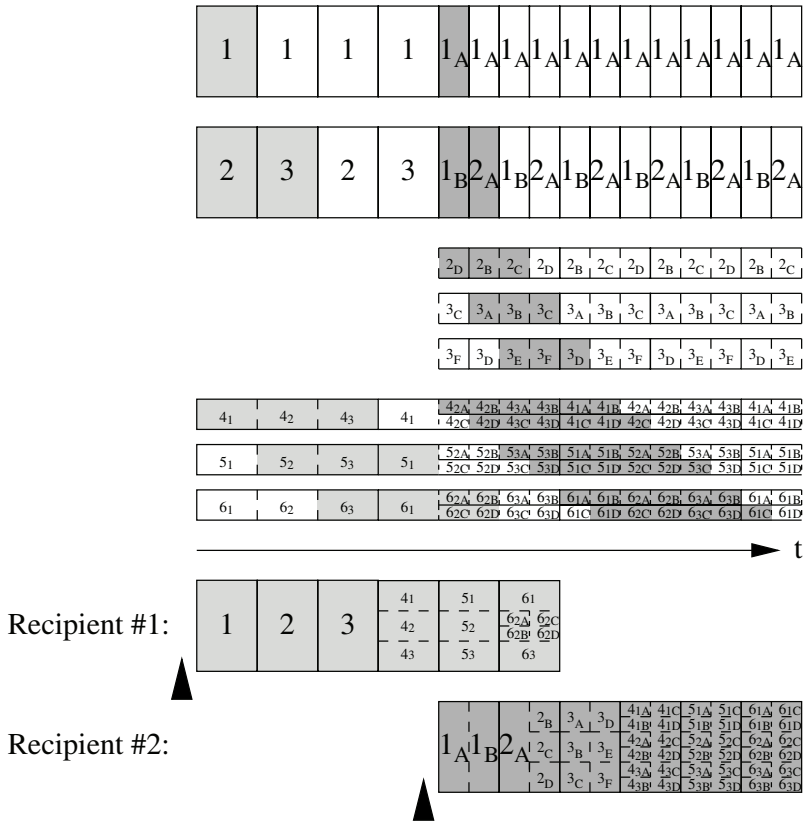


Figure 3.38: Seamless Staircase Broadcasting

3.6.8 Comparison of Bandwidth-Based Segmenting Transmission Schemes

Comparing all bandwidth-based segmenting transmission schemes with each other imposes similar difficulties as the comparison of the size-based segmenting transmissions schemes:

- The Quasi-Harmonic Broadcasting scheme, the Polyharmonic Broadcasting scheme and the Tailored Transmission scheme support a parameter[14], which, if set to a high value, increases the efficiency. For the Polyharmonic Broadcasting scheme and the Tailored Transmission scheme, the efficiency even tends towards 100 % if the parameter is increased indefinitely. (But as this also increases the number of segments, calculations are stopped when the transmission scheme uses more segments than the lower bound bandwidth calculation assumed, otherwise the efficiency of these transmission schemes would advance beyond 100 % because of the smaller segment sizes.)
- The Polyharmonic Broadcast scheme and the Tailored Transmission scheme also supports

[14] In the original article of the Tailored Transmission scheme, the ratio of playback delay to segment size has not been named. As the Tailored Transmission scheme can be seen as an extension of the Polyharmonic Broadcasting scheme, the same parameter definition $m = \frac{W^+}{\delta}$ is used in this thesis.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** **!** Multiple positions per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** **!** Multiple positions per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** **!** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: **!!!** No playback requests can be processed during bandwidth change	

Table 3.34: Classification for Seamless Staircase Broadcasting transmissions

variable-bit-rate media streams and allows restricting the receiver bandwidth, receiver storage size and throughput.

- The Seamless Staircase Broadcasting scheme supports seamless bandwidth changes for ongoing transmissions.
- For the Polyharmonic Broadcasting scheme and the Tailored Transmission scheme, the minimum playback delay equals the maximum playback delay, for the other transmission schemes of the bandwidth based group, the minimum playback delay is zero.

To compare the bandwidth-based transmission schemes, the following assumptions have been made:

- The bandwidth of the receiver systems is high enough to receive any of the transmissions.
- The storage space available at the receiver systems and the throughput of it are high enough to cope with the transmission.
- A constant-bit-rate media stream is transmitted.
- Only the maximum playback delay is taken into account when the efficiency is calculated.

Under these preconditions, figure 3.39 shows the efficiency of the bandwidth-based transmission schemes. (The Harmonic Broadcasting scheme has been omitted as it is not working correctly, see section 3.6.1.) It is clearly visible, that the Polyharmonic Broadcasting scheme and the Tailored

Transmission scheme (which produce identical results under the above preconditions) outperform the other bandwidth-based transmission schemes.

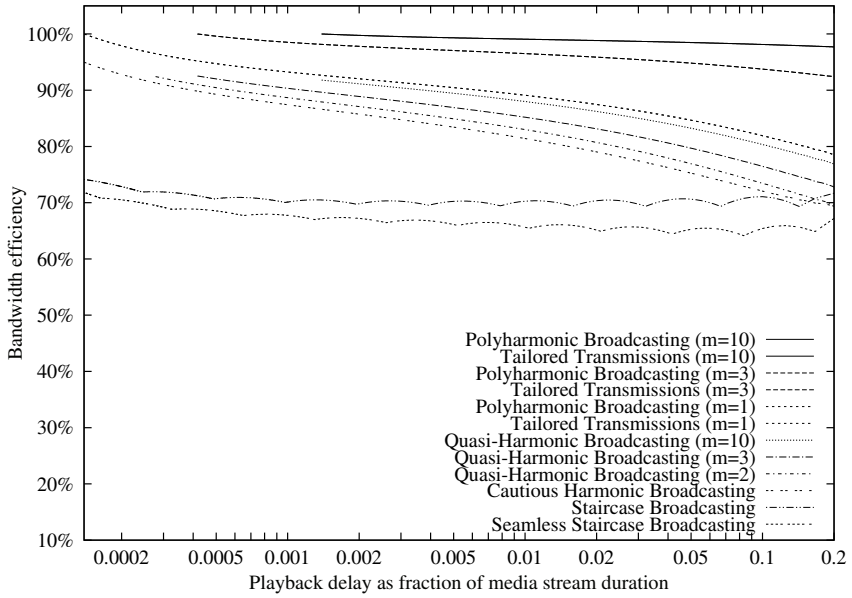


Figure 3.39: Efficiency of bandwidth-based schemes

3.7 Frequency-Based Segmenting Transmission Schemes

The third class of pro-active media-on-demand transmission schemes uses channels of equal bandwidth where segments of equal size are transmitted. Instead of using segments of increasing size or decreasing bandwidth, the frequency-based transmission schemes transmit the segments at decreasing frequency. As the average bandwidth which is used for the transmission of the segments decreases for the later segments, bandwidth is saved.

The main advantage of this procedure is that all segments are of equal size and that all channels use the same bandwidth. But to perform a successful frequency-based transmission, the segments have to be arranged on the channels in such a way that no two segments of one channel have to be transmitted at the same time. As the calculation of the optimal arrangement is NP-hard (this problem is a variant of the bin-packing problem [Wik08g] as proven in [BNBNS98]), only non-perfect solutions are available for large transmission schedules, so the transmission schemes which are presented in this section mainly differ in the algorithm they use for arranging the segments.

3.7.1 Harmonic Equal-Bandwidth Broadcasting

Although never published separately, this broadcasting scheme is referenced several times when it comes to frequency-based transmission schemes. The reason therefore is that it is the most efficient frequency-based transmission scheme: It has an efficiency of 100 %.

The Harmonic Equal-Bandwidth Broadcasting scheme uses a single channel for each segment, each at the media stream bit rate (assuming a constant-bit-rate media stream), and the i-th segment is transmitted every $i + \left\lfloor \frac{W^+}{\delta} \right\rfloor - 1$ slot intervals on its channel. (Similar to the Polyharmonic Broadcasting scheme, it is possible to use a playback delay independent from the segment size to increase the efficiency.) Due to its similarities with the Polyharmonic Broadcasting scheme, the Harmonic Equal-Bandwidth Broadcasting scheme reaches the same efficiency as the Polyharmonic Broadcasting scheme, but without the need for a fixed wait policy.

Unfortunately, the bandwidth of this transmission scheme is extremely varying: At the beginning of the transmission ($t = 0$), only one channel transmits a segment, but at time offset $t = \delta \cdot \operatorname{lcm}_{1 \le i \le n} i$ all n channels are in use[15] which makes the transmission scheme unusable in practical environments. Figure 3.40 shows an example for five channels.

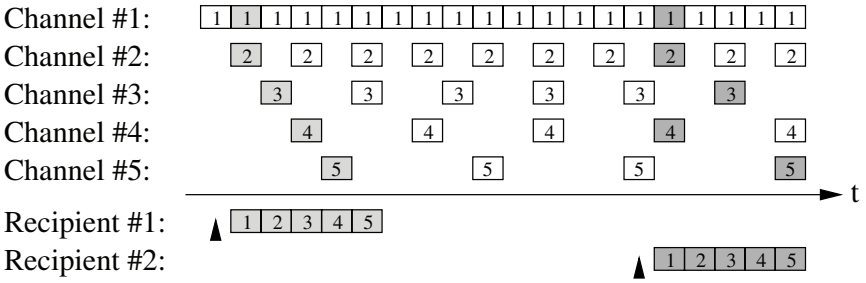


Figure 3.40: Harmonic Equal-Bandwidth Broadcasting

3.7.2 Fast Broadcasting

The Fast Broadcasting transmission scheme [JT98] was the first scheme which has been published for the frequency-based class and started this new, very efficient and useful type of transmission schemes. It uses a simple, binary scheme for the segment arrangement as illustrated in figure 3.41.

To write a frequency-based transmission scheme down, several different notations are used in the literature. The following are the most important ones:

- As frequency-based transmission schemes are periodic, the most simple method for showing a frequency-based transmission scheme is to enumerate the segments of one period for each channel. Sadly, the period becomes very large for some transmission schemes, e. g. the

[15] The lcm operator denotes the least common multiple.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** **!** Full live streaming capable
Comment: **!!!** Very strong bandwidth variations during transmission, practically unusable	

Table 3.35: Classification for Harmonic Equal-Bandwidth Broadcasting transmissions

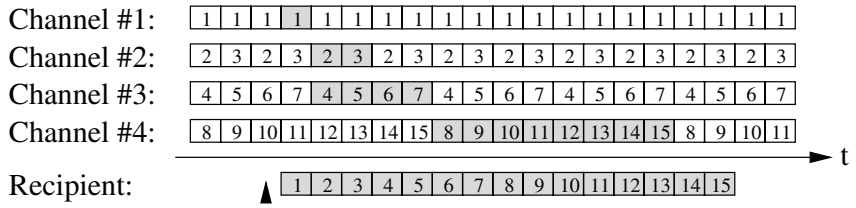


Figure 3.41: Fast Broadcasting

period of one channel of the Greedy Broadcasting scheme using five channels is 5 880 slot intervals, and for the same transmission scheme using eight channels, it is about $91 \cdot 10^{15}$ slot intervals. For this reason, the enumeration is only sensible for a fraction of the frequency-based schemes.

- For the New Pagoda Broadcasting scheme [Pâr99b], a rectangular representation has been invented. As the New Pagoda Broadcasting scheme splits the available bandwidth of each channel twice, it uses two dimensions for the representation. The first split of a channel into a number of subsets from which segments are transmitted round-robin is shown by dividing a rectangle into the appropriate number of rows, and for each row, the contained segments are enumerated for one period. As the segments of each row in the New Pagoda Broadcasting scheme are consecutive, this rectangular representation describes the transmission

schemes of the New Pagoda Broadcasting very well, but it cannot be used efficiently for other, more complex transmission schemes, for the same reasons as above.

- A universal representation can be obtained by creating a table containing the period, the phase shift and the channel number of each segment. This table allows deciding when the sender has to transmit a segment in a very simple way: If t counts the slot intervals since the beginning of the transmission, segment #i has to be transmitted whenever $t \bmod \pi_i = \phi_i$ where π_i and ϕ_i denote the period and phase shift of segment #i, resp. Unfortunately, this table is very confusing and should better be used by machines than humans. Another disadvantage of the table representation is that errors in the transmission scheme are very hard to detect.

- For the Greedy Broadcasting Protocol, the authors developed a tree-based representation for transmission schedules, which not only allows a short, concise, and visual notation of transmission schemes but also enables the development of simple, graph-based algorithms for the creation, manipulation and examination of transmission schemes. (In section 4.12, such an algorithm is presented.) This tree-based representation is based on the fact that the frequency-based transmission schemes use time-multiplexing for dividing the available bandwidth of a channel (or of the resulting sub-channel of a preceding time-multiplexing) into smaller parts until they are used to transmit segments. Therefore, the tree-based representation uses trees with the following characteristics:

  - Each channel is represented by one tree.
  - Each segment is represented by a leaf node in the tree which corresponds to its channel.
  - Each time-multiplexing is represented by a non-leaf node in the tree, dividing the bandwidth to its child nodes by transmitting them round-robin. Thus the number of child nodes of this node gives the multiplicity of the multiplexing process and the order of the child nodes[16] specifies the transmission order of segments from the appropriate subtrees.

  The tree-based representation for a transmission scheme is examined in more detail in section 4.1.3.

Figure 3.42 shows these four diagram types for the above Fast Broadcasting Transmission for comparison. Due to its compactness and clearness, tree-based diagrams are used in most cases to illustrate frequency-based transmission schemes from here on.

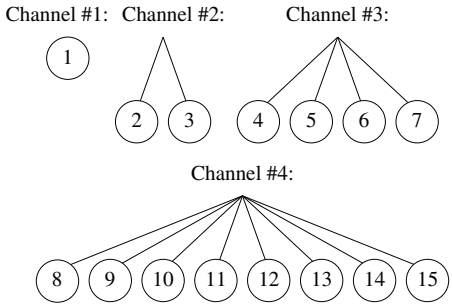
[16] In strict mathematical sense, graphs have no order on their edges or the connected nodes; this may be overcome if a weighted graph is used where the edges are assigned numbers according to the intended order of the child nodes.

(a) Enumerating representation

(b) Rectangular representation

Segment #	Channel #	Period	Phase Shift
1	1	1	0
2	2	2	0
3	2	2	1
4	3	4	0
5	3	4	1
6	3	4	2
7	3	4	3
8	4	8	0
9	4	8	1
...	...	...	...

(c) Tabular representation



(d) Tree representation

Figure 3.42: Comparison of different representations for a frequency-based transmission scheme

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.36: Classification for Fast Broadcasting transmissions

3.7.3 Seamless Fast Broadcasting

The Seamless Fast Broadcasting scheme [THYL+01] is actually nearly identical to the Fast Broadcasting scheme (the segments have just been shifted to another startup position on the channels).

But similar to the Seamless Staircase Broadcasting scheme (see section 3.6.7), the Seamless Fast Broadcasting scheme supports bandwidth changes for ongoing transmissions.

Figure 3.43 shows the process of a seamless channel addition for the Seamless Fast Broadcasting scheme.

Figure 3.43: Example for a channel addition in an ongoing transmission using the Seamless Fast Broadcasting scheme

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** **!** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.37: Classification for Seamless Fast Broadcasting transmissions

3.7.4 Pagoda Broadcasting

As the Fast Broadcasting transmission scheme only performs with poor efficiency, the authors of the Pagoda Broadcasting scheme [YP01] improved it by using another transmission schedule: Instead of a simple binary scheme, Pagoda Broadcasting uses a scheme which has been manually optimized for the first channels while it follows mostly a simple scheme for the latter ones. Figure 3.44 shows an example for four channels and figure 3.45 shows the tree representation of the used transmission schedule.

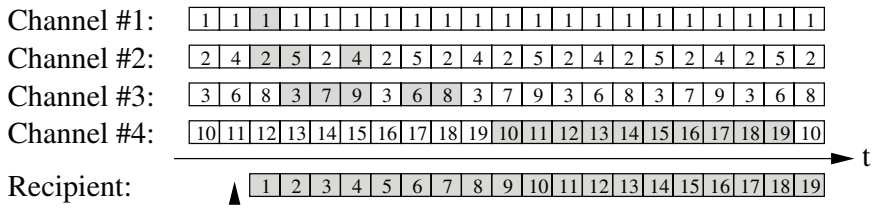


Figure 3.44: Pagoda Broadcasting

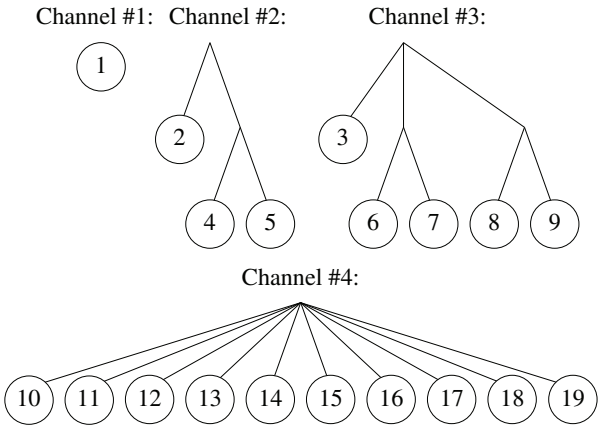


Figure 3.45: Tree representation of a Pagoda Broadcasting transmission scheme

In [PCL99b], the authors extended the Pagoda Broadcasting scheme in two topics: Firstly, the transmission scheme for an even number of channels is improved slightly and secondly, the authors suggest that the last channel may be shared across several media stream transmissions if it is only partially used — a topic which is reviewed in section 4.11.

3.7.5 New Pagoda Broadcasting

Some time after the publication of the Pagoda Broadcasting scheme, the authors of the Pagoda Broadcasting scheme improved their transmission scheme again with the New Pagoda Broadcasting scheme [Pâr99b]. Thereby they used another, hand-optimized segment-to-channel mapping for the first channels and a better algorithm for the latter channels, which is based on the rectangular representation for transmission schemes which they have introduced (see section 3.7.2 for a description of this representation). Figure 3.46 shows an example for four channels, figure 3.47 shows the rectangular and tree representation of the used transmission schedule.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.38: Classification for Pagoda Broadcasting transmissions

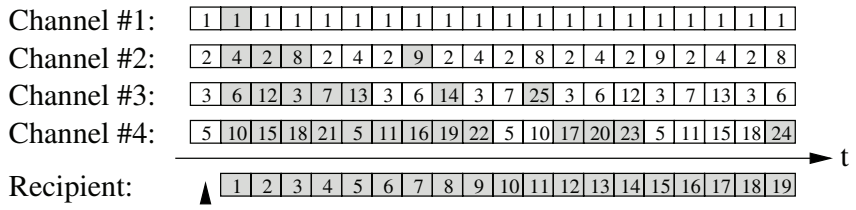


Figure 3.46: New Pagoda Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.39: Classification for New Pagoda Broadcasting transmissions

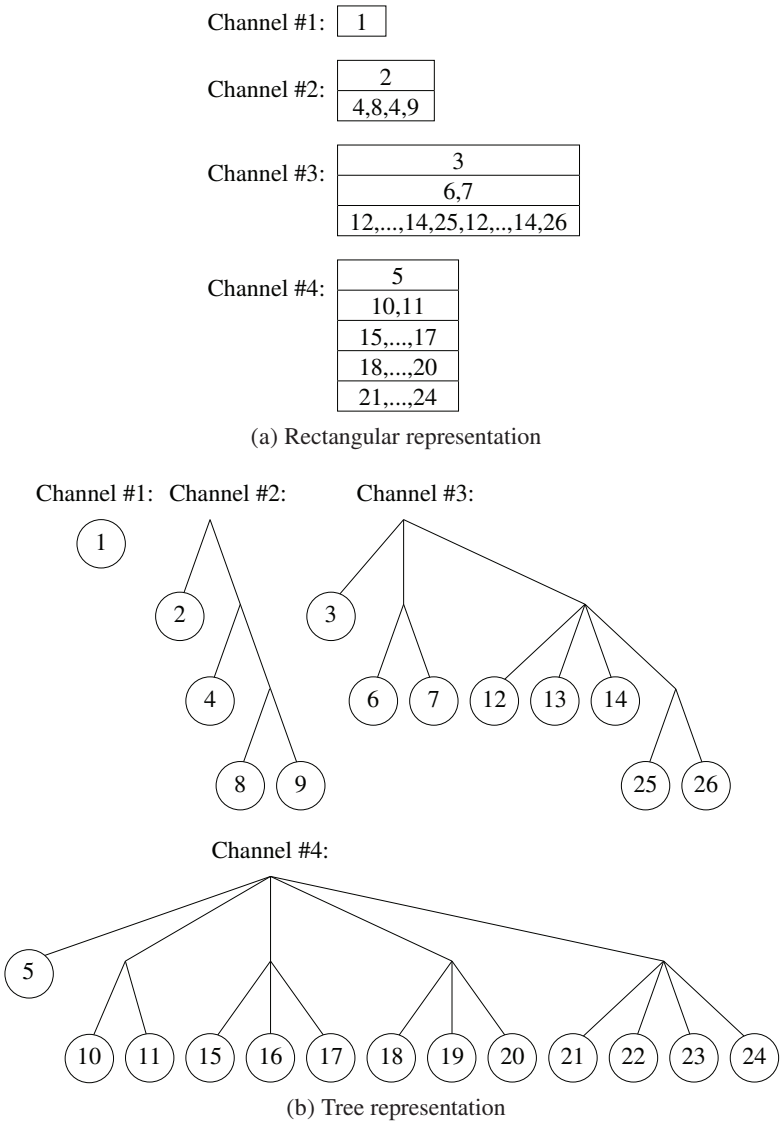


Figure 3.47: Two different representations of a New Pagoda transmission scheme

3.7.6 Fixed Delay Pagoda Broadcasting

An extension to the New Pagoda Broadcasting scheme presented above is the Fixed Delay Pagoda Broadcasting scheme [Pâr01a]. In contrary to the Pagoda and New Pagoda Broadcasting scheme, it is possible to specify an arbitrary playback delay (in multiples of the slot interval) for this transmission scheme. If this playback delay is longer than the one slot interval playback delay of Pagoda/New Pagoda broadcasting, the created transmission schedule requires much less bandwidth. Figure 3.48 shows an example for two channels and a (maximum) playback delay of four slot intervals and figure 3.49 shows the rectangular and tree representation of the used transmission schedule.

Additionally, the Fixed Delay Pagoda Broadcasting scheme allows defining a limit for the receiver bandwidth to create a transmission schedule which does not exceed this limit.

In sections 4.4 and 4.5, the effects of the playback delay and decoupling the playback delay from the slot interval are examined in detail, in section 4.18 the consequences of a limited receiver bandwidth is studied again.

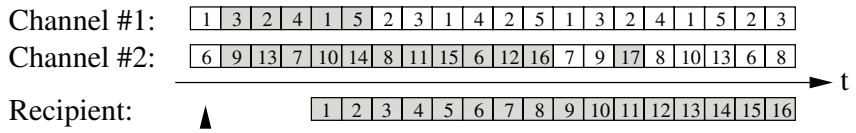


Figure 3.48: Fixed Delay Pagoda Broadcasting

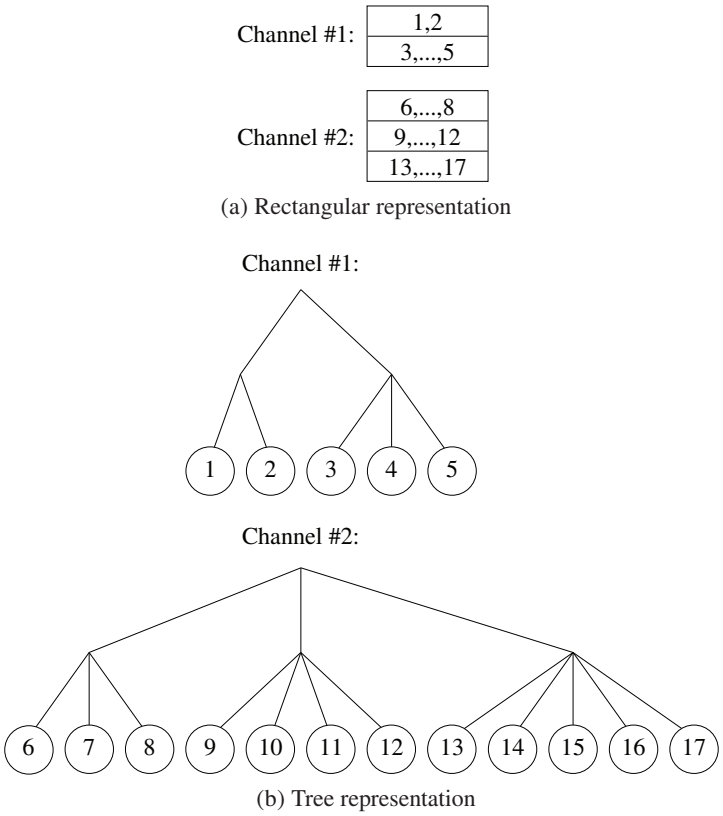


Figure 3.49: Two different representations of a Fixed Delay Pagoda transmission scheme

3.7.7 Variable Bandwidth Broadcasting

The Variable Bandwidth Broadcasting scheme [PL03] is an enhanced version of the New Pagoda Broadcasting scheme: It uses a similar layout and reaches an efficiency similar to the Fixed Delay

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** ! Configurable multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.40: Classification for Fixed-Delay Pagoda Broadcasting transmissions

Pagoda Broadcasting scheme with $W^+ = \delta$. Figure 3.50 shows an example of a Variable Bandwidth Broadcasting transmission.

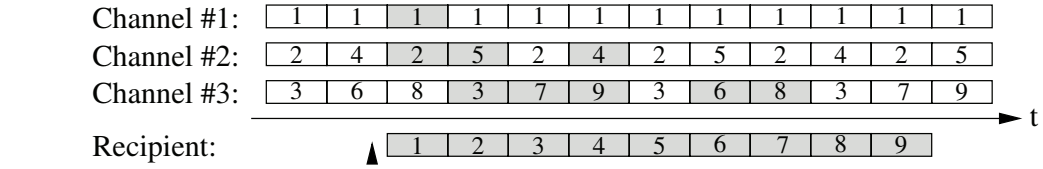


Figure 3.50: Variable Bandwidth Broadcasting

The most interesting point about the Variable Bandwidth Broadcasting scheme is that it supports dynamic bandwidth changes, similar to the Seamless Staircase Broadcasting scheme and the Seamless Fast Broadcasting scheme.

The authors of the Variable Bandwidth Broadcasting scheme also analyzed one of the conditions which is necessary for supporting dynamic bandwidth changes: When a channel is added and the playback delay is decreased from one to a half slot interval, all segments which had been scheduled with their *maximum required period* are sent too infrequently in the new transmission scheme. The Variable Bandwidth Broadcasting scheme solves this problem by moving these segments to new positions (typically by moving all these segments in a chain by one position, i. e. by moving the first segment to the new channel, the second to the old position of the first segment and so on). The authors also proposed the idea to artificially decrease the period by one for the latter segments to prevent that too much segments have to be moved and the transition takes too long.

Unfortunately, the authors missed to recognize a second condition: When a segment is moved

forward in time, it is possible that this interrupts ongoing playbacks. Figure 3.51 shows an example where this segment moving causes a hole in a playback. This approach (together with a solution for the moving problem) is revised in section 4.15.

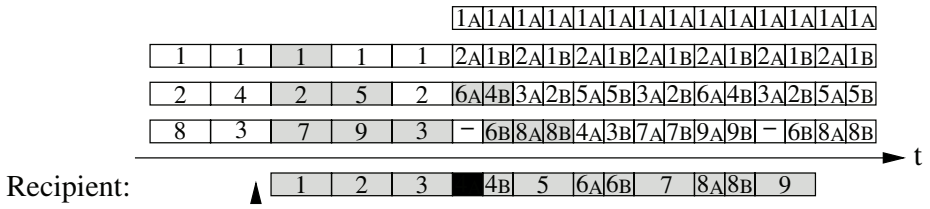


Figure 3.51: Ongoing transmissions may be interrupted when a channel is added using the Variable Bandwidth Broadcasting scheme

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** **!** Full dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: **!!!** Dynamic scheduling change support is malfunctioning	

Table 3.41: Classification for Variable Bandwidth Broadcasting transmissions

3.7.8 Greedy Broadcasting/Recursive Frequency Splitting

The next transmission scheme has been developed independently by two groups at nearly the same time as Greedy Broadcasting scheme [BNL02, BNL03, CST00] and Recursive Frequency Splitting scheme [TYC02].

While the Recursive Frequency Splitting scheme uses a more mathematical description of the algorithm, the authors of the Greedy Broadcasting scheme introduced the tree-based representation for transmission schemes and developed a graph-based algorithm for generating the tree. This

algorithm inserts the segments one-by-one as nodes into the transmission trees at a position which is selected by a simple heuristic:

1. Primarily, the segment is inserted in such a way that the segment is transmitted as often as necessary but as infrequently as possible, so it satisfies the condition for correct playback but wastes the least bandwidth.
2. If the bandwidth loss for several nodes is equal, the node with the highest transmission period is selected, leaving the nodes with low periods for the remaining segments.
3. If this node has a transmission period low enough to carry more than the segment to be inserted, it is split into several parts of equal period, leaving the remaining nodes for other segments. In the tree representation, these splits are represented by non-leaf nodes, while the inserted segments are the leaf-nodes.

Figure 3.52 shows an example for four channels and figure 3.53 shows the tree representation of the used transmission schedule. This transmission scheme will be reviewed extensively in chapter 4 as its full potential has not been tapped yet.

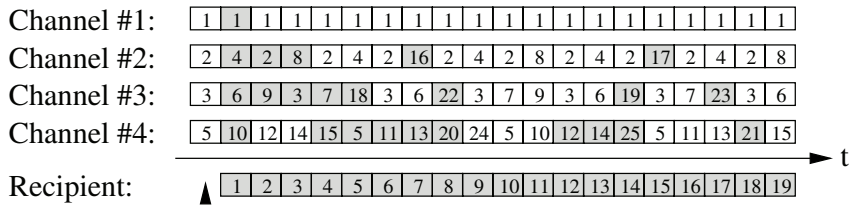


Figure 3.52: Greedy Broadcasting/Recursive Frequency Splitting

3.7.9 Fuzzycast

Fuzzycast [JWX02] is based on the Harmonic Equal-Bandwidth Broadcasting scheme, but removes its high bandwidth requirements using a heuristic algorithm: When it encounters a slot interval which contains too many segments, it decides to move the highest segment(s) to another slot interval, either forward or backward in time. Unfortunately, moving segments has a drawback: If segments are transmitted earlier this way, the bandwidth is increased and the transmission scheme looses some efficiency. If segments are transmitted later, the playback delay has to be increased, which lowers the efficiency, too.

A major disadvantage of the Fuzzycast transmission scheme is that its bandwidth requirement or playback delay cannot be guaranteed for a permanent transmission: As the moving process is heuristic, there is no way to predict how much free slots are available in the neighborhood of a specific segment position (assuming that the granted bandwidth is not so high that nearly no

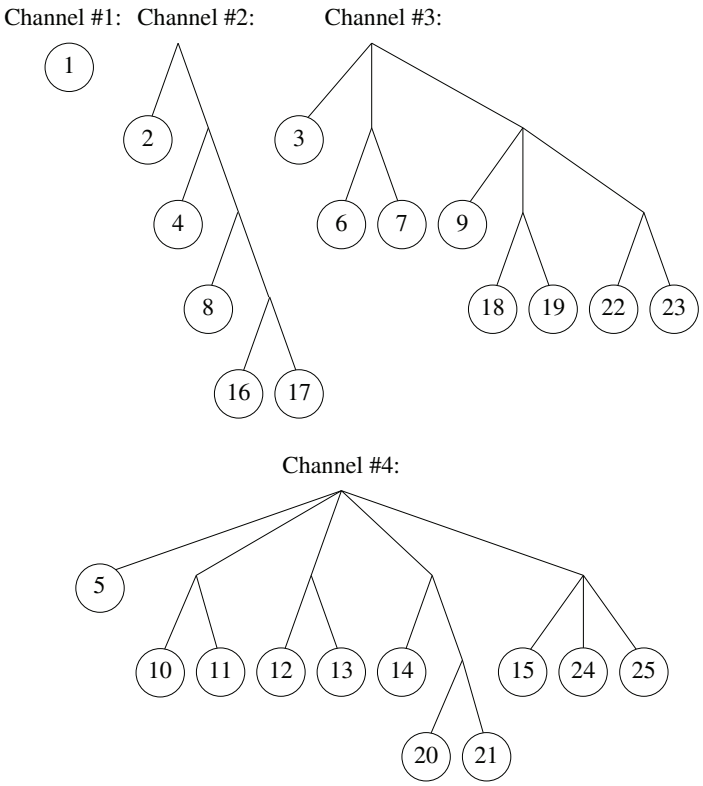


Figure 3.53: Tree representation of a Greedy Broadcasting/Recursive Frequency Splitting transmission scheme

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.42: Classification for Greedy Broadcasting/Recursive Frequency Splitting transmissions

moving is required). As a consequence, the Fuzzycast scheme has an unpredictable bandwidth consumption or playback delay because the bandwidth can only be examined if the playback is

simulated until end. Nevertheless, the idea of moving segments is revised later in section 4.13 in the context of efficient terminations of ongoing transmissions.

One interesting improvement to reduce the number of segment moves has been mentioned by the authors of the Fuzzycast Broadcasting scheme: If several transmissions are grouped together, the probability to find a free slot at the destined position or in its near neighborhood is much higher than for a single transmission. Therefore, the sender bandwidth can be reduced by this grouping for each of the transmissions. The not mentioned drawback of this grouping is that the receiver bandwidth must be higher when one transmission uses free slots of another one. This idea is resumed in section 4.11.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.43: Classification for Fuzzycast Broadcasting transmissions

3.7.10 Dual Broadcasting

The Dual Broadcasting scheme [PCL99a] is a transmission scheme which supports receiver systems without and with local storage. To support both systems at the same time, a transmission scheme which requires no storage (e. g. Round-Robin transmissions or Staggered Broadcasting) and one which uses storage at the recipient systems (e. g. Fast Broadcasting or Pagoda Broadcasting) have to be used at the same time.

But a simple combination of these two transmission schemes is a waste of bandwidth as the recipient systems which are equipped with storage could benefit from the no-storage transmissions. The authors of the Dual Broadcasting scheme therefore developed a transmission scheme where the storage-based transmission relies on the no-storage transmission.

Figure 3.54 shows an example for a Dual Broadcasting transmission. Recipient #1 in this

example does not need any storage whild recipient #2 joined the transmission at a time where storage is required.

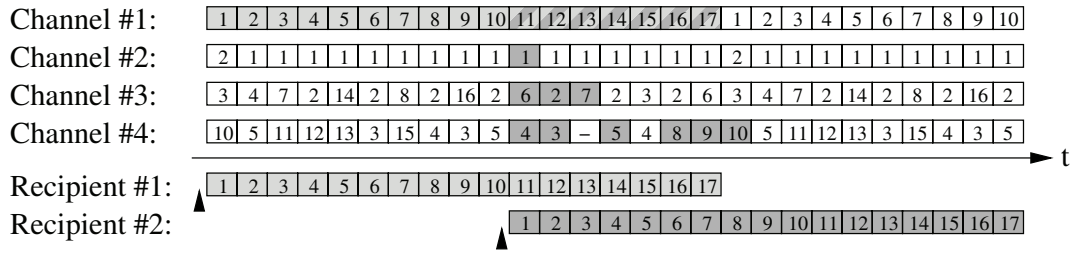


Figure 3.54: Dual Broadcasting

Unfortunately, the Dual Broadcasting scheme is not a pure frequency-based transmission scheme as it transmits the segments at non-constant frequency. As a consequence, transmission schemes according to this scheme cannot be created as simple as the other proposed schemes. Even the authors of the Dual Broadcasting scheme only presented two schedules for this scheme (one for three and one for four channels) without providing an algorithm for general construction.

Another thing to note here is that the Fast Broadcasting scheme (see section 3.7.2) already supports a reception without storage with a maximum playback delay of one media stream duration, but the Dual Broadcasting scheme outperforms this.

Technical classification	**Functional classification**
Communication type: One-to-many **Back-channel:** Not required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** **!** None or partial media stream buffering **Receiver storage seeks:** **!** None or single position per slot	**Media scheduling:** Pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable
Comment: **!** Different service for receiver systems with and without storage	

Table 3.44: Classification for Dual Broadcasting transmissions

3.7.11 Comparison of Frequency-Based Segmenting Transmission Schemes

Comparing the frequency-based segmenting transmission schemes with each other is a much simpler task than comparing size-based or bandwidth-based schemes as the differences between the various frequency-based schemes are much less significant. Besides of the layout of the segments on the channels, the transmission schemes differ in the following points:

- For the Fixed Delay Pagoda Broadcasting scheme and the Harmonic Equal-Bandwidth Broadcasting scheme, the playback delay can be defined independent of the segment size.
- The Fixed Delay Pagoda Broadcasting scheme supports receiver systems with limited receiver bandwidth or storage size.
- The Fast Broadcasting scheme and the Dual Broadcasting scheme support receiver systems without storage.
- The Seamless Fast Broadcasting scheme and the Variable Bandwidth Broadcasting scheme support seamless bandwidth changes for ongoing transmissions.

Similar to the comparison charts for size- and bandwidth-based transmission schemes, it is assumed that the receiver systems provide an unlimited amount of bandwidth and storage to receive the transmission. Additionally, the ability to change the playback delay dynamically is ignored here.

Figure 3.55 shows the efficiency of the frequency-based transmission schemes if the slot duration equals the playback delay. As expected, the (practically unusable) Harmonic Equal-Bandwidth Broadcasting scheme provides the highest efficiency of all these transmission schemes, nearby followed by the Greedy Broadcasting/Recursive Frequency Splitting scheme. But the Harmonic Equal-Bandwidth Broadcasting scheme is still far away from the theoretical lower bound (i. e. from an efficiency $\eta = 1$) if the segment size is bound to the playback delay, esp. for long playback delays.

If the transmission scheme supports to select the playback delay independent from the segment size, the segment size can be decreased while keeping the playback delay constant. Figure 3.56 shows the result of this procedure for the two transmission schemes which support this functionality. In this case, both transmission schemes operate much better, the Harmonic Equal-Bandwidth Broadcasting scheme even operates at 100 %.

The Dual Broadcasting scheme has been omitted from these comparisons as no algorithm for generation of large Dual Broadcasting transmission schedules is available.

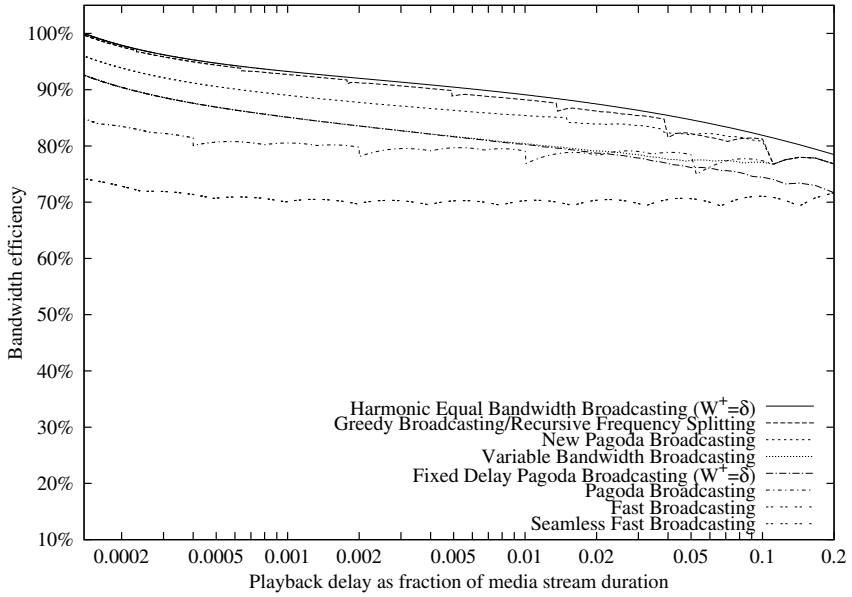


Figure 3.55: Efficiency of frequency-based schemes if the playback delay equals the slot interval

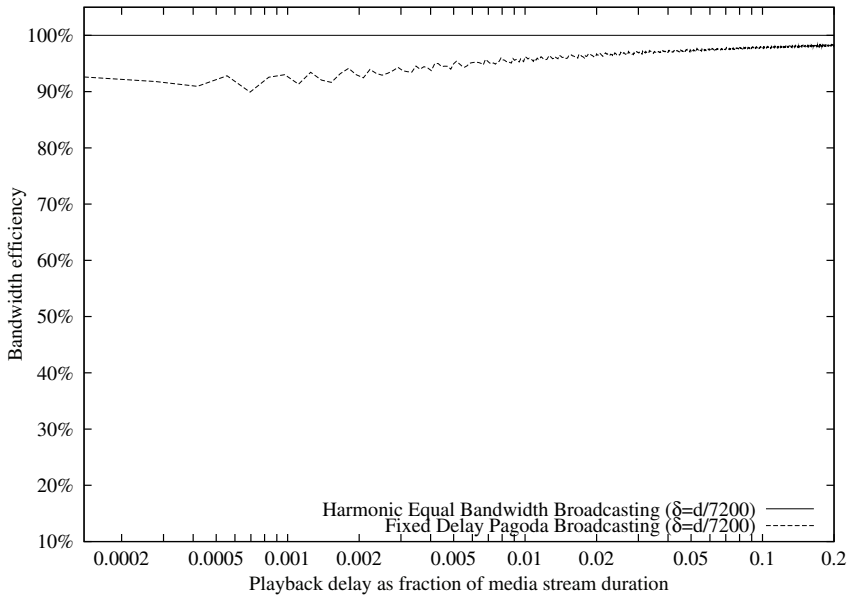


Figure 3.56: Efficiency of frequency-based schemes for which the slot interval can be decreased independent of the playback delay

3.8 Reactive-Pro-Active-Hybrid Transmission Schemes

The above presented transmission schemes are either reactive or pro-active by design. In this section, **reactive-pro-active-hybrid transmission schemes** are presented which use combinations of reactive and pro-active design patterns.

One way to combine reactive and pro-active transmission schemes is to send part of the media stream using a reactive transmission scheme and the remainder using a pro-active scheme. The reason for combining a reactive and a pro-active transmission scheme this way is that the most bandwidth of a media transmission is required for the transmission of the first segments which are received only by a few recipients, thus the most bandwidth is wasted by the transmission of the first part of the media stream if only few recipients are listening to the transmission. If a reactive transmission scheme is used for the first part of the media stream, this bandwidth can be saved, providing at the same time a short playback delay (or even immediate access, depending only on the reactive scheme).

Another way to apply reactive behavior to a pro-active transmission scheme is to use playback request to evaluate which segment transmissions are not needed by any recipient and omitting the transmission of these segments in a pro-active transmission schedule.

3.8.1 Unified Video-on-Demand Broadcasting

The Unified Video-on-Demand Broadcasting scheme [Lee99] is a combination of the Staggered Broadcasting scheme (see section 3.4.2 with Point-to-Point transmissions (see section 3.3.1: As the playback delay of the Staggered Broadcasting scheme is very high, the Unified Video-on-Demand Broadcasting scheme uses Point-to-Point transmissions to transmit the beginning of the media stream to recipients which called in at an ongoing transmission. That way, the Unified Video-on-Demand Broadcasting scheme provides immediate access without the need to transmit the whole media stream to the recipients.

3.8.2 Batching Unified Video-on-Demand Broadcasting

Similar to the Unified Video-on-Demand Broadcasting scheme of the last subsection, the Batching Unified Video-on-Demand Broadcasting scheme [LL00] benefits from the Staggered Broadcasting scheme. But instead of using Point-to-Point transmissions for the beginning of the media streams, this scheme broadcasts the beginning at times determined by a Batching scheme (see section 3.3.2). Consequently, the Batching Unified Video-on-Demand Broadcasting scheme does not offer immediate access any more but still reduces the playback delay of the Staggered Broadcasting scheme.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per channel **Receiver bandwidth:** ! Twice the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per channel	**Media scheduling:** ! Combined reactive/pro-active media-on-demand **Playback control:** ! Fixed jump playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Live streaming capable

Table 3.45: Classification for Unified Video-on-Demand Broadcasting transmissions

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per channel **Receiver bandwidth:** Twice the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per channel	**Media scheduling:** ! Combined reactive/pro-active media-on-demand **Playback control:** ! Fixed jump playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** ! Live streaming capable

Table 3.46: Classification for Batching Unified Video-on-Demand Broadcasting transmissions

3.8.3 Reactive Broadcasting

Similar to the above two transmission schemes, the Reactive Broadcasting scheme [PCL00a] is actually a combination of two transmission schemes: A reactive transmission is used for the transmission of the first segment of the media stream and a pro-active transmission scheme is used for the remaining segments. But instead of using Staggered Broadcasting combined with Point-to-Point Transmissions or Batching, the authors of [PCL00a] suggested to use Tapping (see section 3.3.5) as reactive scheme and a Pagoda-like scheme (see section 3.7.4) as pro-active scheme.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** ! Combined reactive/pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.47: Classification for Reactive Broadcasting transmissions

3.8.4 Dynamic Skyscraper Broadcasting

The Dynamic Skyscraper Broadcasting scheme [EV98] makes two modifications to the Skyscraper Broadcasting scheme presented in section 3.5.3. Firstly, it divides subsequent segment transmissions up to a specific length to **transmission clusters** and allows that transmission clusters can be dropped if no recipient is listening to the appropriate transmission. As they suggest using the same transmission cluster sizes for all media streams, this spare bandwidth can simply be reused for other media stream transmissions. Figure 3.57 illustrates the use of transmission clusters.

Secondly, they proposed three new broadcasting series:

$$\begin{aligned} A: &\quad 1,2,2,4,4,8,8,16,16,32,32,64,64,128,128,\ldots \\ B: &\quad 1,2,2,6,6,12,12,24,24,48,48,96,96,192,192,\ldots \\ C: &\quad 1,2,2,6,6,12,12,36,36,72,72,216,216,432,432,\ldots \end{aligned} \tag{3.38}$$

These broadcasting series avoid conflicts and holes in transmission clusters which would occur if the original broadcasting series are used. Broadcasting series B and C additionally require that the receiver system is able to receive three channels at a time, while for sequence A two channels still suffice.

3.8.5 Partitioned Dynamic Skyscraper Broadcasting

The Partitioned Dynamic Skyscraper Broadcasting scheme [EFV99] is a variant of the Dynamic Skyscraper Broadcasting scheme presented above. Instead of using a single sender, the Partitioned Dynamic Skyscraper Broadcasting scheme assumes a hierarchical setup, consisting of a global

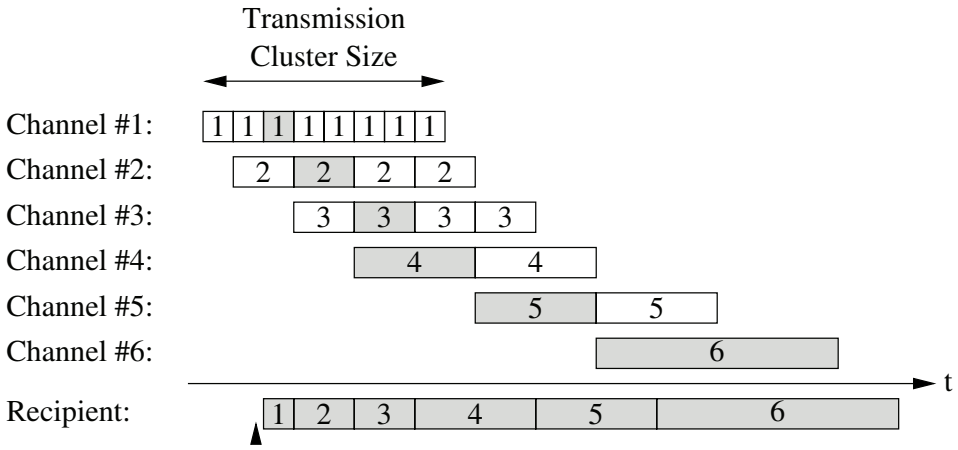


Figure 3.57: Dynamic Skyscraper Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** ! Twice or three times the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** ! Combined reactive/pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.48: Classification for Dynamic Skyscraper Broadcasting transmissions

sender system and a set of regional sender systems. Therefore, the Partitioned Dynamic Skyscraper Broadcasting scheme requires that the receiver systems are able to receive data simultaneously from the global sender system and from its next regional sender system (doubling the bandwidth requirement for the clients and assuming a network which benefits from regional sender systems).

The Partitioned Dynamic Skyscraper Broadcasting scheme uses this setup by transmitting the first segments of a media stream using regional sender systems and sending the latter segments from the global sender system. This reduces the global network utilization as the most bandwidth is required for the first segments which are only transmitted regionally. Figure 3.58 shows an example for a Partitioned Dynamic Skyscraper Broadcasting transmission.

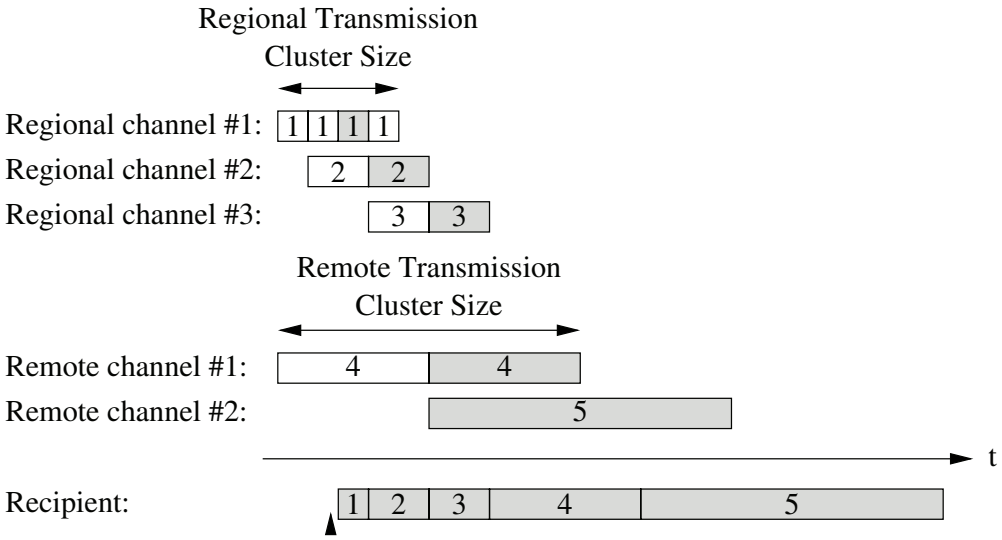


Figure 3.58: Partitioned Dynamic Skyscraper Broadcasting

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** ! Four to six times the media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** ! Combined reactive/pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.49: Classification for Partitioned Dynamic Skyscraper Broadcasting transmissions

3.8.6 Universal Broadcasting

The Universal Broadcasting scheme [PCL00b] is based on the Fast Broadcasting scheme (see section 3.7.2), but it uses two modifications to save bandwidth if only few recipients are requesting the media:

1. Only required segments are sent at the latest possible time. To ascertain which segments are required, this transmission scheme is based on requests from recipient systems to determine at which slot intervals recipients have started a playback.

2. The segments of each channel may be shifted if no ongoing playback is interfered thereby.

Figure 3.59 shows an example for a Universal Broadcasting transmission with two recipients.

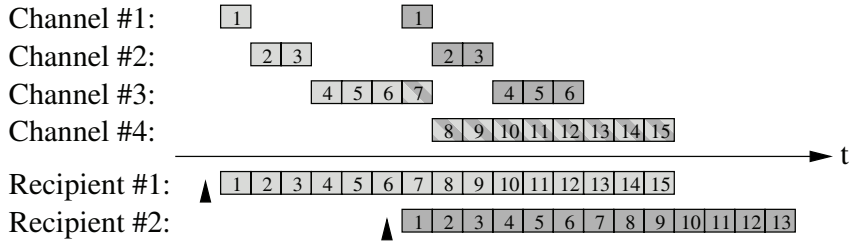


Figure 3.59: Universal Broadcasting

Using these two modifications, the Universal Broadcasting scheme needs no bandwidth if no playbacks are active and it only needs one transmission channel if only one request is active. As the authors have examined, the Universal Broadcasting scheme needs less bandwidth than the (pro-active) New Pagoda transmission scheme for up to 55 requests per hour if a two hour media stream is transmitted in 127 segments. For higher request rates, the Universal Broadcasting scheme converges to the Fast Broadcasting scheme.

Another thing to note here is that it is possible to apply the first of the aforementioned modifications — suppression of segment transmissions which are not needed by any receiver system — to any segmenting transmission scheme. The second modification — shifting of segments in channels — can only by applied to transmission schemes which transmit all segments of a channel with the same frequency, thus it can only be applied to the Fast Broadcasting scheme of the presented frequency-based group of transmission schemes. (By theory, it would also be possible to apply it to transmission schemes of e. g. the size-based group as they only transmit a single segment per channel, but as their efficiency is below that of Fast Broadcasting this is not studied here in more detail.)

In section 4.12, the idea behind this approach is reviewed in a generalized version.

3.8.7 Channel-Based Heuristic Distribution

The Channel-based Heuristic Distribution scheme [ZP02] is the result of a further development of the Universal Broadcasting scheme. It produces similar results as the Universal Broadcasting scheme, but it has been extended to support increased playback delays. Figure 3.60 shows an example for this transmission scheme with a playback delay of two slot intervals.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per slot	**Media scheduling:** ! Combined reactive/pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.50: Classification for Universal Broadcasting transmissions

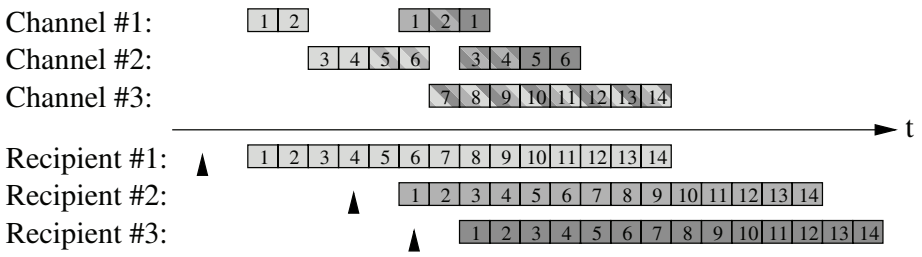


Figure 3.60: Channel-based Heuristic Distribution

3.8.8 Comparison of Reactive-Pro-Active-Hybrid Transmission Schemes

This section provided mainly two different approaches how pro-active transmission schemes can be used in a reactive environment. Firstly, two transmission schemes can be combined: A reactive transmission scheme can be used for the beginning of the media stream and a pro-active transmission scheme for the remainder. This idea has been used by the Unified Video-on-Demand Broadcasting scheme, the Batching Unified Video-on-Demand Broadcasting scheme, the Reactive Broadcasting scheme and the Dual Broadcasting scheme.

Alternatively, it is possible to use solely a pro-active transmission scheme, omitting transmissions which are not needed by any recipient. This approach is followed in the Dynamic Skyscraper Broadcasting scheme, the Partitioned Dynamic Skyscraper Broadcasting scheme, the Universal Broadcasting scheme and the Channel-based Heuristic Broadcasting scheme.

Additional ideas which have been proposed in these transmission schemes include combining segments to transmission clusters (to manage unused bandwidth more simple), dividing the

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** Required **Media-on-demand topology:** Star **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per channel **Receiver bandwidth:** Multiple of media stream bit rate **Receiver storage size:** Partial media stream buffering **Receiver storage seeks:** Single position per channel	**Media scheduling:** ! Combined reactive/pro-active media-on-demand **Playback control:** No playback control **Content navigation:** No non-continuous navigation **Dynamic schedule changes:** No dynamic scheduling change support **Dynamic content changes:** No dynamic content support **Variable-bit-rate media support:** No dynamic content support **Live streaming capability:** Not live streaming capable

Table 3.51: Classification for Channel-based Heuristic Broadcasting transmissions

transmission into a global and several regional parts (to reduce global bandwidth consumption by frequent transmissions of media stream beginnings) and shifting of segments in channels (to lower bandwidth consumption).

For a comparison of hybrid transmission schemes, the scheme which provides immediate service (the Reactive Broadcasting scheme) and the schemes which require a playback delay after requests (the Unified Video-on-Demand Broadcasting, the Batching Unified Video-on-Demand Broadcasting, the Dynamic Skyscraper Broadcasting, the Partitioned Dynamic Skyscraper Broadcasting, the Universal Broadcasting and the Channel-based Heuristic Distribution scheme) have to be examined separately, similar to the comparison of reactive transmission schemes in section 3.3.9.

The Reactive Broadcasting scheme, the only proposed hybrid transmission scheme with immediate service, shows a poor efficiency for very low request rates as shown in figure 3.61, a result of the permanent transmission of the latter part of the media stream. For higher request rates, the efficiency stays nearly constant around the one of the used pro-active Pagoda Broadcasting scheme.

Of the transmission schemes which require a playback delay, the Channel-based Heuristic Distribution scheme provides the best efficiency for all request rates. Figure 3.62 shows the efficiency of these transmission schemes with a playback delay of $\frac{1}{120}$ of the media stream duration.

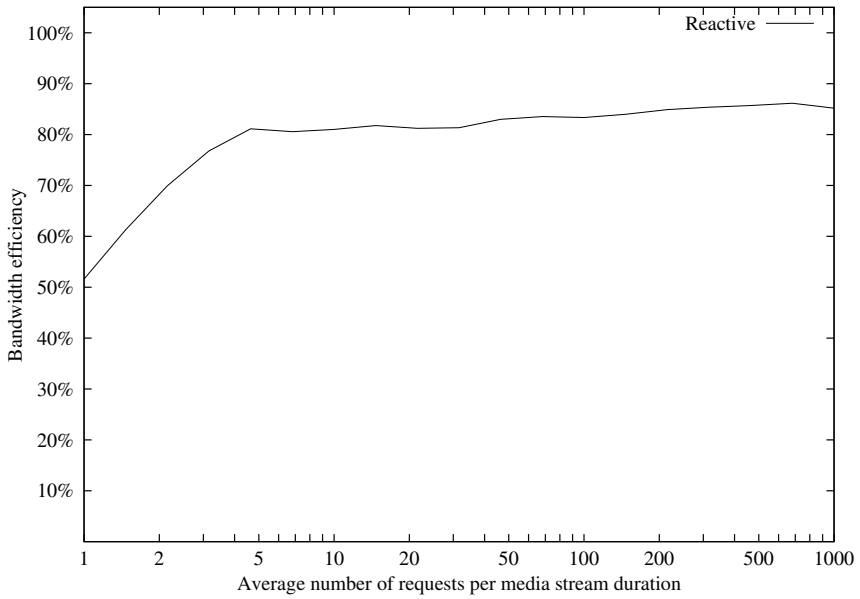


Figure 3.61: Efficiency of hybrid transmission schemes with immediate service

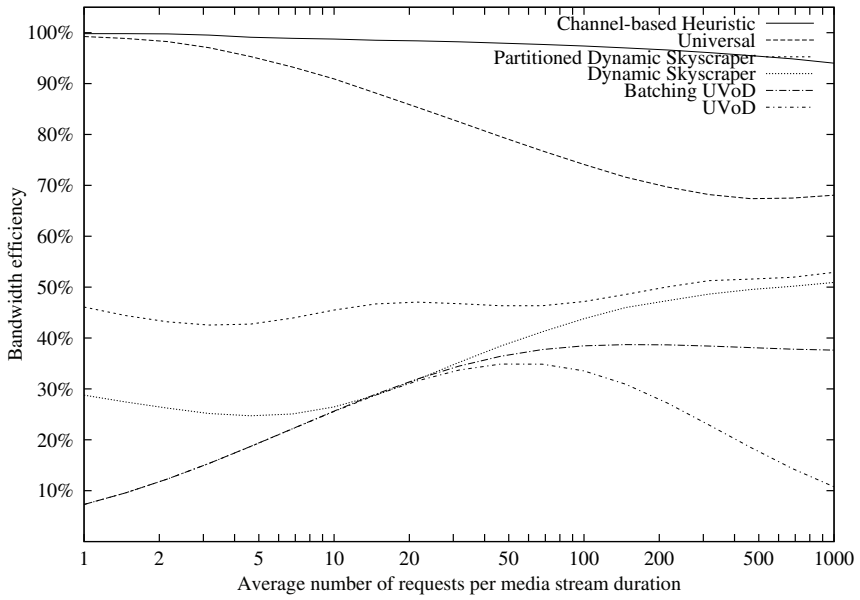


Figure 3.62: Comparison of efficiency of hybrid transmission schemes with delayed service

3.9 Summary

In this chapter, many different approaches for transmission schemes have been presented. Generally, they can be divided into three groups: The reactive transmission schemes which rely on

playback requests from the receiver systems, the pro-active transmission schemes which transmit the media streams independent of any requests and hybrid schemes which combine reactive and pro-active approaches.

The first and third group can be further divided into schemes which deliver a media stream immediately on request and schemes which use a playback delay to reduce bandwidth requirements. Of the schemes which provide immediate service, the ERMT Merging scheme provides the highest efficiency for few to medium requested streams but also requires a lot of calculation for each playback request. The CT Merging and the ϕ-Dyadic Merging schemes are not yet as efficient as the ERMT Merging scheme but are much more simple to implement. Only for high request rates (more than approximately 150 requests per media stream duration), the Reactive Broadcasting scheme performs better than the ERMT Merging scheme. The reason therefore is that the Reactive Broadcasting scheme requires a permanent (pro-active) transmission of the latter part of the media stream which seems to be profitable only if the media stream is requested often enough.

Of the reactive and hybrid group of transmission schemes which require a playback delay, the Channel-based Heuristic Broadcasting scheme performs best.

The segmenting pro-active transmission schemes can be subdivided into three groups, based on the mechanism which is used to reduce the sender bandwidth consumption for a transmission: Size-based transmission schemes use increasing segment sizes at equal bandwidth, bandwidth-based schemes transmit segments of equal size in channels of decreasing bandwidth and frequency-based schemes send segments of equal size on channels of equal bandwidth at decreasing frequency. Although all this three groups reach the same goal (transmitting the latter parts of a media stream less frequently than the former parts), they perform very different:

The size-based group is probably the simplest but at the same time the least efficient one: The reason for this efficiency loss is that the later segments of a media stream become very large which prevents a fine-grained scheduling.

The bandwidth-based schemes provide nearly ideal theoretical conditions: The efficiency can be increased to reach the theoretical limit and many enhancements (e. g. adaption to variable-bit-rate media streams, support for receiver system constraints and seamless bandwidth changes) have been proposed for these schemes. Unfortunately, these transmission schemes perform very poor in practical environments for several reasons:

- If many segments are used (which is necessary to achieve a high efficiency), the bandwidth for the later segment transmissions becomes very low. Effectively, the media bit rate of the last segments is so low that it is practically impossible to transmit data at these bit rates[17].

[17] If a two-hour media stream of 2 Mbit/s is transmitted with a playback delay of 60 seconds and a segment size equal to one second, the last segment would have to be sent at approximately 275.5 bit/s.

Therefore, many channels have to be aggregated into channels of higher bandwidth which imposes several additional problems (e. g. handling of fractions in bit rates, identification of aggregation boundaries for splitting into subsegments at the receiver side and increased damage in case of data losses if one aggregated packet is lost). Additionally, every segment is transmitted at a different bit-rate which makes any calculations more complex.

- The bandwidth-based schemes are based on the fact that part of each segment is transmitted at every slot interval. This means that even a short transmission failure causes holes in all segments if it cannot be repaired. For example for video streams, a loss of all data of one slot interval of a transmission would cause decoding problems in every segment of the whole remaining media stream playback.
- Another consequence of the idea that part of each segment is transmitted at every slot interval is that both the sender and the receiver systems have to access all segments at each slot interval. This increases the hardware requirements for receiver and sender systems dramatically: To save the data of one slot interval to the corresponding segments on a hard disk, the disk must support high-speed random access which makes these systems very expensive[18]. Advanced caching strategies may be used to reduce the number of disk accesses by increasing the needed amount of memory.

The frequency-based schemes do not have any of these problems: All segments have the same size and are transmitted at the same bandwidth, only few segments have to be accessed each slot interval and data losses only affect a few segments at a time. When comparing the frequency-based schemes, the Fixed-Delay Pagoda Broadcasting scheme provides the highest efficiency, nearby followed by the Greedy Broadcasting/Recursive Frequency Splitting scheme. Unfortunately, these schemes do not provide the same efficiency as the bandwidth-based schemes.

Even some of the less efficient transmission schemes qualify for a more precise examination because some of them provide worthwhile enhancements, e. g. dynamic bandwidth changes for ongoing transmissions or support for variable-bit-rate transmissions.

Unfortunately, there is no efficient transmission scheme which provides a high efficiency and which supports all these enhancements at once. For this reason, a new transmission scheme is proposed in the next chapter which benefits from several perceptions of this chapter and which incorporates many approaches of the examined transmission schemes.

[18] For example, during the first slot interval of a two hour media stream with one second playback delay, parts of 7 200 segments are transmitted. This would require 7 200 disk accesses per second and yields seek times of 138 µs (typical seek times of today's disks are around 9 ms). Saving the received data linearly onto disk and assembling it at playback time does not help either, the disk seeks are required at assembly time this way.

Chapter 4

Generalized Greedy Broadcasting Scheme

MANY of the transmission schemes which have been presented in the last chapter provide excellent solutions to individual cases, but none of these transmission schemes is able to combine all the benefits without any major drawback. Some examples:

- The ERMT Merging scheme performs well for low request rates but has a poor efficiency and an unpredictable maximum bandwidth for high request rates.
- The Tailored Transmission scheme allows arbitrary playback delays independent of the segment size and is capable of transmitting variable-bit-rate media streams, but as a member of the bandwidth-based class of transmission schemes, it has high receiver storage throughput requirements and reacts displeasingly to data losses.
- The Greedy Broadcasting/Recursive Frequency Splitting scheme shows a good efficiency in the frequency-based class but does not support to select a playback delay independently from the segment size or to transmit variable-bit-rate media streams efficiently.
- The Fixed Delay Pagoda Broadcasting scheme has a very good efficiency and allows decoupling the segment size from the playback delay but does not support variable-bit-rate media stream transmissions efficiently, too.

Due to these problems, a new transmission scheme is proposed in this chapter. This new scheme, the Generalized Greedy Broadcasting scheme, is a successor of the Greedy Broadcasting scheme/Recursive Frequency Splitting scheme and has been designed to fulfill most of the requirements at once by combining the approaches from several transmission schemes, superseding the existing schemes for mostly any setup.

4.1 Origin of Generalized Greedy Broadcasting

To justify the development of a new transmission scheme, it is important to have a look at the requirements first. These requirements also allow classifying the new scheme in the row of existing schemes as well as referencing solutions for some of the requirements from other schemes.

4.1.1 Requirements

The above proposed transmission schemes have been divided into two major groups: reactive and pro-active schemes. While the reactive transmission schemes provide a good efficiency for few requests, their efficiency decreases when they are used for highly requested media streams. Worse, the maximum server bandwidth of these schemes is not bound to an upper limit without loosing a guaranteed playback delay. Therefore, the only way to provide a predictable and efficient service for any number of recipient systems is to use a pro-active transmission scheme. Additionally, this provides the highest utilizability for the transmission scheme as it will not require a back-channel.

As described in the previous chapter, the group of pro-active transmission schemes can be divided into three subgroups, the size-based, the bandwidth-based and the frequency-based schemes. (The non-segmenting schemes are ignored because of their poor efficiency.) While size-based transmission schemes use a very simple design, they cannot reach the same efficiency as the bandwidth-based or frequency-based ones for short playback delays. The bandwidth-based transmission schemes provide a very high efficiency and flexibility, but they have major disadvantages as described in the section 3.9. Only the frequency-based transmission schemes do not suffer from any of the above problems: They have a good efficiency, do not require much disk seeks (only one segment per channel is received every slot interval) and are comparatively robust to data losses. The only disadvantage of frequency-based transmission schemes is that an efficient transmission schedule has to be found for a particular transmission.

Unfortunately, the generation of an optimal transmission schedule is impossible in an acceptable amount of time: The algorithmic problem behind this is related to the bin packing problem [HSR98a] and has been proven to be NP complete [BNBNS98]. Fortunately, it is possible to create nearly optimal solutions using heuristic algorithms.

Besides high bandwidth efficiency for any setup, the new transmission scheme should provide support for:

- playback delays independent from the slot interval,
- (optional) partial preloading,
- live transmissions,
- dynamic change of transmission parameters, e. g. change of playback delay for ongoing transmissions,

- seamless media changes,
- receiver systems with limited receiver bandwidth or storage capacity,
- a channel bandwidth independent of the media bit rate,
- adoption to variable-bit-rate media streams,
- pro-active environments,
- bandwidth savings by evaluating requests in reactive environments.

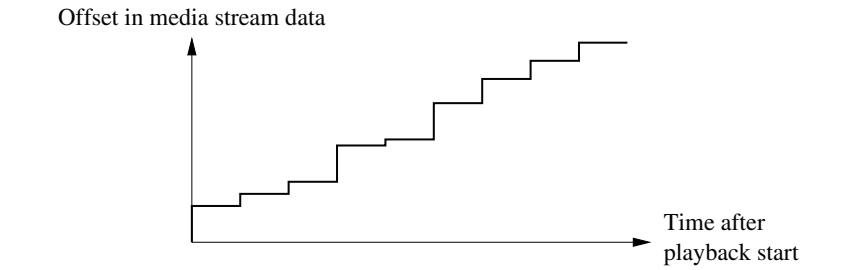
4.1.2 Playback Function, Transmission Function and Cumulative Bandwidth Function

To formalize a playback including a playback delay, partial preloading, breaks and possibly variable-bit-rate transmissions (as described in the next sections), a function is useful which maps each point in time after the playback request to an offset in the media stream data. As this **playback function** is the result of the application of several distinct tasks (insertion of breaks, preloading part of the media stream, applying a playback delay) to a media stream (constant- or variable-bit-rate encoded), it is defined in the following sections in exactly this way: by applying **transmission functions** (each describing a single task which is performed to the transmission) to a **cumulative bandwidth function** (which describes the media stream bandwidth variations).

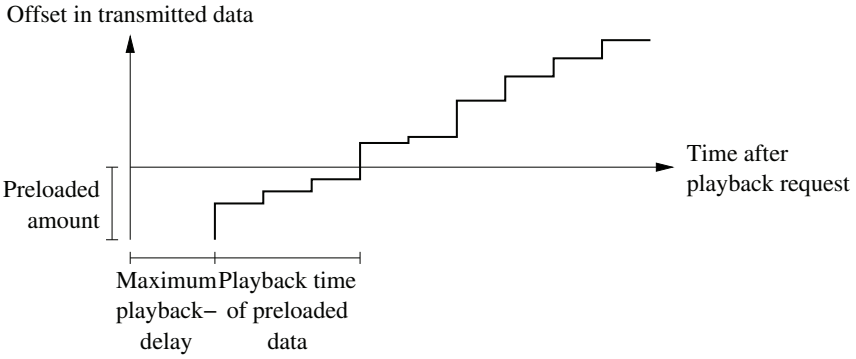
The cumulative bandwidth function constitutes the base of the playback function: It counts the amount of data which has to be received by the receiver system at a given time after start of the playback[1]. That is, the cumulative bandwidth function increases every time when the receiver system needs data for decoding and stays constant during the remaining time. (Depending on the media encoding, decoding and displaying of data can happen at different times: For example for displaying a B-frame of an MPEG encoded media stream, the next succeeding P- or I-frame must be decoded in advance.)

Starting with this cumulative bandwidth function, several different transmission functions can be applied to obtain the resulting playback function which characterizes the media data consumption at the receiver system. Figure 4.1a shows an example for the beginning of a cumulative bandwidth function for a variable-bit-rate media stream and figure 4.1b gives an example of a playback function for the same media stream after application of a playback delay and partial preloading.

[1] When variable-bit-rate media stream transmissions are examined, one could assume a (direct) bandwidth function is more simple to obtain, but the reverse is true: As the receiver system needs the media data at singular points (i. e. when it needs to decode a frame in case of video), the bandwidth at these points is theoretically infinitely high for an infinitely short duration, making any meaningful definition of this function impossible. Defining a "cumulative" bandwidth function instead (which can be looked at as the antiderivative of the bandwidth function) is no problem as the singular points convert into points of discontinuity, i. e. the cumulative bandwidth function has a staircase step with a finite increase at each such point.



(a) Example for a cumulative bandwidth function



(b) Example for a playback function

Figure 4.1: Cumulative bandwidth and playback function

In general, the cumulative bandwidth function can be defined by

$$f(t) = \sum_{\substack{1 \le i \le n \\ \tau_i \le t}} \sigma_i \tag{4.1}$$

where τ_i is the decode time of segment #i after the beginning of the playback, σ_i is the size of segment #i and n is the number of segments of the media stream. For constant-bit-rate media streams, this formula is equivalent to

$$f(t) = \left\lfloor \frac{t}{\delta} + 1 \right\rfloor \cdot \sigma \tag{4.2}$$

where δ is the duration of the slot interval and σ is the size of the segments. Transmission functions for application of a playback delay, partial preloading and insertion of breaks are given in sections 4.4, 4.6 and 4.7, respectively

4.1.3 Tree-Based Representation for Transmission Schemes

One of the best heuristic algorithms for the generation of transmission schedules has been proposed by the authors of the Greedy Broadcasting scheme/Recursive Frequency Splitting scheme.

The authors of these two transmission schemes used a tree-based representation for the transmission schedules[2] which has already been described shortly in section 3.7.2 and is defined in more detail in the following:

Accurately treated, each tree-based representation is an ordered list of trees where the order of child nodes of each node is significant[3]. In these trees, each segment is attached to a different leaf node.

To extract the tabular representation from a tree-based one, the following algorithm can be used: Each node N in the trees of the tree-based representation is assigned a period Π_N, a phase shift Φ_N and a channel K_N using the following set of rules:

1. If N is a node of the i-th tree (starting with one), the channel of N is $K_N = i$.

2a. If N is a root node, its period is $\Pi_N = 1$.

2b. If N is not a root node, let P be its parent node and c be the number of child nodes of P (i. e. the number of sibling nodes of N including N itself). Then the period of N is $\Pi_N = c \cdot \Pi_P$.

3a. If N is a root node, its phase shift is $\Phi_N = 0$.

3b. If N is not a root node, let P be its parent node and i count what number N is of P in the line of children (starting with zero). Then the phase shift of N is $\Phi_N = \Phi_P + i \cdot \Pi_P$.

Using these rules, the period Π_N, phase shift Φ_N and channel K_N of each leaf node N specify the period π_i, phase shift ϕ_i and channel κ_i for the segment $\#i$ which is attached to the leaf node. Figure 4.2 shows an example for a tabular and a tree-based representation together with the period and phase shift values of the nodes and figure 4.3 illustrates by an example how these leaf nodes map to the time slots of a transmission scheme. The complete algorithm for this conversion is given in appendix A.1.

The above description of the algorithm guarantees that

- Each node of the trees is assigned exactly to one channel,
- the bandwidth of a node is either completely available to the node itself (if the node is a leaf node) or divided equally to its child nodes by time-multiplexing (if the node is not a leaf node), and

[2]The authors of the Recursive Frequency Splitting scheme used a recursive list representation as notation which is equivalent to the tree notation of the Greedy Broadcasting scheme.

[3]If the order of the trees and the order of the child nodes within the trees is changed, valid transmission schemes with the same average and maximum bandwidth are obtained, so it may appear as if the order is insignificant. But to obtain a unambiguous mapping from a tree-based to a tabular representation, the order is significant. Another ramification of the order of the nodes in the trees is shown in sections 4.12 and 4.19 when a method for reducing the bandwidth at the beginning of a transmission and an algorithm for determining the memory requirements at the receiver system side are proposed.

Tabular representation:

Segment No.	Channel	Period	Phase Shift
A	1	3	0
B	1	6	1
C	1	6	4
D	1	6	2
E	1	6	5

Tree−based representation:

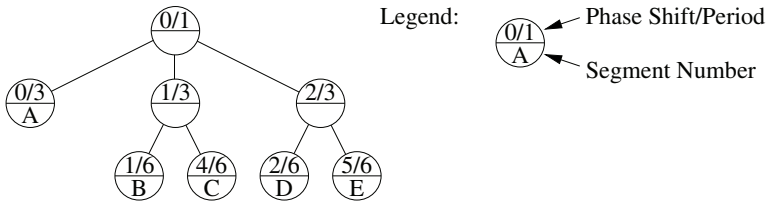


Figure 4.2: Example for the a transmission scheme in tabular and in tree representation

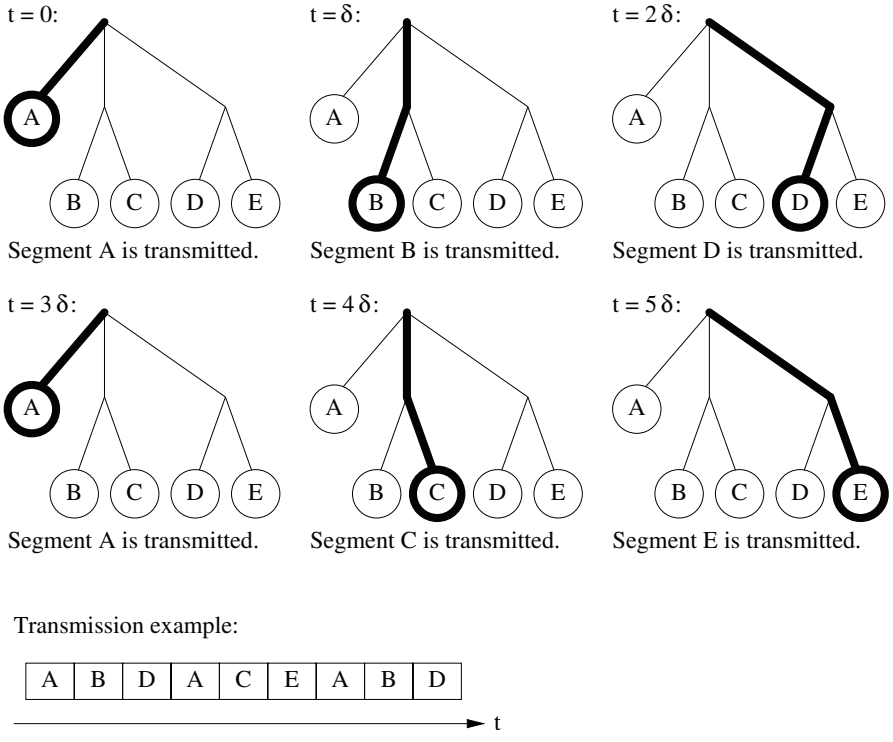


Figure 4.3: Example for the transmission order for a transmission scheme in tree representation

- the phase shift of each multiplexing differs in such a way that the child nodes are transmitted round-robin and only at the times which are specified by the period and phase shift of their parent node (as $\Phi_N \bmod \Pi_P = \Phi_P$ and $\Phi_N \operatorname{div} \Pi_P = j$ if N is the j-th child node of P).

This means that the leaf nodes of each tree identify one-to-one the time slots of a channel.

Conversely, this means that a transmission scheme can be created by assigning all segments to leaf nodes of transmission trees in such a way that the period of the node is less or equal to the maximum period of the assigned segment, which is the basic principle of the functioning of the Greedy Broadcasting/Recursive Frequency Splitting scheme.

In the opposite way, to extract a tree-based representation from a tabular one, the following algorithm can be used:

1. Start with as many empty trees as channels are used.
2. Let i be one of the segments which still has to be inserted into the transmission trees. Search the node P in the transmission trees with the highest node period Π_P, a matching phase shift $\Phi_P = \pi_i \bmod \Pi_P$ and a matching tree number $K_P = \kappa_i$. (This node always exists and is unambiguous; it is the deepest existing node in the current trees where the segment is a child of.)
   (a) If $\Pi_P = \pi_i$, the node for the segment has been found and the segment is attached to the node.
   (b) Otherwise calculate k as the greatest common divisor of all segment periods π_j of the segments $\#j$ where $\phi_j \bmod \Pi_P = \Phi_P$ and $\kappa_j = K_P$, add $\frac{k}{\Pi_P}$ child nodes to P and redo step 2. The search for the greatest common divisor is needed to assure that all child nodes of P can be inserted later.

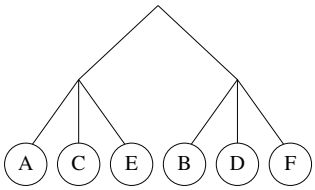
   Redo this step until all segments have been inserted into the trees.

This algorithm, which is given in more detail in appendix A.2, completes the interchangeability of tree-based and tabular representations for frequency-based transmission schemes.

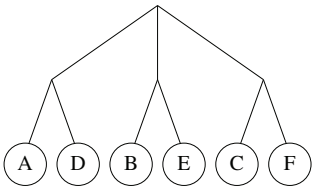
Please note that there may exist several tree representations for one transmission scheme, e. g. figure 4.4 shows three different tree representations for one transmission scheme. A canonical tree representation can only be obtained if additional constraints are defined, e. g. by only allowing nodes with a prime number of child nodes and using the smaller number of child nodes in the parent node when several different tree representations are possible.

4.1.4 Greedy Broadcasting/Recursive Frequency Splitting Scheme

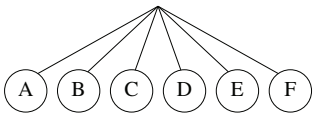
The algorithms of the Greedy Broadcasting scheme and Recursive Frequency Splitting scheme benefit from the tree-based representation by using them to construct transmission schedules in a graph-based way. They start with one empty tree (i. e. root node) per transmission channel and a sorted list of segments which have to be inserted into the transmission schedule. Then they take



(a) One tree representation



(b) Another tree representation for the same scheme



(c) A tree representation combining the two hierarchical divisions into a single one

Segment #	Period	Phase Shift
A	6	0
B	6	1
C	6	2
D	6	3
E	6	4
F	6	5

(d) Tabular representation of the transmission schedule

Figure 4.4: Example for three different tree representations for one and the same transmission schedule

one segment from the list after another in ascending order and insert them at appropriate positions into the trees:

- Primarily, the segments are inserted in such a way that the segment is transmitted as often as necessary but as infrequently as possible, so the transmission schedule is valid but the least bandwidth is wasted.
- Secondly, if the bandwidth usage for several nodes is equal, the node with the highest transmission period is selected, leaving the nodes with low periods for the remaining segments.
- And last but not least, if a selected node has a transmission period low enough to carry more than the segment to be inserted (i. e. if the period of the segment is greater or equal to twice the period of the node), the node is split into several "parts" (represented as child nodes) of equal frequency and only one of these parts is used for the current segment.

More strictly speaking, to insert segment #*i* into the transmission schedule, the algorithms first select all nodes N where $\Pi_N \leq \pi_i^+$, i. e. which have a period low enough to transmit the segment. Then they calculate $\pi_i^+ \bmod \Pi_N$ as a measurement for the bandwidth loss for these nodes (the modulo operation takes the splits of the third item into account) and look for the node with the lowest result, i. e. the lowest bandwidth loss. If the bandwidth losses are equal for several nodes, the node with the highest period Π_N is selected thereof. (If even then several nodes qualify with the same results, the nodes are regarded as equally good and anyone is chosen.) The identified node is then split into $\pi_i^+ \operatorname{div} \Pi_N$ parts if $\pi_i^+ \operatorname{div} \Pi_N > 1$. This splitting is represented in the trees by adding $\pi_i^+ \operatorname{div} \Pi_N$ child nodes to the found node and selecting one of them. The selected node is then marked as allocated for segment #*i* and the algorithm proceeds with the next segment.

Figure 4.5 shows an example for the creation of a transmission schedule for nine segments:

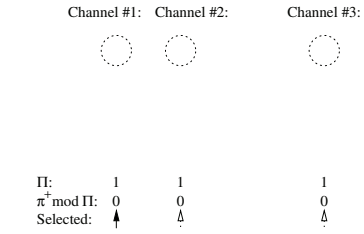
1. In the first step, segment #1 with period $\pi_1^+ = 1$ is inserted. All three root nodes can be used for this segment in the same way, so it does not matter which of them is selected. In this example, the first one has been chosen.
2. Segment #2 with period $\pi_2^+ = 2$ can be inserted at both remaining trees. As the period of the segment is two, only half of the bandwidth of one of the root nodes is needed, so the segment can be inserted as one of two children of one of the root node. All four positions have the same properties, so the first one of the second tree is selected in this example.
3. Segment #3 with period $\pi_3^+ = 3$ can be inserted as other child of the second tree or as one of three children of the root node of the third tree. The bandwidth usage for the second node is higher than for the positions in the third tree, so one of the positions in the third tree is selected.
4. Similarly, segment #4 can be inserted as one of two children of the free node of the second tree or in one of the nodes of the third tree. This time, insertion into the second node is better.
5. The same applies to segment #5.
6. Segments #6 to #9 are inserted into child nodes of the third tree.

One further optimization has been stated in [BNL02]: When a node is split into a non-prime number of child nodes, it is advantageous to perform one split for each prime factor (best in ascending order) instead of one huge split. This way, more large fragments are left which can be used more efficiently for the insertion of the remaining segments. Figure 4.6 shows examples where a node is split into six parts, once leaving five nodes with a period of six unused and the other time leaving one node of period two and two nodes of period six unused.

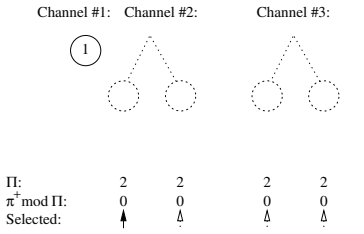
For comparison with later enhancements, figure 4.7 shows the efficiency of the Greedy Broadcasting/Recursive Frequency Splitting scheme.

4.2 Generalization of Greedy Broadcasting/Recursive Frequency Splitting

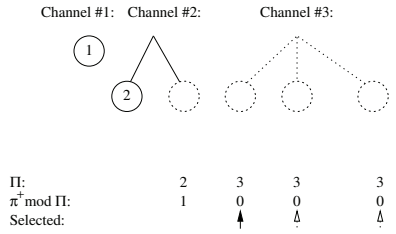
The authors of the Greedy Broadcasting scheme only considered constant-bit-rate media streams where $\pi_i^+ = i$, i. e. they assumed that segments with periods $1, 2, \ldots, n$ are inserted into the trees and they reached a fine efficiency when doing so. But the true potential of this idea is not tapped this way: Instead of inserting segments #1 to #n into the transmission schedule with periods 1 to



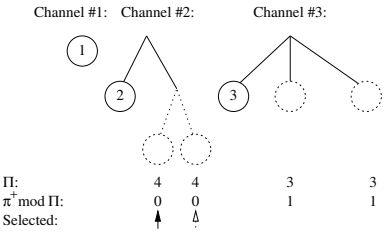
(a) Insertion of the first segment



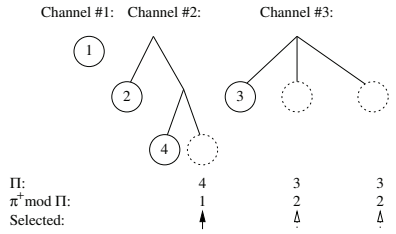
(b) Insertion of the second segment



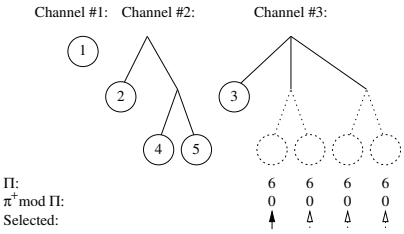
(c) Insertion of the third segment



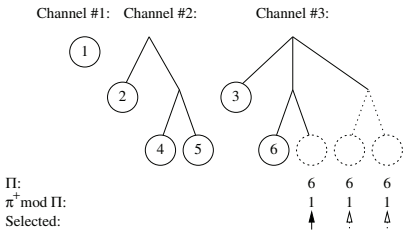
(d) Insertion of the fourth segment



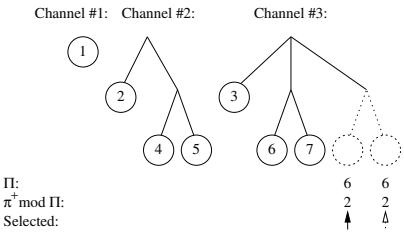
(e) Insertion of the fifth segment



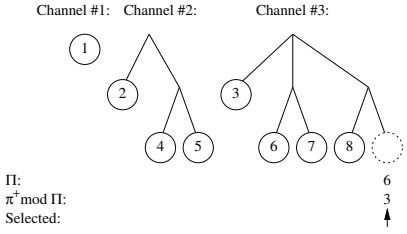
(f) Insertion of the sixth segment



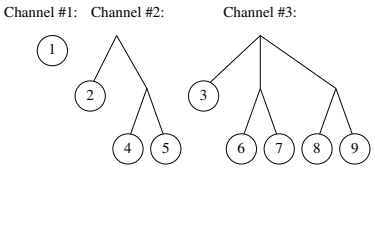
(g) Insertion of the seventh segment



(h) Insertion of the eighth segment



(i) Insertion of the ninth segment



(j) Resulting transmission schedule

Figure 4.5: Step-by-step generation of a transmission schedule using the Greedy Broadcasting/Recursive Frequency Splitting scheme for three channels

(a) Single split of a node into six parts

(b) Prime factor split into two and three parts

Figure 4.6: Prime factor splitting of the Greedy Broadcasting scheme

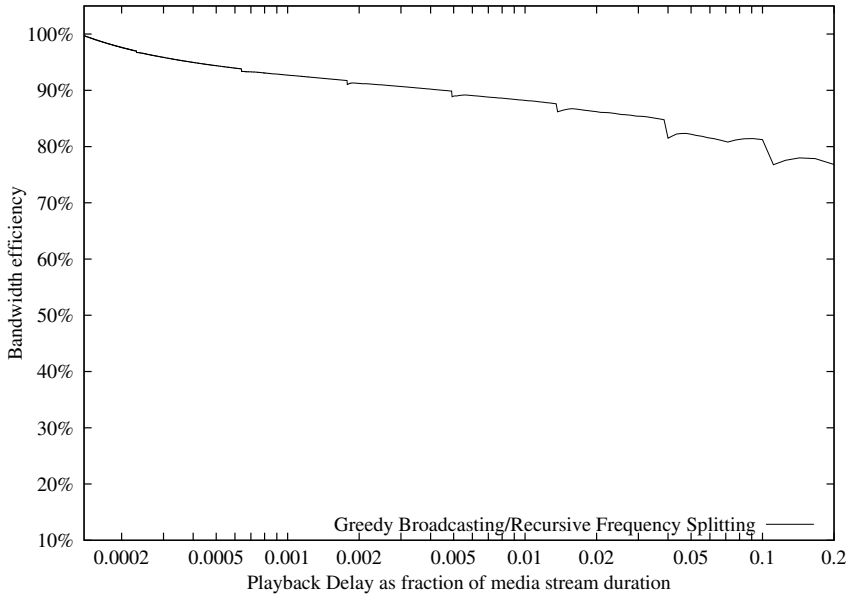


Figure 4.7: Efficiency of the Greedy Broadcasting/Recursive Frequency Splitting scheme

n, the above algorithm of the Greedy Broadcasting scheme/Recursive Frequency Splitting scheme can be generalized to insert any list of segment periods into a schedule.

Unfortunately, the efficiency of the Greedy Broadcasting scheme is very unsteady when the segments to insert are not sequentially ascending and starting from one: The simple heuristic used in the Greedy Broadcasting algorithm seems to benefit from the fact that insertion of the first segments prepares the transmission trees for an efficient insertion of the latter segments. Figure 4.8 shows the efficiency of the Greedy Broadcasting scheme using the generalization when the segment periods are $\pi_i^+ = i + k$ for different values of k and 7 200 segments (which equals one second segments for a two hour media stream). (In fact, this scenario is an important one as it occurs when an additional playback delay or partial preloading is used as shown below in sections 4.5 and 4.6.)

In the following section, the efficiency of this generalized version of the Greedy Broadcasting scheme is therefore improved. The resulting version of the Greedy Broadcasting scheme,

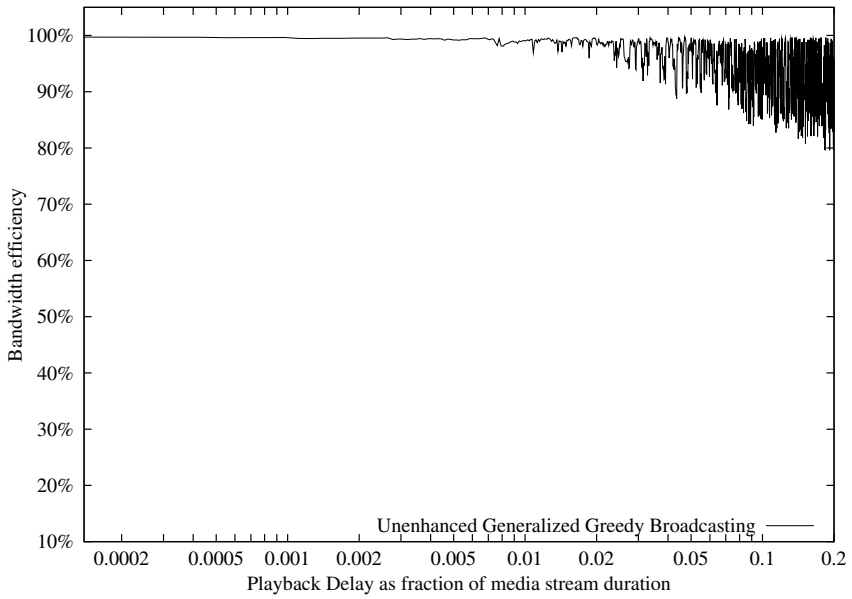


Figure 4.8: Efficiency of the generalized version of the Greedy Broadcasting scheme for different starting periods and 7 200 segments

called Generalized Greedy Broadcasting scheme in the following, shows not only a steady high efficiency, it even outperforms any transmission scheme of the frequency-based class.

In sections 4.5 to 4.11, the positive effects of this generalization are further exploited in several application cases which partially have been proposed in chapter 3 in the context of other transmission schemes.

4.3 Efficiency Improvements

To improve the efficiency and to make the efficiency less fluctuating at the same time, the heuristic decision of the Greedy Broadcasting scheme has to be improved: As described above, the Greedy Broadcasting scheme takes two measurements for each node N into account when it searches the best place for a segment #i:

1. The wasted bandwidth, measured by $\pi_i^+ \bmod \Pi_N$ and
2. the period of the node Π_N.

The Greedy Broadcasting algorithm evaluates these two measurements in this order, i. e. it first checks which nodes provide the lowest bandwidth wastage and examines the second condition only for the nodes which qualified equally according to the first condition.

Better results can be obtained if these two measurements are combined to a single “quality” function. For this purpose the measurements have to be normalized and weighted: The following

normalizations restrict the value of each measurement to the range 0 (poor) and 1 (best) and weights them with weighs $(1 - W^{II})$ and W^{II}:

$$\begin{aligned} Q^{I}_{i,N} &= \frac{\pi_i^+ \bmod \Pi_N}{\Pi_N} \\ Q^{II}_{i,N} &= \frac{\Pi_N}{\pi_i^+} \\ Q_{i,N} &= (1 - W^{II}) \cdot Q^{I}_{i,N} + W^{II} \cdot Q^{II}_{i,N} \end{aligned} \tag{4.3}$$

Experiments have shown that a weight $W^{II} \approx 0.005$ gives good results in most cases. Even better results can be obtained if several different weights are tried and the best resulting transmission schedule is selected. As further experiments have shown, very good results can be obtained for $W^{II} \in \{10^{\frac{k}{20}} | k \in \{-100, \ldots, 10\}\}$. Thus all given results for the Generalized Greedy Broadcasting scheme in this thesis assume that all weights $W^{II} \in \{10^{\frac{k}{20}} | k \in \{-100, \ldots, 10\}\}$ are tested and the best obtained schedule is chosen.

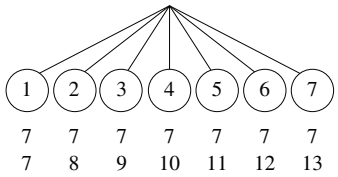
Another improvement can be made if the splitting process is examined more carefully: As described above, the Greedy Broadcasting scheme has been improved by performing prime factor splits to leave more huge parts whenever a non-prime split is needed. But this idea is not applicable if a node is to be split into a prime number of child nodes, and the impacts of this are the worse the higher the number for splitting and the lower the node period is. If the number of child nodes is reduced by one for these cases, the number of child nodes is not a prime any more and the prime factor splitting approach works again. By this mechanism, more large parts are left for the remaining segments at the cost of somewhat more bandwidth for the current segment.

Figure 4.9 shows an example for an application of this enhancement: If a segment with period $\pi^+ = 7$ is inserted into an empty tree, the root node of the tree would have to be split into seven child nodes if the proposed enhancement is not applied. This allows addition of six further segments if their periods are $\pi^+ = 8, \ldots, 13$, resulting in a total of seven segments in this tree. If the enhancement is applied, the root node will not be split into seven child nodes: Instead of this, it is split as if a segment with $\pi^+ - 1 = 6$ is inserted. As 6 is not prime, the prime factor optimization is applied, so the root node is split into two child nodes and one of the child nodes into three grandchild nodes. If further segments are inserted into this tree with ascending periods, the tree can be filled with seven further segments, one more than without the enhancement.

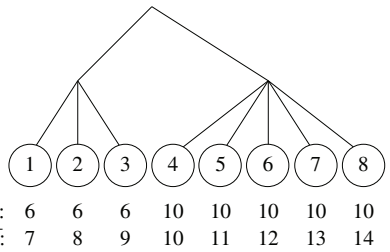
For the case that segments with ascending periods are to be inserted into one single node, it is possible to calculate the condition when this splitting enhancement is advantageous:

Assume that only a single node with period Π_N is available and that segments starting with period π_i^+ should be inserted. If the resulting number for splitting $p = \left\lfloor \frac{\pi_i^+}{\Pi_N} \right\rfloor > 2$ is a prime number[4], the original Greedy Broadcasting algorithm splits the node into p parts with a node

[4]It is assumed that $p > 2$, for $p = 2$ no enhanced splitting is possible.



(a) Splitting the root node into seven segments gives a transmission schedule with seven segments



(b) Splitting the root node into six segments instead allows to use prime factor optimization (split by two first and then by three) thus yielding eight segments in total

Figure 4.9: Example for increased efficiency from enhanced splitting

period $p \cdot \Pi_N$ each. These nodes can then only be used to transmit p segments as the segment periods do not allow further splitting. (See figure 4.9a for an example.)

But when the node is split into $p-1$ parts instead (as done in figure 4.9b), the prime factor splitting can be used because $p-1$ is dividable by 2 if $p > 2$ is a prime. Thus this prime factor splitting gives two nodes C_1 and C_2 of period $2 \cdot \Pi_N$. C_1 is then split further and can carry at least $\frac{p-1}{2}$ nodes (the exact number may be higher if $\frac{p-1}{2}$ is not prime and prime splitting can be applied again or if this enhanced splitting is advantageous for C_1). When these parts are used up, C_2 has to be split into (at least) $\left\lfloor \frac{\pi_i^+ + \frac{p-1}{2}}{2 \cdot \Pi_N} \right\rfloor$ nodes which allows to insert (at least) $\left\lfloor \frac{\pi_i^+ + \frac{p-1}{2}}{2 \cdot \Pi_N} \right\rfloor$ further segments. Therefore, the total number of inserted segments using enhanced splitting is (at least)

$$\begin{aligned}
n &\geq \tfrac{p-1}{2} + \left\lfloor \frac{\pi_i^+ + \frac{p-1}{2}}{2 \cdot \Pi_N} \right\rfloor \\
&\geq \tfrac{p-1}{2} + \left\lfloor \frac{\pi_i^+ + \Pi_N}{2 \cdot \Pi_N} \right\rfloor + \left\lfloor \frac{\frac{p-1}{2} - \Pi_N}{2 \cdot \Pi_N} \right\rfloor \\
&= \tfrac{p-1}{2} + \left\lfloor \frac{\frac{\pi_i^+}{\Pi_i} + 1}{2} \right\rfloor + \left\lfloor \frac{\frac{p-1}{2} - \Pi_N}{2 \cdot \Pi_N} \right\rfloor \\
&\geq \tfrac{p-1}{2} + \left\lfloor \frac{\left\lfloor \frac{\pi_i^+}{\Pi_i} \right\rfloor + 1}{2} \right\rfloor + \left\lfloor \frac{\frac{p-1}{2} - \Pi_N}{2 \cdot \Pi_N} \right\rfloor \\
&= \tfrac{p-1}{2} + \tfrac{p+1}{2} + \left\lfloor \frac{\frac{p-1}{2} - \Pi_N}{2 \cdot \Pi_N} \right\rfloor \\
&= p + \left\lfloor \frac{\frac{p-1}{2} - \Pi_N}{2 \cdot \Pi_N} \right\rfloor \\
&= p + \left\lfloor \frac{p-1-2 \cdot \Pi_N}{4 \cdot \Pi_N} \right\rfloor
\end{aligned} \tag{4.4}$$

This means that decreasing the split number if it is prime yields at least the same number of segments if $p \geq 2 \cdot \Pi_N + 1$ and definitively more segments if $p \geq 6 \cdot \Pi_N + 1$.

Even though this result cannot be generally applied if more than one node is available or the segment periods are not continuously increasing by one, it shows that it may be beneficial to

prohibit splitting by a large prime number. Experiments have shown that good results are gained if prime number splits for $p \geq L \cdot \Pi_N + 1$ with $L = 7$ are prevented. Better results can be obtained again if several different limits for the above fraction are applied and the best result is used. For the results in this thesis, the limits $L \in \{5, 7, \infty\}$ have been used.

Figure 4.10 shows the efficiency of the enhanced Generalized Greedy Broadcasting scheme for different playback delays, once with $W^{\Pi} = 10^{-\frac{46}{20}}$ and $L = 7$ and once using the best result from using $W^{\Pi} \in \{10^{\frac{k}{20}} | k \in \{-100, \ldots, 10\}\}$ and $L \in \{5, 7, \infty\}$. Although this enhancement seems to show only minor effect when compared to figure 4.7, the enhancement justifies itself when examining the efficiency for different starting periods as shown in figure 4.11 compared to figure 4.8.

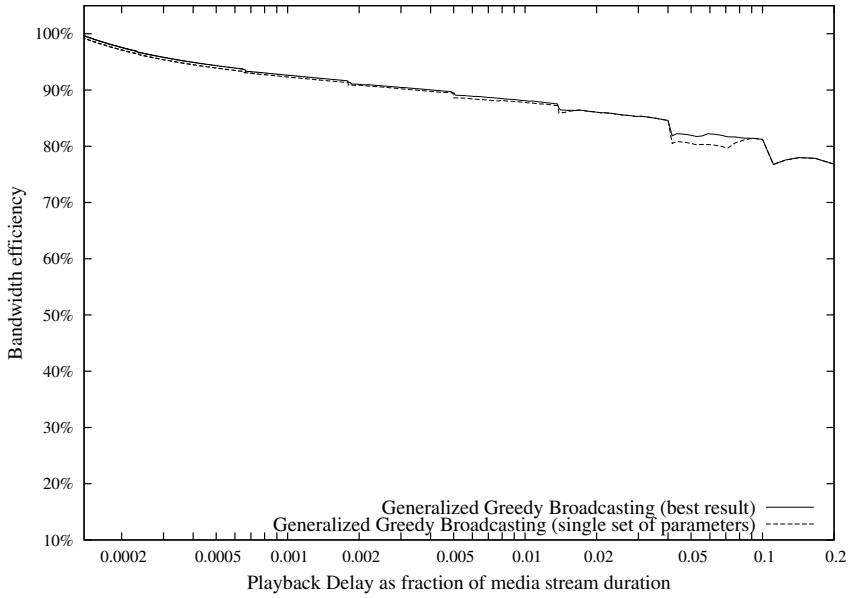


Figure 4.10: Efficiency of the enhanced Generalized Greedy Broadcasting scheme

4.4 Increasing the Playback Delay

Starting with this section, several enhancements for media-on-demand transmissions are examined. Many of these enhancements are part of one of the transmission schemes which have been proposed in chapter 3, but it is often possible to extract the mechanisms behind them and apply the underlying techniques to other transmission schemes — including the Generalized Greedy Broadcasting scheme — to improve the efficiency or enhance the usability.

In this section and the next section, the effect of modifying the playback delay to the required server bandwidth is examined. In general, nearly every pro-active transmission scheme

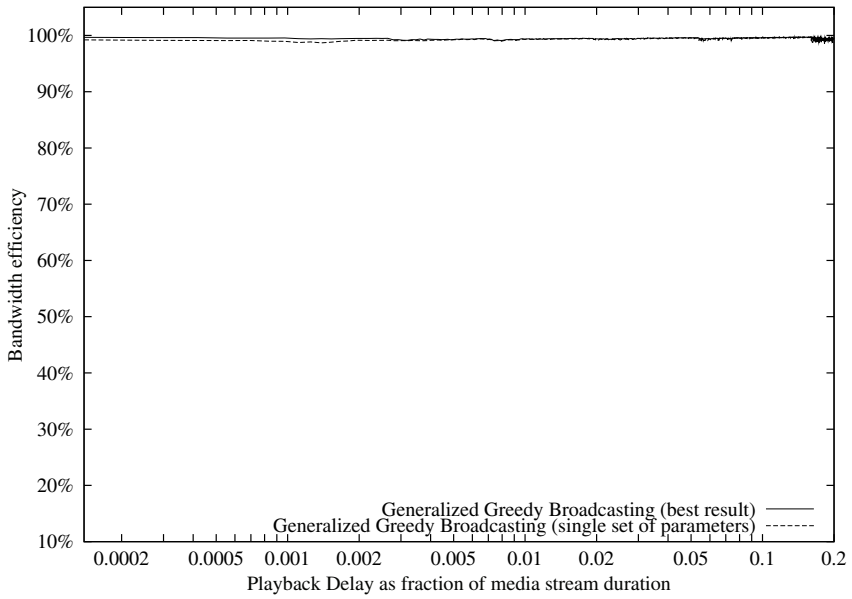


Figure 4.11: Efficiency of the enhanced Generalized Greedy Broadcasting scheme for different starting periods and 7 200 segments

(the Round-Robin Broadcasting scheme is an exception) supports modifications of the playback delay: For some transmission schemes, the playback delay is strictly coupled to the duration of the first segment and can only be modified by changing the segment sizes, but other transmission schemes allow separate modifications to the playback delay. As a first step, all calculations in this section assume that the playback delay is equal to the duration of the first segment (i. e. to the slot interval). The effects of modifying the playback delay independently of the slot interval are examined in the next following section.

4.4.1 Intention and Effects

The advantage of increasing the playback delay is that the receiver system has more time for receiving each segment, or in other words: All segments can be transmitted less frequently which lowers the bandwidth requirements. The bandwidth savings which are gained thereby can be approximated from the formula for the minimum required bandwidth: If the playback delay is increased from $W^+ = \delta$ to $W^{+\prime} = \delta'$, the required bandwidth reduces by

$$\begin{aligned}
\frac{B^-}{b} - \frac{B^{-\prime}}{b} &= \left(\Psi_0\left(\tfrac{d+W^+}{\delta}\right) - \Psi_0\left(\tfrac{W^+}{\delta}\right)\right) - \left(\Psi_0\left(\tfrac{d+W^{+\prime}}{\delta'}\right) - \Psi_0\left(\tfrac{W^{+\prime}}{\delta'}\right)\right) \\
&\approx \ln\left(\tfrac{d+W^+}{W^+}\right) + \gamma - \ln\left(\tfrac{d+W^{+\prime}}{W^{+\prime}}\right) - \gamma \\
&= \ln\left(\tfrac{d+W^+}{d+W^{+\prime}}\right) - \ln\left(\tfrac{W^{+\prime}}{W^+}\right) \\
&\approx \ln\left(\tfrac{W^+}{W^{+\prime}}\right)
\end{aligned} \tag{4.5}$$

The last approximation is only valid if the media stream duration is much larger than the playback delay.

This formula shows that an increase of the playback delay by a factor of approximately $e \approx 2.718$ (the Euler constant) reduces the bandwidth by one channel of media stream bit rate. Figure 4.12 shows the effect of modifying the playback delay to the minimum required bandwidth, still assuming that the slot interval is equal to the playback delay.

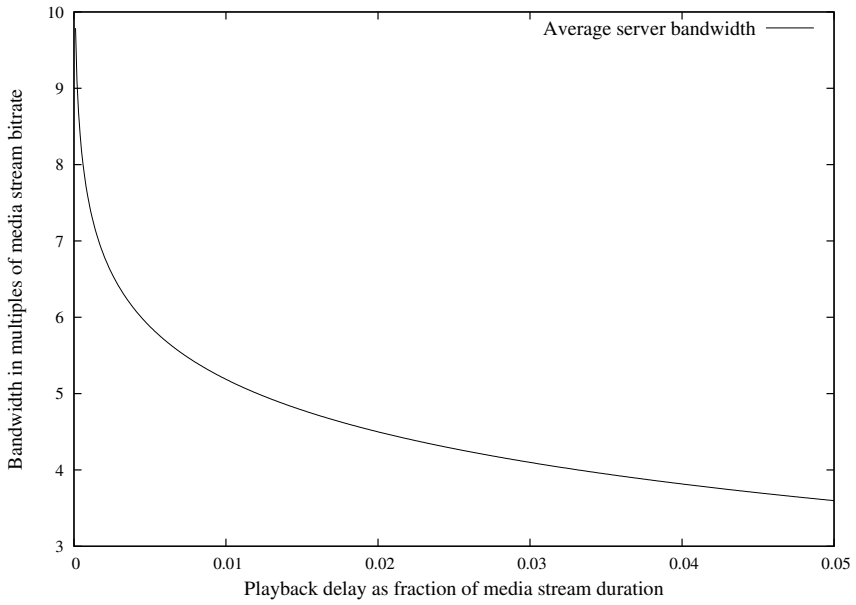


Figure 4.12: Impact of playback delay to minimum required bandwidth

The above formulas can also be used in the reverse case to estimate the playback delay which has to be used if only a limited amount of bandwidth is available.

4.4.2 Transmission Function for Playback Delays

As the playback function should define how much data of the media stream is required at a given time at the receiver system (relative to a playback request), a playback delay simply shifts the cumulative bandwidth function by the duration of the playback delay in the direction of the positive time axis, so that the corresponding transmission function can be defined by

$$T_{\text{playbackdelay}}(t) = t - W^{+} \tag{4.6}$$

The resulting playback function for a playback delay is

$$\begin{aligned} F(t) &= f(T_{\text{playbackdelay}}(t)) \\ &= f(t - W^{+}) \end{aligned} \tag{4.7}$$

where F denotes the playback function and f is the cumulative bandwidth function of the media stream.

Figure 4.13 shows an example for a transmission function where a playback delay has been applied to.

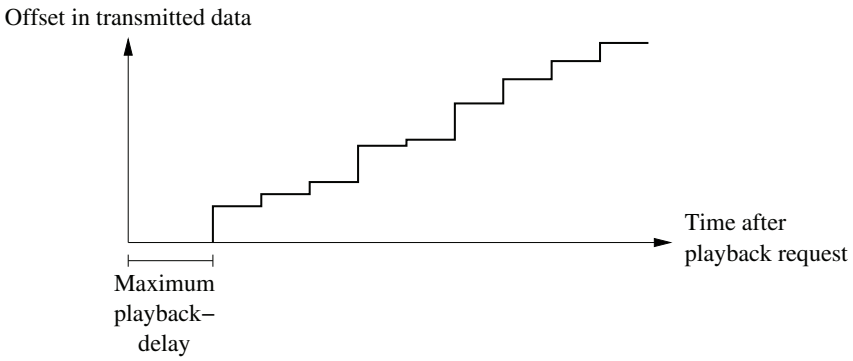


Figure 4.13: Playback function when a playback delay is in effect

4.4.3 Application to the Generalized Greedy Broadcasting Scheme

Increasing the playback delay for transmissions using the Generalized Greedy Broadcasting scheme by increasing the segment sizes gives nearly the same results as of the bandwidth calculations above. Only for very large segment sizes a higher efficiency loss emerges. A solution to this problem is presented in the following section.

4.5 Reducing Segment Sizes

This section examines the idea of decoupling the slot interval from the playback delay. For most transmission schemes, the slot interval for a transmission is determined by the playback delay: After the recipient system has joined the transmission, it awaits the next slot boundary and starts playback when the boundary is reached. (Both the network delay for joining the transmission and a delay for compensation of network jitter are disregarded here.) Therefore, the maximum playback delay is equal to the duration of the slot interval and the minimum playback delay is zero.

When the transmission scheme allows to select the playback delay independently (or at least as an integer multiple) of the slot interval, it is possible to reduce the segment size while keeping the maximum playback delay constant. In this case, the receiver system does not start the playback immediately when reaching the first slot boundary but after an **additional playback delay**.

4.5.1 Intention and Effects

Reducing the segment size independent of the playback delay has several advantages: Firstly, it is possible to use a segment size which matches the requirements of the media stream, independent of the playback delay. Secondly, the frequency-based transmission schemes perform at a higher efficiency in the case of a long playback delay: By using a slot interval much shorter than the playback delay, the number of segments is increased which allows for a more fine-grained scheduling of the segments. Thirdly, by decoupling the playback delay from the slot interval, the minimum playback delay is increased which reduces the bandwidth requirements without an increase of the maximum playback delay.

To demonstrate the bandwidth saving which is gained this way, a media stream of two segments is assumed as shown in figure 4.14a. When the slot interval is halved, the maximum playback delay would be halved at the same time, so an additional playback delay of one slot interval is introduced as demonstrated in figure 4.14b. But according to the additional playback delay, the even numbered halved segments can be transmitted less frequently as before, causing a bandwidth saving of about a forth of the media stream bit rate in this example. (For better understanding, the Harmonic Equal-Bandwidth Broadcasting scheme has been used, even if this transmission scheme is practically unusable.)

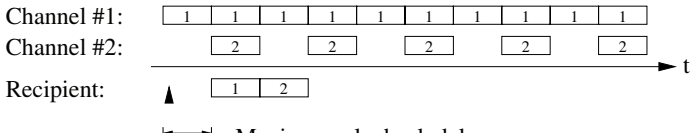
Figure 4.15 shows the bandwidth saving which is gained by reducing the segment size for a playback delay of $\frac{1}{120}$ of the media stream duration. Please note that the bandwidth does not diverge towards infinity if the segment size tends to zero:

$$\begin{aligned}\lim_{\delta\to 0}\frac{B^-}{b} &= \lim_{\delta\to 0}\left(\Psi_0\left(\tfrac{d+W^+}{\delta}\right)-\Psi_0\left(\tfrac{W^+}{\delta}\right)\right)\\ &\approx \lim_{\delta\to 0}\left(\ln\left(\tfrac{d+W^+}{\delta}\right)-\ln\left(\tfrac{W^+}{\delta}\right)\right)\\ &= \ln\left(\tfrac{d+W^+}{W^+}\right)\\ &= \ln\left(1+\tfrac{d}{W^+}\right)\end{aligned} \tag{4.8}$$

Often this formula is used to calculate the minimum required bandwidth for a media on demand transmission (as an approximation to equation (3.9) in section 3.1.3), disregarding that the segment size may not become infinitely small.

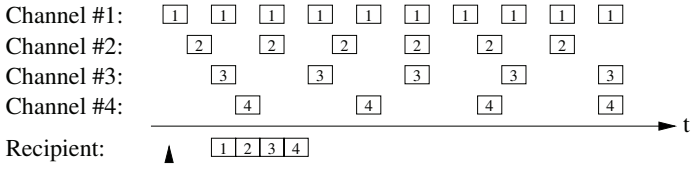
Due to this convergence, the amount of bandwidth which can be saved by reducing the segment size is also limited: If the slot interval is reduced from $\delta = W^+$ to an infinitely small value $\delta' \to 0$, the bandwidth savings are

$$\begin{aligned}\tfrac{B^-}{b}-\tfrac{B^{-\prime}}{b} &= \lim_{\delta\to W^+}\left(\Psi_0\left(\tfrac{d+W^+}{\delta}\right)-\Psi_0\left(\tfrac{W^+}{\delta}\right)\right)-\lim_{\delta'\to 0}\left(\Psi_0\left(\tfrac{d+W^+}{\delta'}\right)-\Psi_0\left(\tfrac{W^+}{\delta'}\right)\right)\\ &\approx \left(\ln\left(1+\tfrac{d}{W^+}\right)+\gamma\right)-\left(\ln\left(1+\tfrac{d}{W^+}\right)\right)\\ &= \gamma\end{aligned} \tag{4.9}$$



Segment #	Bandwidth usage
1	1 b
2	1/2 b
Total	3/2 b

(a) Transmission schedule with original segment size and no additional playback delay



Segment #	Bandwidth usage
1	1/2 b
2	1/3 b
3	1/4 b
4	1/5 b
Total	77/60 b

(b) Transmission schedule with halved segment size and an additional playback delay

Figure 4.14: Example for bandwidth saving by halving the segment size

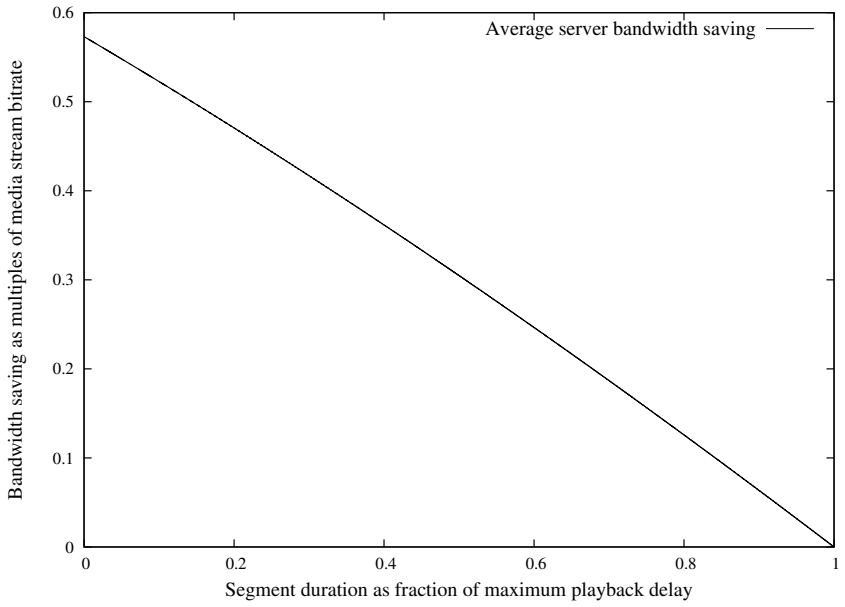


Figure 4.15: Bandwidth saving from reduction of slot interval

Depending on the ratio of the playback delay to the media stream duration, the bandwidth saving can be up to $\gamma \approx 0.577\,215\,665$ times the media stream bit rate.

As mentioned above, the minimum playback delay increases when an additional playback delay is introduced, thus this optimization must not be applied if the minimum (or average) playback delay is to be minimized.

4.5.2 Transmission Function for Reduced Segment Sizes

Reducing segment sizes has no effect on the transmission function because no part of the media stream is played back earlier or later. The change of the segment size only influences the granularity with which the resulting playback function is sampled by the transmission scheme when splitting the media stream into segments.

4.5.3 Application to the Generalized Greedy Broadcasting Scheme

The ability to decouple the segment size from other parameters like playback delay and preloaded amount of data (see next section) is one major advantage of the Generalized Greedy Broadcasting scheme: As the Generalized Greedy Broadcasting scheme allows to specify arbitrary segment periods when a transmission schedule is created, the segment periods can be calculated to respect any desired playback delay up to the resolution of one slot interval[5]. Figure 4.16 shows the efficiency of the Generalized Greedy Broadcasting scheme, once for the case that the playback delay equals the slot interval and once if the playback delay is decoupled from the slot interval and 7 200 segments are used. It is clearly visible that the decoupling of the playback delay increases the efficiency and removes the inefficiency peaks of long playback delays which result from bad scheduling capabilities for low numbers of segments.

The needed segment periods can be calculated using the following formula for constant-bit-rate media streams:

$$\pi_i^+ = i - 1 + \left\lfloor \frac{W^+}{\delta} \right\rfloor \tag{4.10}$$

The transmission of variable-bit-rate streams is examined in section 4.8 in more detail, including a new formula for segment period calculation which takes playback delay and partial preloading together with bandwidth variations into account.

[5] Although the Generalized Greedy Broadcasting scheme makes no assumption about segment sizes, it is suggested to align them to media integral boundaries (e. g. group-of-pictures or single frames) or to the size of data units of the lower-level protocols.

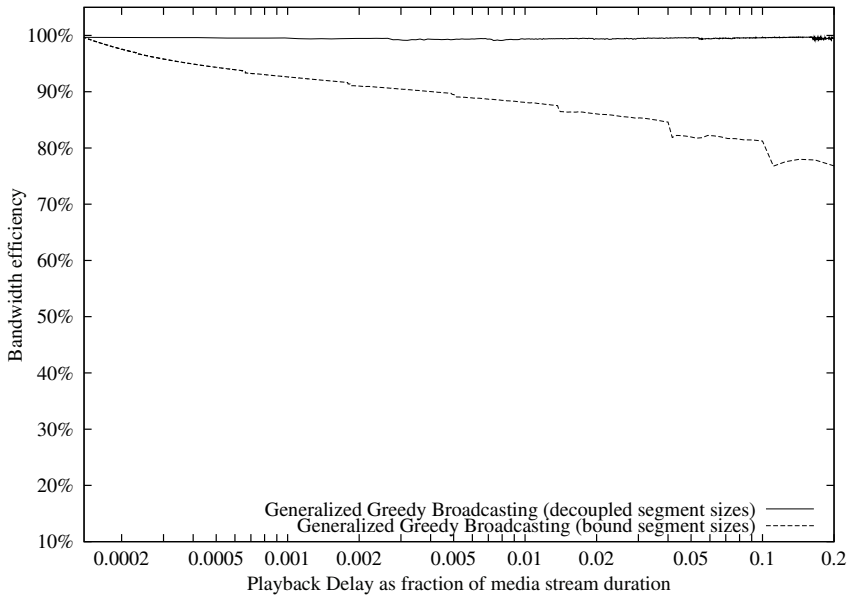


Figure 4.16: Comparison of Generalized Greedy Broadcasting scheme for bound and decoupled segment sizes

4.6 Partial Preloading

This section examines a mechanism called partial preloading [PL01, Pâr02, Pâr01c, PLM99] which has similar bandwidth savings as an increased playback delay. Partial preloading means that a part of the media stream is preloaded to the recipient systems storage at some time before the transmission is requested by the consumer.

4.6.1 Intention and Effects

Preloading has several impacts on the setup of the receiver system:

- The receiver system must always (or at least at the time of preloading) be activated if the media assortment is modified and preloading has to take place.
- The receiver system must always (or at least at the time of preloading) be connected to the network. In case of power or network failures, it may be possible that the customer cannot view a media stream until the preloaded part has been transmitted again.
- The receiver system must reserve part of its bandwidth for the reception of the preloaded data.
- The receiver system has to receive the preloaded parts before it can be used for the first time

(i. e. after buying the system, it cannot be used immediately unless the latest version of the preloaded parts have been stored on the system).

- The receiver system must reserve storage space for preloaded data.

But partial preloading has a lot of advantages at the same time:

- Sender and receiver bandwidth requirements for media stream transmissions are significantly lower,
- zero-delay playback is possible, even for pro-active transmission schemes, and
- preview of the beginning of media streams is possible at any time without additional bandwidth consumption or playback delay.

Theoretically, every media transmission scheme can use partial preloading by simply transmitting some part of the media stream to the receiver system in advance and applying the transmission scheme to the remaining part of the media stream (increasing the maximum playback delay by the duration of the preloaded part).

As the first segments of the media stream consume the most bandwidth in a transmission, the highest bandwidth saving is achieved if the beginning of a media stream is preloaded. More precisely, if a part of size P of the media stream is preloaded, the minimum required bandwidth of the non-preloaded transmission can be calculated using the following formula:

$$\begin{aligned} \frac{B^-}{b} &= \Psi_0\left(\frac{d+W^+}{\delta}\right) - \Psi_0\left(\frac{f^{-1}(P)+W^+}{\delta}\right) \\ &\approx \ln\left(\frac{d+W^+}{f^{-1}(P)+W^+}\right) \end{aligned} \tag{4.11}$$

Therefrom follows that the required bandwidth is reduced by partial preloading by

$$\begin{aligned} \frac{B^-}{b} - \frac{B^{-\prime}}{b} &= \left(\Psi_0\left(\frac{d+W^+}{\delta}\right) - \Psi_0\left(\frac{W^+}{\delta}\right)\right) - \left(\Psi_0\left(\frac{d+W^+}{\delta}\right) - \Psi_0\left(\frac{f^{-1}(P)+W^+}{\delta}\right)\right) \\ &\approx \ln\left(\frac{d+W^+}{W^+}\right) - \ln\left(\frac{d+W^+}{f^{-1}(P)+W^+}\right) \\ &= \ln\left(1+\frac{f^{-1}(P)}{W^+}\right) \end{aligned} \tag{4.12}$$

Figure 4.17 shows the effect of modifying the amount of preloading to the minimum required bandwidth for a constant-bit-rate media stream transmission with different numbers of segments.

As mentioned above, partial preloading also allows for zero-delay playback. This can be identified when the minimum bandwidth formula is examined for the case that W^+ tends toward 0: While the bandwidth without preloading diverges to infinity, the bandwidth stays limited if

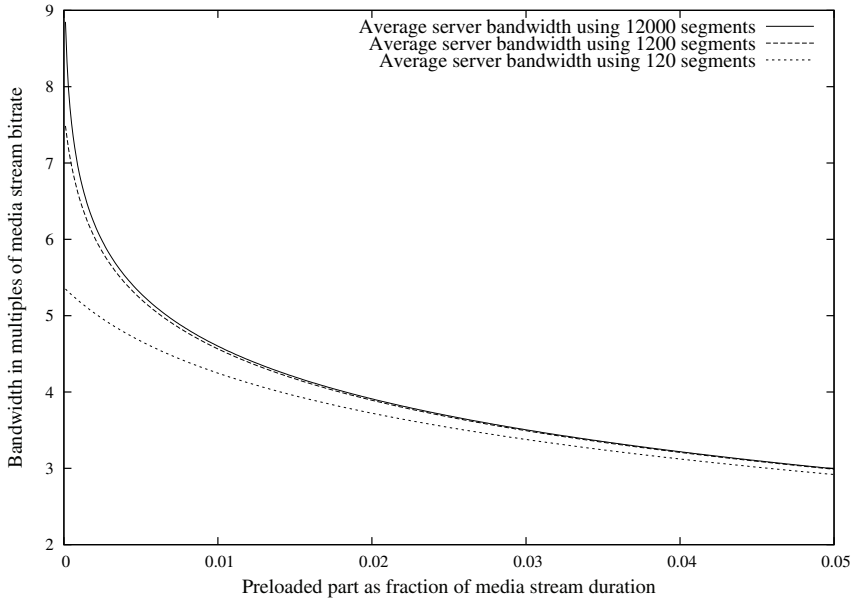


Figure 4.17: Impact of partial preloading to minimum required bandwidth

preloading is in effect:

$$\begin{aligned}\lim_{W^+\to 0}\frac{B^-}{b} &= \lim_{W^+\to 0}\Psi_0\left(\tfrac{d+W^+}{\delta}\right) - \lim_{W^+\to 0}\Psi_0\left(\tfrac{W^+}{\delta}\right)\\ &= \Psi_0\left(\tfrac{d}{\delta}\right) - \lim_{x\to 0}\Psi_0(x)\\ &= \infty\end{aligned} \tag{4.13}$$

$$\begin{aligned}\lim_{W^+\to 0}\frac{B^{-\prime}}{b} &= \lim_{W^+\to 0}\Psi_0\left(\tfrac{d+W^+}{\delta}\right) - \lim_{W^+\to 0}\Psi_0\left(\tfrac{f^{-1}(P)+W^+}{\delta}\right)\\ &= \Psi_0\left(\tfrac{d}{\delta}\right) - \Psi_0\left(\tfrac{f^{-1}(P)}{\delta}\right)\\ &\approx \ln\left(\tfrac{W^+}{f^{-1}(P)}\right)\end{aligned} \tag{4.14}$$

As for partial preloading in general, every media transmission scheme can use partial preloading with zero-delay playback if part of the media stream is transmitted in advance and if the transmission scheme is applied to the remaining part of the media stream with a maximum playback delay equal (or less) to the duration of the preloaded part.

To cope with the problem that partial preloading presupposes that the preloaded part must be present at the receiver system before the playback is started, the authors of [Pâr02] proposed to transmit the preloaded part additionally with a high playback delay. This gives the customer the choice between using partial preloading and gaining a short or even zero-delay access or refusing partial preloading and accepting a longer playback delay. This solution can also be used to handle transmission failures for the preloaded parts.

4.6.2 Transmission Function for Partial Preloading

Similar to a playback delay, partial preloading shifts the cumulative bandwidth function, but in a different direction: As partial preloading modifies the amount of data which has to be transmitted to the receiver system while leaving the playback time of any part of the media stream unmodified, the cumulative bandwidth function is shifted in the direction of the transmission offset in the media stream by the amount of preloaded data P:

$$T_{\text{partialpreloading}}(p) = p - P \tag{4.15}$$

The resulting playback function can be obtained the following way:

$$\begin{aligned} F(t) &= T_{\text{partialpreloading}}(f(t)) \\ &= f(t) - P \end{aligned} \tag{4.16}$$

Again, F denotes the playback function and f the cumulative bandwidth function of the media stream.

Figure 4.18 shows an example for a playback function where partial preloading has been used.

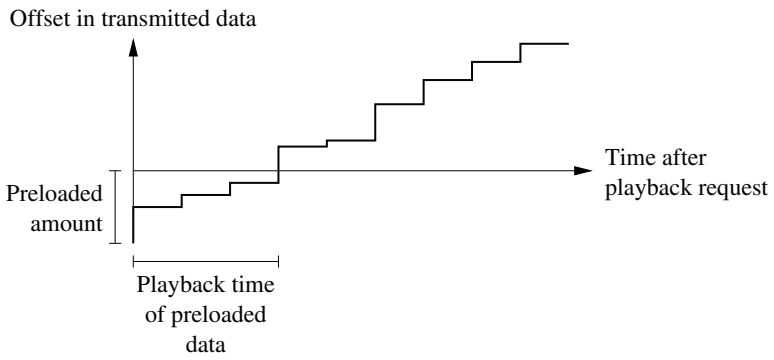


Figure 4.18: Playback function when partial preloading is used

4.6.3 Application to the Generalized Greedy Broadcasting Scheme

Preloading a part of the media stream equal to the segment size is the simplest way to use partial preloading with the Generalized Greedy Broadcasting scheme. But if a larger amount is to be preloaded, the segment sizes become very large this way which causes some efficiency losses. One way to circumvent this problem (which also occurred in the section 4.4 in the context of increased playback delay) is to shorten the slot interval independently of the amount of preloaded data as described in the previous section for the playback delay. The needed segment periods can be calculated in case of partial preloading using the following formula for constant-bit-rate media

streams:

$$\pi_i^+ = i - 1 + \left\lfloor \frac{W^+ + f^{-1}(P)}{\delta} \right\rfloor \tag{4.17}$$

From this equation, it is readily identifiable that $W^+ + f^{-1}(P) \geq \delta$ is necessary to ensure that all segment periods are non-zero, i. e. that it is possible to create a transmission schedule.

Optional partial preloading, i. e. transmitting the preloaded part additionally with a higher playback delay for receiver systems that do not have the preloaded parts, is also possible with the Generalized Greedy Broadcasting scheme: As the Generalized Greedy Broadcasting scheme allows to define an arbitrary period for each segment, any desired additional playback delay can be applied to the segments of the preloaded part.

4.7 Introducing Breaks into Transmissions

While an additional playback delay can be seen as a break at the beginning of the playback, it is also possible to insert breaks at any position into the media stream [PQ02]. These breaks may be required by the broadcasting organizations to show advertisements which can be transmitted alongside the media transmission or which have been preloaded to the receiver system.

4.7.1 Intention and Effects

Besides of commercial reasons in case of advertisements, breaks have a lowering effect on the bandwidth requirements: The part of the media stream after the break has to be transmitted less frequently after insertion of a break because the receiver system can use the duration of the break for receiving further media stream data.

Unfortunately, the bandwidth requirements of later parts of a media stream transmission need much less bandwidth than the former parts, so the saved bandwidth is the lower the later the break is inserted. Figure 4.19 shows the bandwidth reducing effect of a break of $\frac{1}{24}$ of the stream duration (this equals a five minute break in a two hour stream) using a playback delay of $\frac{1}{120}$ of the stream duration (this corresponds to one minute of a two hour media stream) depending on the time of the break.

4.7.2 Transmission Function for Insertion of Breaks

Inserting breaks into a transmission has a similar effect to the playback function as a playback delay; the only difference is that the shifting is delayed to the position of the break. Therefore, the

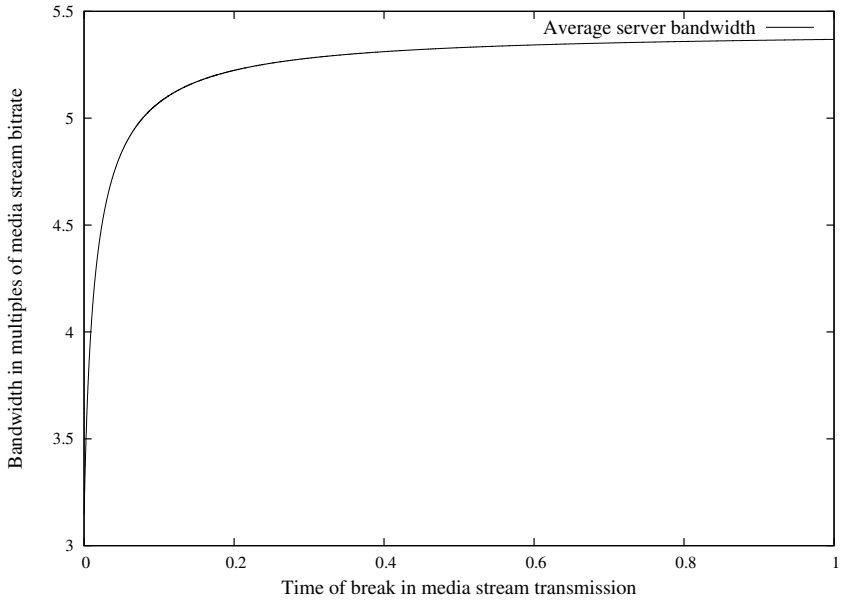


Figure 4.19: Impact of insertion of breaks into a stream transmission at different times

transmission function looks like this:

$$T_{\text{breakinsertion}}(t) = \begin{cases} t & \text{if } t < t_{\text{break}} \\ t_{\text{break}} & \text{if } t_{\text{break}} \leq t < t_{\text{break}} + d_{\text{break}} \\ t - d_{\text{break}} & \text{if } t_{\text{break}} + d_{\text{break}} \leq t \end{cases} \tag{4.18}$$

where t_{break} is the time of the break (relative to the beginning of the media stream) and d_{break} is the duration of the break.

The resulting playback function after applying this transmission function to a cumulative bandwidth function is

$$\begin{aligned} F(t) &= f(T_{\text{breakinsertion}}(t)) \\ &= \begin{cases} f(t) & \text{if } t < t_{\text{break}} \\ f(t_{\text{break}}) & \text{if } t_{\text{break}} \leq t < t_{\text{break}} + d_{\text{break}} \\ f(t - d_{\text{break}}) & \text{if } t_{\text{break}} + d_{\text{break}} \leq t \end{cases} \end{aligned} \tag{4.19}$$

Again, F denotes the playback function and f the cumulative bandwidth function of the media stream.

To insert several breaks into one media stream, transmission functions for each break can be

applied:

$$\begin{aligned} F(t) &= f((T_{\text{breakinsertion}_k} \circ \cdots \circ T_{\text{breakinsertion}_1})(t)) \\ &= f(T_{\text{breakinsertion}_k}(\cdots(T_{\text{breakinsertion}_1}(t))\cdots)) \end{aligned} \tag{4.20}$$

where $T_{\text{breakinsertion}_1}, \ldots, T_{\text{breakinsertion}_k}$ are transmission functions for each break, sorted by the time of the break (i. e. $T_{\text{breakinsertion}_1}$ is the transmission function for the first break). The times of the breaks must be specified relative to the beginning of the media stream (ignoring all breaks) if the transmission functions are applied in this order.

Figure 4.20 shows an example for a playback function where one break has been inserted.

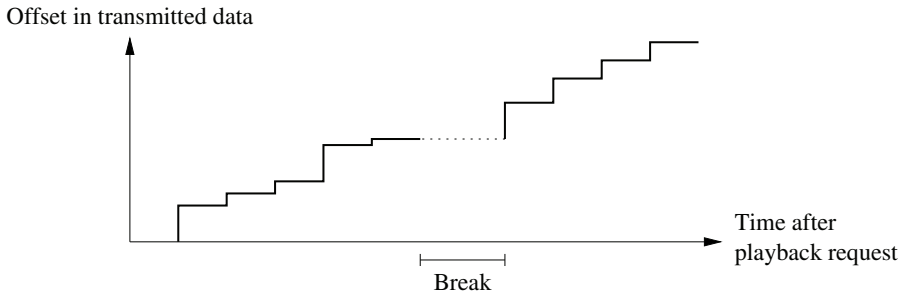


Figure 4.20: Playback function with a break inserted

4.7.3 Application to the Generalized Greedy Broadcasting Scheme

Using the Generalized Greedy Broadcasting scheme, breaks can be introduced at any time by adjusting the segment periods accordingly. If F^{-1} denotes the inverse of the desired playback function[6] the segment periods can be calculated by the following formula:

$$\pi_i^+ = \left\lfloor \frac{F^{-1}((i-1)\cdot\sigma)}{\delta} \right\rfloor \tag{4.21}$$

These segment periods can then be used by the Generalized Greedy Broadcasting algorithm to create a transmission schedule which benefits from partial preloading, increased playback delay and breaks.

Another interesting detail is, that it is still possible to derive that $F^{-1}(0) = f^{-1}(0+P)+W^+ = f^{-1}(P)+W^+ \geq \delta$ must be fulfilled to prevent zero segment periods from the above formulas, i. e. that the sum of the playback delay and the duration of the preloaded part must be greater than or equal to the slot interval.

[6] As the playback function is a staircase function, it is not invertible in a strict mathematical sense as it is not bijective. Assume therefore the following definition for the inverse playback function: $F^{-1}(p) = \min\{t \mid F(t) \geq p\}$

4.8 Variable-Bit-Rate Media Transmissions

The transmission of variable-bit-rate media streams is an important issue because most of the media encodings used today (e. g. H.261 [Itu93a], MPEG-1 part 2 [Itu93b], MPEG-2 part 2 [Itu00], H.263 [Itu05a], MPEG-4 part 2 [Itu04], H.264/MPEG-4 part 10 AVC [Itu05b], VC-1 [Soc06] and many others) perform better when generating variable-bit-rate output instead of (nearly) constant-bit-rate streams: To reduce the bandwidth as much as possible, the video compression algorithms benefit from two facts (see also section 2.5.3):

- The frames of the media stream contain a high amount of redundancy (which can be exploited through spatial compression).
- Consecutive frames often only differ in minor parts (which is often exploited through temporal compression).

Although many compression algorithms support generation of constant-bit-rate output by adapting the compression rate on-the-fly, the result has a lower quality than a variable-bit-rate encoded media stream of the same average bandwidth (and often even of a lower than average bandwidth): As the media encoder has to produce the same bit rate for parts of the media stream with low redundancy (complex frames, high motion) as for parts with high redundancy (simple frames, less or no motion), the further ones have to be encoded with a high compression rate (loosing many details) while the latter ones can be encoded using a low compression rate (producing a higher quality than average for these parts).

4.8.1 Smoothing Variable-Bit-Rate Media Streams

Instead of creating constant-bit-rate output, it is possible to use variable-bit-rate streams and apply a smoothing algorithm to them for transmission. **Smoothing** [SRR99, SRR04, ASS01, RR99] of a media stream is a mechanism to convert a variable-bit-rate media stream into a constant-bit-rate one where high bandwidth parts of the variable-bit-rate media stream are transmitted in advance using times of lower bandwidth for compensation. The advantage of smoothing is that the created output can be transmitted the same way as a constant-bit-rate media stream without loss of quality or increase of the average media bandwidth. Therefore, any media-on-demand transmission scheme can handle variable-bit-rate media streams by smoothing them beforehand.

Smoothing can benefit from the playback delay by using part of the delay for smoothing which results in a lower smoothed media bit rate. This is esp. important for the beginning of the media stream as no preceding part is available where high bandwidth parts can be moved to. On the other side, lowering the playback delay this way increases the bandwidth which is needed for the transmission in the same way as a reduction of the playback delay or the preloaded amount.

Therefore, one difficulty of smoothing is that the playback delay has to be divided optimally into a part which is used by the transmission schedule and a part which is used for smoothing.

Figure 4.21 shows an example for smoothing a variable-bit-rate media stream transmission which uses different amounts of the playback delay for smoothing.

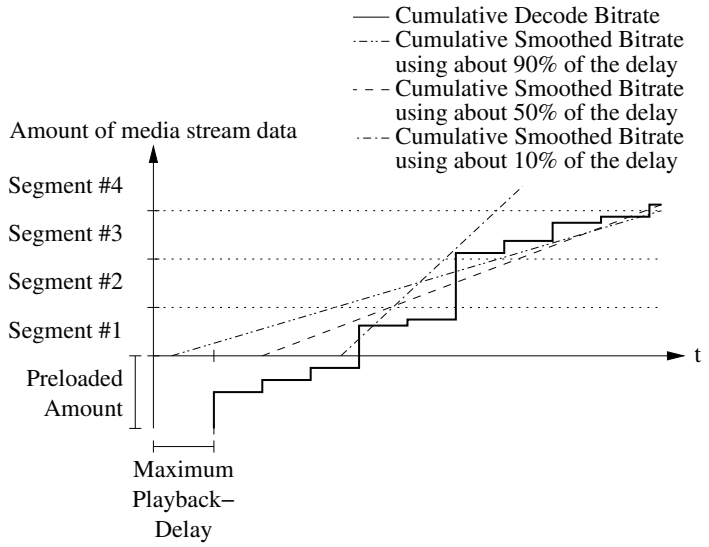


Figure 4.21: Example for smoothing of a variable-bit-rate media stream

4.8.2 Piecewise Smoothing of Variable-Bit-Rate Media Streams

Using one smoothing bandwidth for the whole media stream has mainly two disadvantages: Firstly, the smoothing bandwidth may be much higher than e. g. the average bandwidth of the media stream which reduces the efficiency of the transmission scheme. And secondly, smoothing increases the memory requirements at the receiver side as any part which is transmitted in advance due to smoothing has to be stored at the receiver system. The authors of [LT95] and [SZKT96] therefore developed heuristic algorithms to split a media stream into pieces of different bit rate so that the bit rate within each piece remains constant and a given smoothing buffer size is not exceeded. But even for piecewise smoothing, a compromise between a low number of bandwidth changes and a small buffer size for smoothing has to be found.

To benefit from piecewise smoothing in case of a frequency-based transmission scheme which cannot transmit variable-bit-rate media streams, each piece of the media stream for which a new bandwidth has been selected has to be scheduled in a separate transmission scheme. This creates a lot of scheduling overhead and complexity, so piecewise smoothing looses lots of its advantages for this type of transmission schemes.

4.8.3 Immediate Transmission of Variable-Bit-Rate Media Streams

While the smoothing approaches remove the complexity of variable-bit-rate transmissions from the transmission scheme, they have several disadvantages:

- By moving high bit rate parts of the media stream backward in time to a position of low bit rate, the needed transmission bandwidth is increased (and therefore the efficiency of the transmission reduced) as the former parts of a transmission are transmitted more frequently and thus require more bandwidth than the latter parts.
- Live transmissions cannot be smoothed (if no further delay for smoothing is added) as it is not possible to transmit any media stream data in advance, i. e. before it has been captured.
- The storage requirements at the receiver system are increased slightly by smoothing as the earlier transmitted parts have to be stored until they are played.

To support variable-bit-rate media streams directly, the transmission scheme must be able to create a transmission schedule which adapts to the bandwidth variations of the media stream as much as possible. For example, the authors of [SRR99] and [LN99] proposed algorithms to calculate broadcasting series for size-based transmission schemes which performs this adaption to the bandwidth variations, the authors of [Pâr99a] proposed bandwidth calculations for the bandwidth-based Cautious Harmonic Broadcasting scheme and the authors of [LN00] for the bandwidth-based Staircase Broadcasting scheme.

4.8.4 Application to the Generalized Greedy Broadcasting Scheme

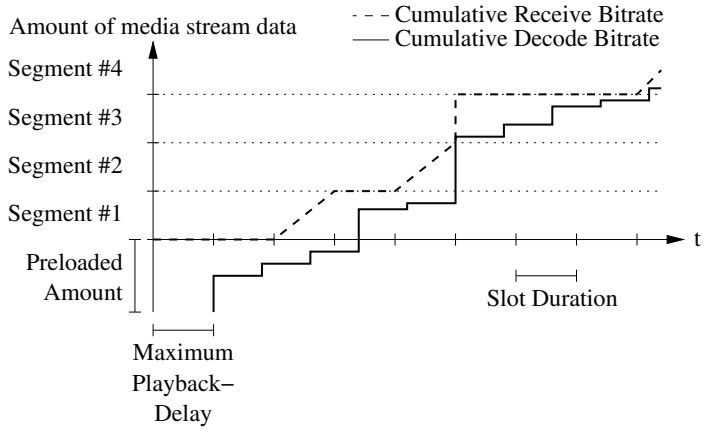
Using the inverse playback function F^{-1}, the maximum period for segment #i of a variable-bit-rate media stream can be calculated as follows:

$$\pi_i^+ = \min_{0 \le x \le \sigma} \left\lfloor \frac{F^{-1}((i-1)\sigma + x) - \frac{x}{\beta}}{\delta} \right\rfloor \tag{4.22}$$

This function calculates the maximum period of a segment by examining each offset within the segment and using the minimum of the calculated periods. This minimum search is necessary to prevent that a buffer underflow happens within a segment at the receiver system: Even if the beginning of a segment arrives just-in-time, it is not guaranteed that all data of the segment is received when it is needed due to the bit rate variations within the segment. By searching the minimum period for every stream offset of the segment, the segment period is adjusted accordingly.

As the Generalized Greedy Broadcasting scheme allows defining an arbitrary sequence of non-descending segment periods when generating a transmission schedule, the above formula suffices to create a schedule which can be used to transmit the media stream in an efficient way. Figure 4.22

shows an example transmission of the beginning of a variable-bit-rate transmission together with the needed segment periods for the same transmission.



Segment #	Segment Period
1	3
2	5
3	5
4	9

Figure 4.22: Example for a variable-bit-rate transmission using the maximum possible segment period for each segment

To verify the results for adapting to a variable-bit-rate media stream instead of smoothing to constant-bit-rate, several publicly available video traces [FR01] have been used. To get comparable results, all video traces have been extended to two hours. Additionally, a playback delay of 60 seconds has been assumed to make smoothing possible. Table 4.1 shows the additional bandwidth requirements of the video traces for three transmission methods (Fixed-Delay Pagoda Broadcasting and Generalized Greedy Broadcasting, both using a smoothed version of the variable-bit-rate media stream, and the Generalized Greedy Broadcasting using the variable-bit-rate media stream directly) relative to the theoretical minimum of the bandwidth for the transmission.

Although the results for the smoothed video streams show strong variations (between 11 % and 53 % more bandwidth than the theoretical minimum) a significant decrease of the bandwidth requirements is achieved when the Generalized Greedy Broadcasting scheme is applied to the variable-bit-rate media stream: While the average bandwidth requirement for the smoothed media streams lies 25 % above the theoretical minimum, it is less than 1 % away from the theoretical minimum for all examined video traces when using the variable-bit-rate media stream directly.

Media name	Additional Bandwidth (smoothed, FDPB)	Additional Bandwidth (smoothed, GGB)	Additional Bandwidth (VBR, GGB)
Aladdin	+40.7 %	+36.0 %	+0.4 %
Die Hard (III)	+26.9 %	+23.3 %	+0.3 %
From Dusk till Dawn	+22.9 %	+18.6 %	+0.4 %
Jurassic Park	+24.0 %	+20.2 %	+0.3 %
Mr. Bean	+27.0 %	+22.8 %	+0.4 %
Robin Hood	+32.8 %	+28.8 %	+0.2 %
Silence of the Lambs	+57.0 %	+52.9 %	+0.5 %
Simpsons	+14.2 %	+11.3 %	+0.2 %
South Park	+17.9 %	+14.4 %	+0.5 %
Star Trek, First Contact	+36.9 %	+30.9 %	+0.5 %
Star Wars IV	+27.7 %	+21.4 %	+0.7 %
Starship Troopers	+25.2 %	+21.0 %	+0.4 %
The Firm	+27.8 %	+22.0 %	+0.7 %
Average	+29.3 %	+24.9 %	+0.4 %

Table 4.1: Bandwidth requirements for transmission of variable-bit-rate media streams compared to theoretical minimum

4.9 Immediate Segment Reception

For the Polyharmonic Broadcasting scheme (see section 3.6.4), the authors introduced a fixed-wait policy to circumvent a design error of the Harmonic Broadcasting scheme (see section 3.6.1). This fixed-wait policy requires that the reception is started immediately when the channels are joined instead of delaying the reception until the next segment boundary is reached[7] and starting the playback after a fixed delay instead when reaching the next segment boundary.

4.9.1 Intention and Effects

Joining an ongoing transmission immediately has the advantage that it is assured that a segment has been completely received after a delay equivalent to the segment period, even if the bandwidth of the transmission channel is lower than the media bit rate. (This is the way how the Polyharmonic Broadcasting scheme works around the problem of the Harmonic Broadcasting scheme.) But the same advantage can also be used in other cases where the bandwidth of the transmission channel is (permanently or temporarily) lower than the media bit rate, e. g. when transmitting a variable-bit-rate media stream as explained in the previous section. Figure 4.23 shows an example of the same frequency-based transmission of a variable-bit-rate media stream as in section 4.8, this time

[7] Strictly speaking, it is not possible to join an ongoing transmission immediately, at least delays from link layer access (e. g. tuning to the right satellite frequency), routing (e. g. IGMP group join delays) and packetization (for packet-based protocols) must be accepted. If these delays are added to the maximum playback at a later stage, they can be ignored here for improved comprehension.

utilizing immediate receptions: As segments are never received concurrently to their playback in this case, the cumulative receive bandwidth is a staircase function (figure 4.23b shows the result of the immediate reception compared to figure 4.23a with delayed reception). This allows to increase the segment period for some segments (as illustrated in figure 4.23c).

The only disadvantage of immediately receiving data instead of awaiting the next segment boundary is a gain of a little bit more complexity, slightly higher storage requirements and an increase of the minimum playback delay up to the maximum playback delay.

If the bandwidth of the transmission channels equal the media bit rate (e. g. when a constant-bit-rate media stream is transmitted using a typical frequency-based transmission scheme) starting the reception immediately has no advantages as the data is delivered with the same rate as it is consumed.

4.9.2 Application to the Generalized Greedy Broadcasting Scheme

The Generalized Greedy Broadcasting scheme benefits from immediate segment receptions for variable-bit-rate transmissions as it allows lowering of segment periods in some cases. Even the complexity of the segment period calculation is lowered when immediate segment receptions are allowed: The maximum period for segment #i can then be calculated by:

$$\pi_i^+ = \left\lfloor \frac{F^{-1}((i-1)\cdot\sigma)}{\delta} \right\rfloor \tag{4.23}$$

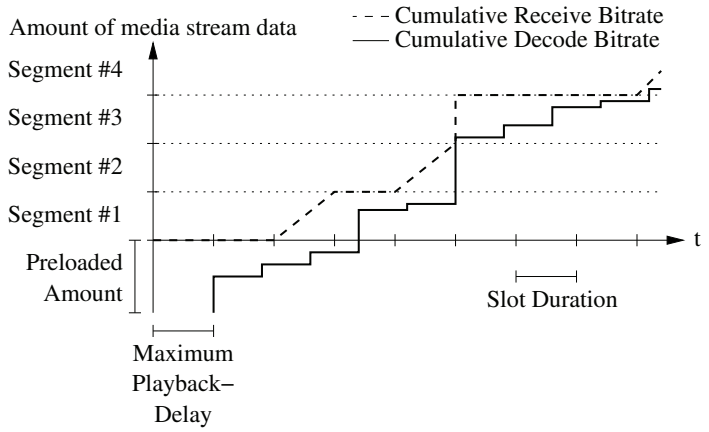
Compared to equation (4.22) of section 4.8, this formula lacks the minimum search as it is sufficient in this case to check if the beginning of a segment arrives just-in-time.

4.10 Decoupled Channel-Bandwidth

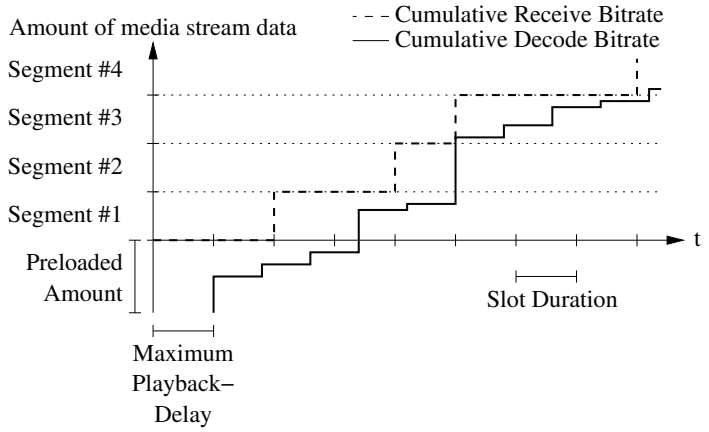
Most of the frequency-based transmission schemes require that the channel bandwidth equals the media bit rate. Although this makes the transmission scheme more simple, it may not fit to existing environments, e. g. if the network provides channels of fixed bandwidth or if the total available bandwidth is not an integer multiple of the media bit rate.

4.10.1 Intention and Effects

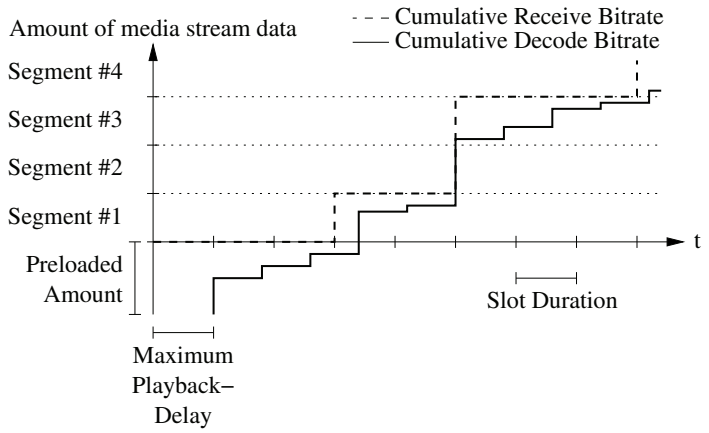
In these cases, selecting an arbitrary channel bandwidth according to the network and independent of the media bit rate helps to utilize the full network bandwidth. Theoretically, any channel bandwidth is possible as long as the resulting maximum period for the first non-preloaded segment is non-zero. But practically, a low channel bandwidth prohibits a fine-grained scheduling of segments as the slot interval increases (assuming that the segment size is kept constant).



(a) Transmission with delayed reception



(b) Transmission with immediate reception but unmodified segment periods



(c) Transmission with immediate reception when segment periods are increased

Figure 4.23: Example for bandwidth saving by starting reception immediately

accordingly, so the total bandwidth requirements stay the same (apart from rounding errors in the period calculation).

4.10.2 Application to the Generalized Greedy Broadcasting Scheme

As the Generalized Greedy Broadcasting scheme supports variable-bit-rate media streams directly, it does not rely on the media stream bit rate for channels. The segment periods have to be calculated in this case in the same way as for variable-bit-rate media streams.

If channels of high bandwidth are used and the receiver systems are able to receive all channels at once, no significant advantage is measured for the Generalized Greedy Broadcasting scheme as it already operates near the theoretic limit.

Nevertheless, decoupling the channel bandwidth may be beneficial when the maximum receiver bandwidth is not an exact multiple of the media bit rate (see also section 4.18) and a decoupling of the channel bandwidth from the media bit rate may be advantageous if the channel bandwidth cannot be modified.

4.11 Channel Sharing

When the segment size and playback function are specified in advance, it is the normal case that the transmission scheme used does not consume all needed channels completely. This section covers topics how the remainders of partially used channels can be utilized.

4.11.1 Intention and Effects

The unused bandwidth of the last channel may be used for several purposes (e. g. transmission of media stream beginnings for partial preloading, preloading of commercials for breaks, software updates, electronic program guides or other non-real-time services), but the authors of [PCL99b] suggested that channel may be **shared** among several transmissions so this bandwidth can still be used for media-on-demand transmissions.

Unfortunately, sharing channels raises several problems, esp. for frequency-based transmission schemes:

- The receiver system has to identify the media streams on the shared channel so it can discard any data from foreign transmissions.
- The channel parameters (segment size, channel bandwidth) must be equal for all transmissions which use the shared channels to allow an efficient sharing.
- In some cases, it may be necessary that more than one shared channel has to be used for a single transmission (e. g. if three transmission schemes all need two thirds of a channel,

one of them has to use the leftover parts of the others). This has to be considered when determining the receiver system bandwidth requirements.

- When a new schedule is created, the free slots of the shared channel have to be known. As a consequence, transmission schedules cannot be generated in advance but must be created at media startup.
- Additionally, a fragmentation problem might occur after some media allocations and releases, comparable to the fragmentation problems of dynamic heap memory allocation. A channel compaction algorithm may resolve this issue but must take care not to disrupt ongoing transmissions.

Some of these problems are relatively simple to solve, others — depending on the transmission scheme — much more difficult:

- To distinguish several transmissions on a shared channel, different addresses (or media stream identifiers if addressing within a channel is not supported by the lower layer protocols) can be used for each transmission.
- Using equal segment sizes and bandwidths for all transmissions is no problem if the transmission scheme supports that the segment size can be selected independently from the playback delay and that the channel bandwidth can be chosen independently from the media stream bit rate.
- The increase of the receiver system bandwidth has to be taken into account. If the receiver systems do not support a higher bandwidth, channel sharing has to be limited accordingly.

For the last two problems, a better solution is provided below.

4.11.2 Application to the Generalized Greedy Broadcasting Scheme

One way to apply channel sharing to the Generalized Greedy Broadcasting scheme is to generate a schedule which exactly consumes the holes of the currently active transmissions. But this procedure has two big disadvantages: Firstly, the unused parts of the transmission schedule are distributed across several channels which make it difficult to utilize them by other schedules without increasing the receiver bandwidth requirements too much. Secondly, when the active transmission terminates, segments from later started transmissions still allocate some segment positions in the transmission schedule and prevent that the schedule can be torn down. As a result, the next transmission schedule has to align itself with the remaining tree structure of the previous transmission which typically results in a worse efficiency.

As a solution, it is advisable to put a fixed structure on the last channel (e. g. split it into a number of equally sized subtrees) and modify the schedule generating algorithm in such a way that as

few parts of the last channel are used as possible when the transmission scheme is generated. This way, the "holes" are forcibly combined to the unused parts where they can simply be exchanged by parts from other schedules which have been created in the same way. Another advantage of this procedure is that the transmission schedule generation is decoupled from active transmissions so the schedule can still be generated in advance.

4.12 Enhanced Startup

A topic which has been paid very little attention to in recent pro-active transmission scheme development is startup and termination of media-on-demand transmissions. This section provides an algorithm for enhancing the startup of the Generalized Greedy Broadcasting scheme which can also be applied to other frequency-based schemes.

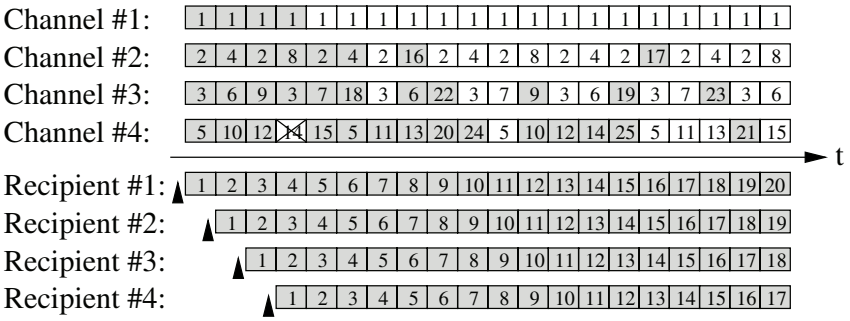
4.12.1 Intention and Effects

When a media transmission is started, all segments of the media stream are transmitted in the order which is defined by the transmission schedule. An interesting fact at startup is that some segments are transmitted this way although no receiver system will use them[8]: If a segment is transmitted at a period lower than necessary, it is possible that the segment is received twice by the receiver system before it is played[9]. While, at a later stage of a transmission, none of the segment transmissions can be omitted without disrupting possibly ongoing transmissions, it is possible to leave out the first of these broadcasts at the beginning of a newly started media stream transmission. Figure 4.24 shows an example where the transmission of segment #14 can be omitted once at media startup without destroying the continuity of any playback.

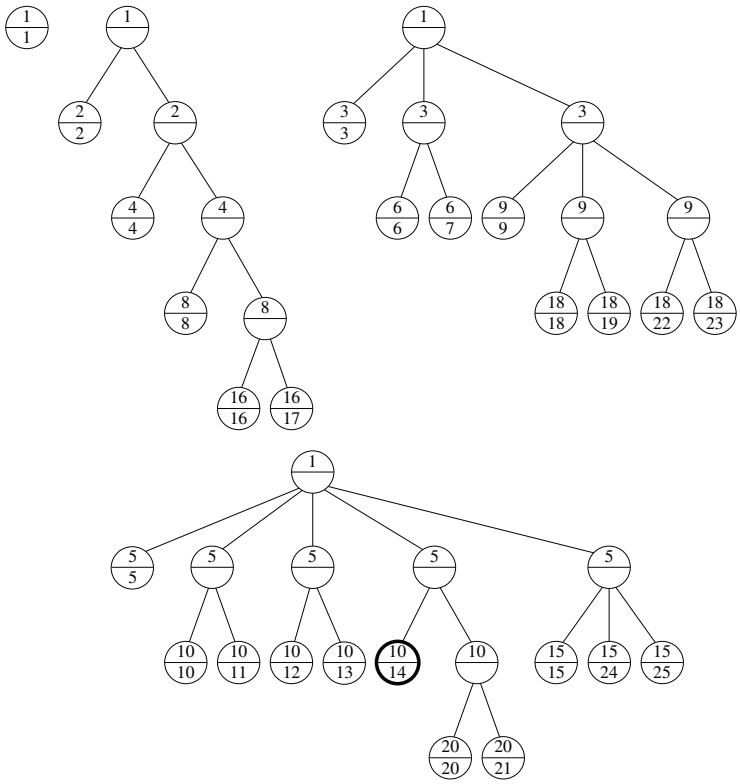
Although an examination of Generalized Greedy transmission schedules shows that only very few segment transmissions can be dropped this way (e. g. none of the 1 572 segments of a transmission schedule for eight channels and only two out of the 11 936 segments of a schedule for ten channels), this approach shows its full benefits when applied to transmission schedules which have been prepared accordingly.

[8] In this thesis, it is assumed that the receiver systems always try to receive only the latest transmission of a segment. Another approach would be to store any received segment, even if the transmission schedule would guarantee that the segment is transmitted in due time again. Although the latter approach provides a lower loss probability for these segments, it is not examined further as the loss probability is only decreased for a limited number of segments and only for a low number of receiver systems, so this effect is barely utilizable (see section 5.1 for better error correction schemes). Another disadvantage is an increase of the storage requirements when segments are stored prematurely.

[9] Although a transmission with a lower period means that the transmission scheme is non-optimal, it is the common case for most of the segments: For example, only about 17 % of all segments of a Generalized Greedy Broadcasting transmission schedule using eight channels are scheduled with their maximum period.



(a) The first transmission of segment #14 can be omitted



(b) Transmission trees of the used schedule

Figure 4.24: Example showing that it is possible to leave out a segment transmission at media startup without interfering with any receptions

4.12.2 Modifying Transmission Schedules

One way to get transmission schedules which allow leaving out more segments at startup is to modify the transmission schedule algorithm of the Generalized Greedy Broadcasting scheme, e. g. the quality function which is responsible for the placement of the segments in the transmission trees. Although this approach allows creating transmission schedules where more segments can be left out at startup, the long term efficiency of the Generalized Greedy Broadcasting scheme

is lowered for a short term startup enhancement. For this reason, another approach is proposed here which enhances the startup of a previously created, frequency-based transmission schedule without changing its (long-term) efficiency.

To enhance the startup of a transmission schedule it is important to identify the condition when a segment transmission can be left out. The basic principle has already been stated above: The first transmission of a segment can be left out if all receivers get the segment twice before they need it for playback, i. e. if segment #i is received twice within the first π_i^+ slot intervals. As segment #i is transmitted at $t = \phi_i \cdot \delta$ the first time and at $t = (\phi_i + \pi_i) \cdot \delta$ the second time, this means, the algorithm has to move as many segments as possible in such a way that $\phi_i + \pi_i < \pi_i^+$ is satisfied for them.

To preserve the efficiency of the transmission scheme, only a limited number of modifications are allowed. The proposed algorithm actually only uses the following two transformations:

T1 Exchange of segments of the transmission trees if the periods of all affected nodes are less than or equal to the required periods of the newly assigned segments[10].

T2 Exchange of (sub)trees if the period of their root nodes are equal.

With these basic principles in the mind, the enhancement algorithm works the following way: As a first step, it extracts the interior structure of the transmission schedule, i. e. for each node of the tree representation of the transmission schedule, the node period and the number of child nodes are fetched and saved.

In a second step, the transmission schedule is rebuilt from scratch, using only the extracted information and the list of segments and their required periods. This step works like assembling pieces of a puzzle: The algorithm starts with empty trees (one for each channel) and re-inserts all segments into them in ascending order. To find the best position where a segment is inserted, the algorithm tries every possible leaf node in the trees which can be constructed using the (remaining) extracted information pieces about the former structure of the trees. Thereby, it assembles the extracted information pieces using the following rules:

1. Only a piece with period “1” can be used as root of a tree.

   This is necessary according to transformation T2 for the whole tree.

2. A piece can only be the root of the subtree if its period matches the period of the position where it is inserted.

   This is necessary according to transformation T2 for a subtree.

[10]This transformation requires that both the needed and utilized period is known for all segments. While the utilized periods can be extracted from the transmission schedule even if the playback function of the media stream is not known any more, the required periods have to be provided additionally.

3. Each node must have exactly the number of child nodes as defined in the information piece which has been assigned to it.

   This is necessary according to transformation T2, only a whole subtree may be exchanged.

4. A segment can only be assigned to a leaf node, i. e. the associated piece must define that the node has no children. The period of the node must additionally be less or equal to the required period of the segment.

   This is a direct implication of transformation T1.

5. Each piece must be used exactly once in all trees together.

   Transformations T1 and T2 only allow exchanges of segments or (sub)trees, so omissions and duplications are not allowed.

This yields a non-empty[11] list of positions $\mathcal{N}$ where segment #i can be inserted. From this list, a node N is selected according to the following rules:

Let $\mathcal{P} = \left\{ N \in \mathcal{N} \,\middle|\, \Phi_N + \Pi_N < \pi_i^+ \right\}$ be the list of segment positions so that segment #i can be placed in such a way that its first transmission can be omitted.

1. If $\mathcal{P} \neq \varnothing$, select (one of) the node(s) of this set which provides the latest segment transmission, i. e. $N \in \underset{M \in \mathcal{P}}{\operatorname{argmax}}(\Phi_M + \Pi_M)$.
2. If $\mathcal{P} = \varnothing$, select the node with the latest segment transmission of all possible nodes, i. e. $N \in \underset{M \in \mathcal{N}}{\operatorname{argmax}}(\Phi_M + \Pi_M)$.

Thereafter the selected node N is realized in the trees, the segment is assigned to the leaf node and the algorithm continues with the next segment.

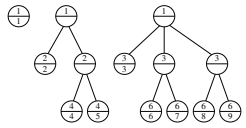
If the first of the above alternatives for the position of a segment has been selected, the segment has been put at a position where it can be left out at its first transmission, if the second one has been selected, it cannot be left out. The maximum conditions in both alternatives are used to leave better positions for the remaining segments if possible.

Figure 4.25 demonstrates the application of the algorithm to a transmission schedule using three logical channels. The alternative positions for each segment have been plotted into the figure using dashed lines.

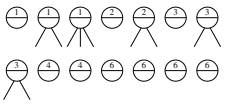
4.12.3 Improved Startup Enhancement Algorithm

The startup enhancement algorithm can even be improved further if the segments are inserted in two rounds in the transmission trees: In the first round, only segments are inserted which can be

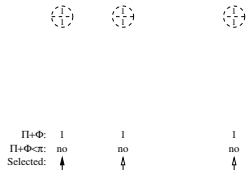
[11] It is always possible to complete the trees as the used information has been extracted from a valid construction and the periods are kept the same for each subtree. The segments can always be assigned to the leaf nodes as they are inserted in ascending order, i. e. even if a segment uses a leaf node with a lower period than originally, it is assured that the segment which previously occupied the leaf node has already been assigned to another node.



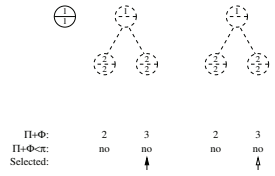
(a) Original transmission trees



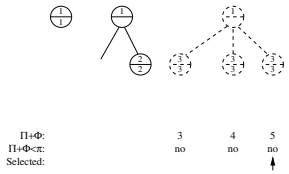
(b) Extracted information



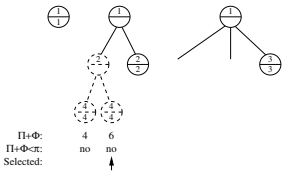
(c) Insertion of first segment



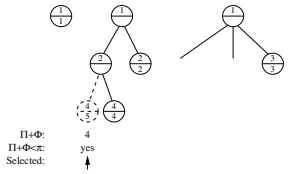
(d) Insertion of second segment



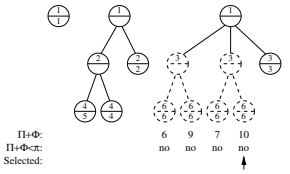
(e) Insertion of third segment



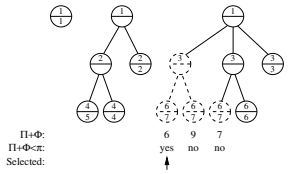
(f) Insertion of fourth segment



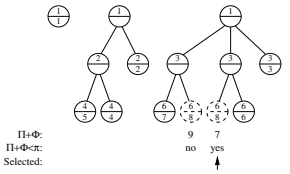
(g) Insertion of fifth segment



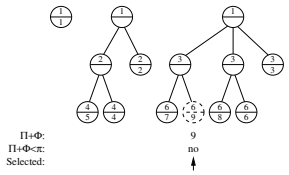
(h) Insertion of sixth segment



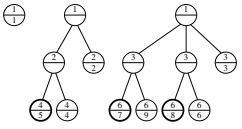
(i) Insertion of seventh segment



(j) Insertion of eighth segment



(k) Insertion of ninth segment



(l) Resulting transmission trees

Figure 4.25: Example showing functioning of the startup enhancement algorithm

positioned in such a way that they can be left out at their first transmission, i. e. the segments for which it is possible to apply the first alternative of the above description. In a second round the remaining segments are inserted. This enables the remaining segments to catch better positions and therefore increases the number of segments which can be dropped at their first transmission.

When using two rounds, it is important to observe that the remaining segments can be inserted into the transmission trees in the second round: As the segments are not inserted in ascending order any more (and therefore not in ascending order of their required periods), it is possible that segments of the first round occupy the positions in the trees which are needed for segments of the second round.

This problem can be solved if a lower bound for the period is applied when the best position for a segment is searched. To keep track of the positions which are needed for the segments in the second round, two lists have to be maintained by the algorithm: The first list contains the required periods of all segments which have been rejected in the first round, sorted in ascending order. The second list enumerates the periods of the free leaf nodes, i. e. the periods of the leaf node information pieces, sorted in ascending order, too. It is obvious that each element of the first list must be greater than or equal to its corresponding element of the second list and that these relations guarantee that all rejected segments can be inserted into the trees in the second round. To prevent that removal of an element from the second list destroys this condition, one can try which of the elements can be removed to find the allowed periods. A more simple approach is to remove the first element of the second list and match the remaining elements to the elements of the first list. The last element in the second list which violates the relation then gives the lower bound for the period for the insertion. Figure 4.26 shows an example how this lower bound search works when a position for the seventh segment of the example of figure 4.25 is searched.

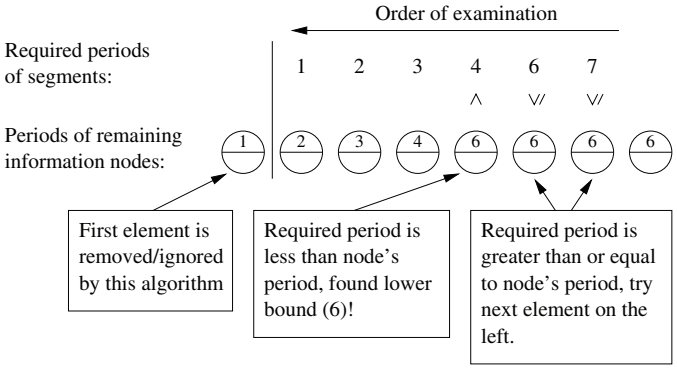
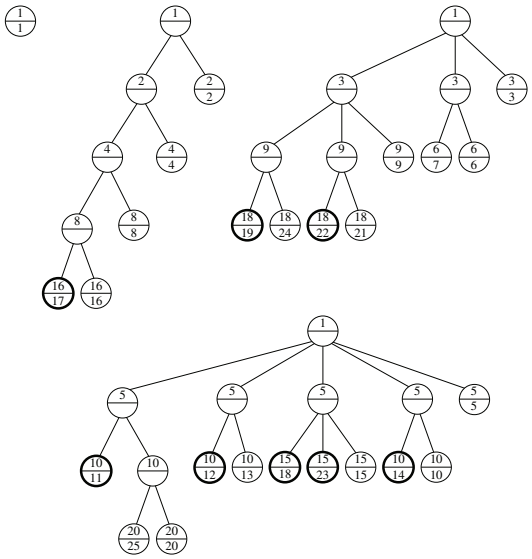
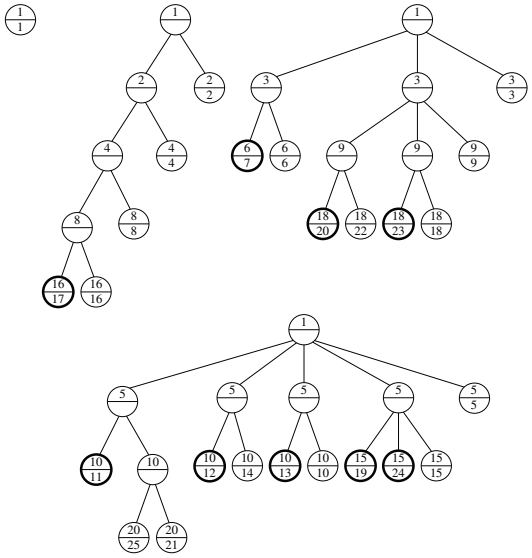


Figure 4.26: Example for determining the lower bound for a segment to be inserted

Figure 4.27 shows the results of both the simple and two rounds variants of the enhancement algorithm for a four channel transmission scheme.



(a) Resulting transmission trees of the one round enhancement algorithm; eight segments can be left out at startup



(b) Resulting transmission trees of the two round enhancement algorithm; nine segments can be left out at startup

Figure 4.27: Example showing the difference between the one-round and the two-round startup enhancement algorithm

As mentioned earlier, this startup enhancement algorithm can be applied to any frequency-based transmission scheme. For this reason, table 4.2 shows the result of the enhancement algorithms not only to the Generalized Greedy Broadcasting scheme but also for some other transmission schemes. The results show that it is possible to save up to 13 % of one media transmission through this enhancement for the Generalized Greedy Broadcasting scheme, and even more for other transmission schemes.

Broadcasting Protocol	Saved b/w by omitting segments at startup	Saved b/w when using proposed algorithm
Fast Broadcasting:		
N=4, n=15	0 %/73.3 %	73.3 %
N=6, n=63	0 %/90.5 %	90.5 %
N=8, n=255	0 %/96.9 %	96.9 %
Greedy Broadcasting:		
N=4, n=26	3.8 %	26.9 %
N=6, n=203	0.5 %	21.2 %
N=8, n=1 563	0.3 %	14.3 %
New Pagoda:		
N=4, n=26	0.0 %	30.8 %
N=6, n=172	3.5 %	27.7 %
N=8, n=1 166	2.0 %	17.4 %
Fixed Delay Pagoda Broadcasting:		
N=2, W^+=100δ, n=568	1.9 %	17.1 %
N=4, W^+=20δ, n=802	2.9 %	20.7 %
N=6, W^+=10δ, n=2 367	1.3 %	14.7 %
Generalized Greedy Broadcasting:		
N=2, W^+=100δ, n=609	0.7 %	13.0 %
N=4, W^+=20δ, n=1 028	0.0 %	9.9 %
N=6, W^+=10δ, n=4 022	0.7 %	7.8 %

Table 4.2: Amount of saved bandwidth of the first media transmission

4.12.4 Lowering the Efficiency to Improve the Startup

The reason why some transmission schemes benefit more from the startup enhancement algorithm than others (esp. than the Generalized Greedy Broadcasting scheme) is the high efficiency of these transmission schemes: The worse the efficiency of a transmission scheme is, the higher is the difference between the required period and the utilized period for the segments, and the higher this difference is, the higher is the probability to find a position where the segment can be left out in the first transmission. This raises the question if it is possible to improve the startup further by reducing the efficiency artificially, e. g. by lowering the required period for some segments when a transmission schedule is created.

It is interesting to note in this context is that the bandwidth gain when suppressing a segment transmission at startup is independent of the segment period: A segment with a high period gives the same bandwidth savings at startup as a segment with a low period, in both cases one segment transmission can be omitted at startup. This is a main difference to the bandwidth requirements of the continuous transmission where the average bandwidth requirements for the transmission of a segment are inversely proportional to the utilized period of the segment. As a consequence, it is

sufficient to decrease the period of the later segments of the transmission so the total bandwidth requirements are nearly uneffected.

Table 4.3 shows the bandwidth requirements of a transmission with 7 200 segments and a playback delay of 60 slot intervals for different artificially reductions of the segment periods. As this table shows, bandwidth increase (i. e. efficiency loss) from the reduction of periods is very minor (less than 0.1 % in most cases), the number of omissable segments can be increased significantly. Nevertheless, it is important to recognize that the efficiency loss will outweigh the additional savings for long-term transmissions. The last column in the table indicates this time in multiples of the media stream duration.

First segment with lowered period	Amount of period reduction	Average b/w requirements in multiples of media bit rate	Percentage of bandwidth increase	Number of omissable segments at startup	Percentage of omissable segments at startup	Number of transmissions where segment omissions outweigh b/w increase
—	0	4.827 136	0.00 %	221	3.1 %	—
1 200	1	4.827 880	0.02 %	305	4.2 %	15.7
1 200	2	4.828 569	0.03 %	366	5.1 %	14.1
1 200	5	4.830 575	0.07 %	502	7.0 %	11.3
1 200	10	4.834 049	0.14 %	653	9.1 %	8.7
1 200	20	4.840 500	0.28 %	857	11.9 %	6.6
2 400	1	4.827 407	0.01 %	274	3.8 %	27.2
2 400	2	4.827 677	0.01 %	324	4.5 %	26.4
2 400	5	4.828 707	0.03 %	443	6.2 %	19.6
2 400	10	4.830 060	0.06 %	570	7.9 %	16.6
2 400	20	4.832 975	0.12 %	746	10.4 %	12.5
3 600	1	4.827 271	0.00 %	262	3.6 %	42.2
3 600	2	4.827 406	0.01 %	304	4.2 %	42.7
3 600	5	4.827 858	0.01 %	387	5.4 %	31.9
3 600	10	4.828 535	0.03 %	482	6.7 %	25.9
3 600	20	4.830 194	0.06 %	624	8.7 %	18.3
4 800	1	4.827 203	0.00 %	249	3.5 %	58.0
4 800	2	4.827 271	0.00 %	275	3.8 %	55.6
4 800	5	4.827 474	0.01 %	326	4.5 %	43.1
4 800	10	4.827 924	0.02 %	389	5.4 %	29.6
4 800	20	4.828 610	0.03 %	480	6.7 %	24.4

Table 4.3: Effects of lowering segment periods artificially to enhance startup

4.13 Enhanced Termination

After enhancing the startup of media-on-demand transmissions, this section discusses the media termination. Similar to the startup, it is possible to save bandwidth by enhancing it.

4.13.1 Intention and Effects

When a transmission is to be terminated properly (allowing all ongoing transmissions to finish), it is not possible to abort the transmission immediately. Instead, all segments have to be transmitted as long as their reception was set aside in the schedule until the lastly joined receiver does not rely on the transmission any more. When this time has reached, the concerned segments need not to be transmitted any further and the saved bandwidth can be used for other purposes.

But it is still possible to improve the termination a little bit: Firstly, it is possible to stop the transmission of some segments earlier than described above: In cases where the lastly joined receiver system gets a segment twice (which is possible if the utilized period is lower than the required period), the latter transmission can be left out if the sender system can tell the receiver system in any manner that the second transmission will not take place. In other words, it is sufficient if each segment is transmitted exactly once after the time of the last possible reception.

Secondly, the order of segment transmissions can be changed when the termination is in progress: To keep the ongoing transmission running, no segment must be transmitted later than specified in the transmission schedule, but it is possible to transmit segments earlier. As segments must only be transmitted once as described above, the transmission channels have several holes which can be used this way to transmit segments in advance.

Two possible main strategies can be pursued this way: Terminate the transmission as fast as possible or release channels as fast as possible. In the first case, the last segments of the transmission can be moved to the holes, in the latter case, segments from the channel to release next are moved to the holes.

The only disadvantage of the enhanced termination approach is that the storage requirements for the receiver systems are increased: As segments have to be received earlier than planned in both described enhancements, they have to be stored longer than originally intended.

The worst case storage requirement for this scenario can be calculated the following way: If the receiver system gets one segment on each channel each slot interval while playing the media, the worst case storage requirement $M^+(t)$ at time t after joining the transmission can be calculated by

$$M^+(t) = \begin{cases} N \cdot \beta \cdot t - F^+(t) & \text{if } t \leq t_1 \\ M^+(t_1) + \frac{t-t_1}{t_2-t_1} \cdot (M^+(t_2) - M^+(t_1)) & \text{if } t_1 < t < t_2 \\ s - F^+(t) & \text{if } t \geq t_2 \end{cases} \tag{4.24}$$

where N is the number of channels of the transmission, n is the number of segments of the media stream, β is the channel bandwidth, δ the slot interval, $t_1 = \delta \cdot \left\lfloor \frac{n}{N} \right\rfloor$ the time when the receiver system receives the last n mod N segments and $t_2 = \delta \cdot \left\lceil \frac{n}{N} \right\rceil$ the time when the receiver system has received all segments.

Figure 4.28 shows the storage requirements for a constant-bit-rate transmission for different numbers of channels. It can be observed that the worst case storage requirement rises to the fraction $\frac{N-1}{N}$ of the whole media stream, thus this enhancement can only be used if the receiver systems has enough storage for nearly the whole media stream.

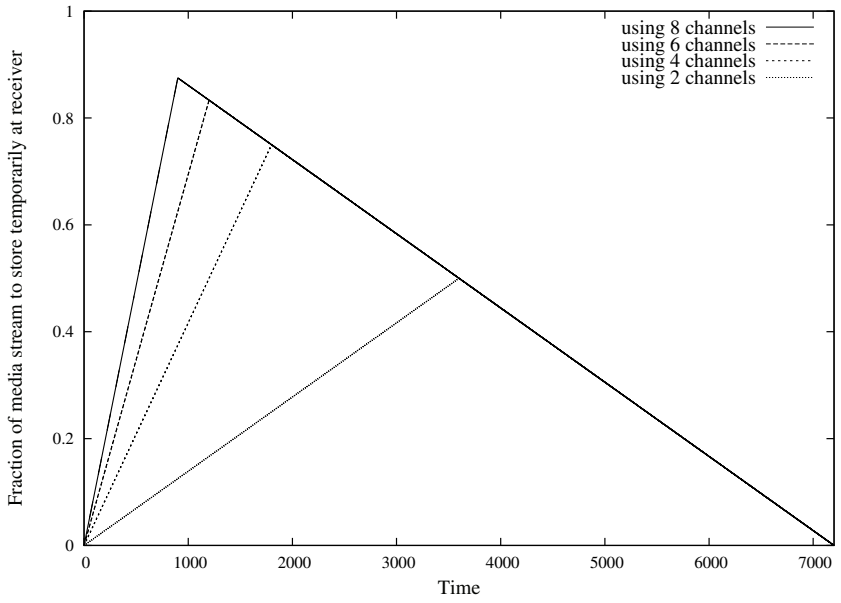


Figure 4.28: Worst case storage requirements when enhanced termination is used

4.13.2 Application to Generalized Greedy Broadcasting Scheme

The proposed enhancement is applicable to any frequency-based transmission scheme, including the Generalized Greedy Broadcasting scheme.

4.14 Seamless Media Change

When the provider decides to replace streams in its media assortment, another problem emerges: To allow active recipients to finish the reception of the replaced media stream, its transmission cannot be terminated immediately. On the other hand, the new media stream needs bandwidth for its transmissions.

4.14.1 Intention and Effects

To prevent that the bandwidth requirements increase when replacing media streams, the authors of [THYL+01] showed that the Fast Broadcasting scheme supports a seamless media change: As the Fast Broadcasting schemes transmits 2^k segments on the k-th channel and as the terminating

transmission has to send each segment once after the time of the last possible reception, channel #k becomes available after 2^k slot intervals. If the new transmission uses the enhanced startup mechanisms of the Fast Broadcasting scheme, the new transmission schedule does not need to send anything within the first $2^k - 1$ slot intervals on the k-th channel after startup, so the startup of the new media stream can commence just a single slot interval after the last possible reception of the old media stream. Figure 4.29 illustrates a seamless media change for the Fast Broadcasting scheme. The same procedure can be applied to the Harmonic Broadcasting and Staircase Broadcasting scheme.

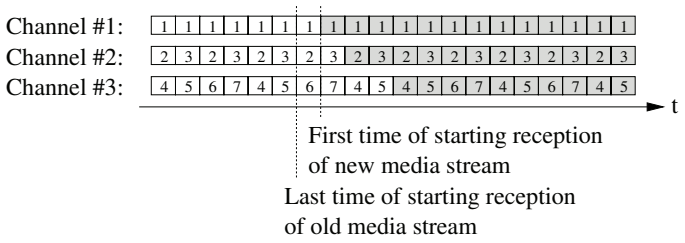


Figure 4.29: Example of a seamless media change using the Fast Broadcasting scheme

4.14.2 Application to Generalized Greedy Broadcasting Scheme

Unfortunately, other frequency-based transmission schemes do not support a seamless media change this way, nor does the Generalized Greedy Broadcasting scheme: As the terminating transmission needs to send all segments once after the time of the last possible reception, the channels become too slow available for the upcoming transmission. Thus either both transmissions have to take place in parallel (which requires additional bandwidth) or the upcoming transmission has to be delayed until the channels become available. Figures 4.30 and 4.31 show an example for a media change where the upcoming transmission is either sent simultaneously to the terminating one or is delayed until the terminating transmission finished, resp.

To shorten this gap, the proposed mechanisms for enhanced media startup (see section 4.12) and enhanced media termination (see section 4.13) can be used to overlap the two transmissions by exploiting the holes in the newly started transmission. Figures 4.32 and 4.33 show examples of a media change using only startup and termination enhancement or using overlapping additionally, resp.

Besides of these straight forward enhancements for media exchange, another approach is possible if the terminating and the upcoming media streams share the same playback function (which is nearly impossible for variable-bit-rate media streams but fulfilled for constant-bit-rate media streams of equal bandwidth, duration, playback delay, preloading and break positioning), i. e. if the media streams share the same transmission schedule:

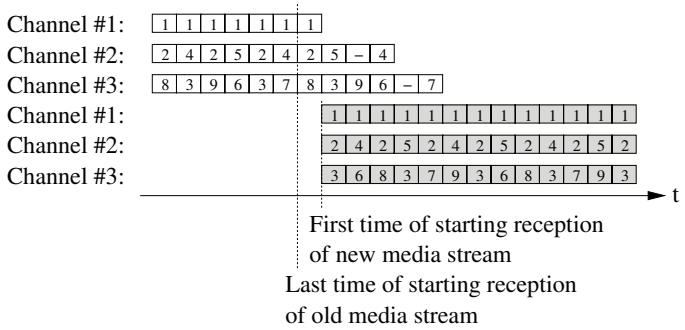


Figure 4.30: Example of a media change for the Generalized Greedy Broadcasting scheme where the terminating and the upcoming media streams are transmitted simultaneously

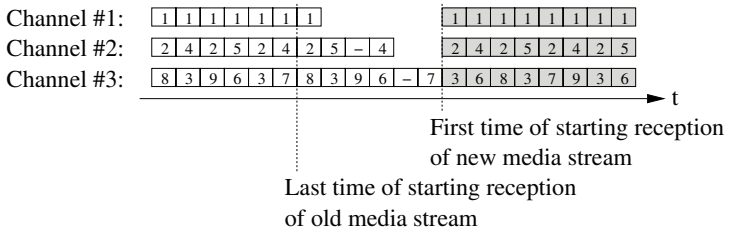


Figure 4.31: Example of a media change for the Generalized Greedy Broadcasting scheme where the upcoming media stream is delayed until the terminating media stream has released the channels

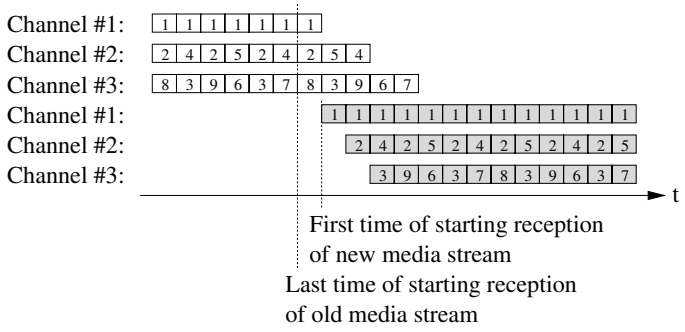


Figure 4.32: Example of a seamless media change for the Generalized Greedy Broadcasting scheme using enhanced startup and termination

If the schedule is not changed, it is possible to just replace the slot content in the old transmission with the new content as soon as it is not needed by any ongoing transmission, i. e. if it has been transmitted once after the last time for a reception of the old media stream. But as segment #i is used by the old schedule for at most π_i slot intervals after the last time for a reception of the old media stream, it is not possible for the new schedule to transmit segment #i within the first π_i slot intervals. As this is similar a requirement of live streaming (see requirement R1 in sec-

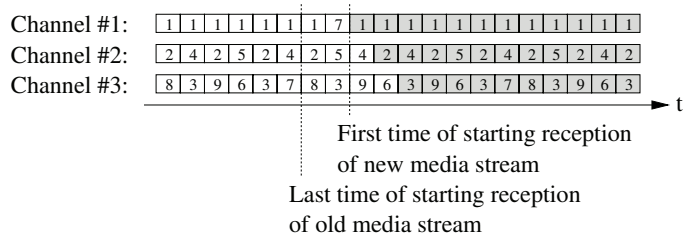


Figure 4.33: Example of an overlapping seamless media change for the Generalized Greedy Broadcasting scheme exploiting savings from startup and termination enhancement by overlapping the transmissions

tion 4.17), this problem can be solved by adding a single transmission of the whole media stream on a separate channel. Figure 4.34 presents an example for a seamless media change using this algorithm.

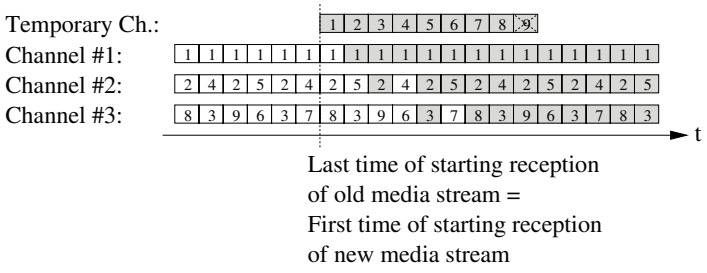


Figure 4.34: Example of an enhanced seamless media change for the Generalized Greedy Broadcasting scheme using temporarily an additional channel

4.15 Change of Media Bit Rate for Ongoing Transmissions

Instead of only starting and terminating a media-on-demand transmission, it is often desired to modify an ongoing transmission. These modifications can be necessary when the media content has to be updated (e. g. for a news transmission) or if the provider wants to change some parameters of the transmission (e. g. to reduce the bandwidth or reduce the playback delay, depending on the current request statistics or to adapt to expected intra-day request variations). Some of these changes are discussed in this and the following sections.

4.15.1 Intention and Effects

A similar problem as for media replacements occurs if the bit rate of a media stream is changed: Ongoing transmissions must not be disturbed while new transmissions should benefit from the

new media bit rate. But as the media content is identifical, some special cases can be examined to improve the performance.

When the media bit rate is changed, several cases can be distinguished how ongoing transmissions should act:

- Ongoing transmissions should be continued with the old media stream; only future requests are served using the new media bit rate:

  One solution for this case is simple but requires additional bandwidth for the time of transition: As the current transmission should continue, it has to be terminated orderly (transmitting all segments which are needed to allow a proper playback for ongoing transmissions, see section 4.13) while the new media stream transmission is started at the same time. This means that two transmissions of the same media stream (with different bit rates) take place at the same time until the first transmission is finished. As no receiver of one of these transmissions can benefit from the other transmission, this change is identical to a media content change (see section 4.14).

  If the transmission schedule can be kept (e. g. if a constant-bit-rate media stream is sent or if the bit rate change has been foreseen when the schedule has been created), the transition can be performed in the same way as it has been proposed for media content change with identical transmission schedule. In this case the bit rate of the channels has to be set to the maximum of the former and later channel bit rate during transition and one additional channel of the target media bit rate is temporarily needed.

- Ongoing transmissions can be continued with the old or the new media stream, changing back and forth at each segment boundary:

  If segments can be played independently of each other and corresponding segments of both media streams contain the same content (e. g. if segment boundaries are aligned to group-of-pictures and if each segment from either transmission contains the same frames or group-of-pictures) the change can simply be performed at the next segment boundary: The transmission schedule is simply continued but the segments are replaced with the new ones and the bandwidth of the channels is adjusted according to the new schedule. Ongoing transmissions will play segments of the old bit rate if these segments have been received earlier and segments of the new bit rate if they are received after the change.

  This is another case where no bandwidth is wasted, but difficult to achieve for compressed media streams because of the fixed assignment of frames or group-of-pictures to segments.

- Ongoing transmissions can be continued with the old media stream or the new media stream, but only a single point of change from the old to the new media stream is allowed:

This variant poses a huge problem: As a change back to the old media stream is not allowed, each ongoing transmission would have to play the new media stream after some time. This means, switching to the new bit rate causes expiry of all segments which have been received earlier for the time after the change. Thus to assure a continuous playback, the receiver system must be able to get all these segments again in due time.

If switching to the new bit rate is scheduled with a fixed delay after start of the new transmission, all segments of the new transmission must be transmitted at least once within this delay (at least for pure pro-active systems). Unfortunately, this means that a multiple of the new media stream bit rate is needed during the transition time (similar to the Staggered Broadcasting scheme), so the first solution is preferred to this one.

- Ongoing transmissions can be continued with the old media stream or the new media stream, but changes require additional data:

  For the same reasons as of the previous case, no change should be applied to ongoing transmissions and the first variant should be used instead.

Summarizing the above, if ongoing transmissions can change back and forth between the old and the new media stream, it is most efficient (and most simple) to perform the change immediately at the next segment boundary. Otherwise ongoing transmissions should continue at their current bit rate and the bit rate change is handled like a media change.

4.15.2 Application to Generalized Greedy Broadcasting Scheme

The proposed enhancement is applicable to any frequency-based transmission scheme, including the Generalized Greedy Broadcasting scheme.

4.15.3 Alternative Solution: Layered Media Encodings

An alternative solution is possible if layered encodings are used (see also section 5.2): In this case, a bit rate change can be realized by adding or removing enhancement layers. Similar to the above, this can be performed in different ways:

- If ongoing transmissions should stay at their current bit rate, enhancement layers can be added or removed as described for media startup or termination, respectively.
- If it is acceptable that ongoing transmissions will change back and forth between a different number of layers at each segment boundary, enhancement layers can be added or removed at the next segment boundary.

- If a single point of change is acceptable, it is also possible to add or remove the enhancement layers and request all ongoing transmissions to ignore the added enhancement layer or to drop data previously received for the enhancement layer, respectively.
- Changes which require additional data should still be avoided.

As the base layer (and probably some of the enhancement layers) is shared for different bit rates, layered encodings allow a more efficient media bit rate change if ongoing transitions should continue at their current bit rate. The only disadvantage of layered encodings is a slightly higher total bit rate for layered media streams compared to non-layered ones.

4.16 Change of Sender Bandwidth and Playback Delay for Ongoing Transmissions

Another possible request for modifications of ongoing transmissions is to change the playback delay and thereby the sender bandwidth.

The simplest solution for changing the bandwidth and the playback delay is to terminate the current transmission properly and start a new one (with a new transmission schedule) which utilizes the new bandwidth and playback delay. This solution is very inefficient as several segments are transmitted more often than necessary.

In the following, two better solutions are proposed to reduce the overhead of the change: For the first one, a transmission schedule is created which supports halving the playback delay as well as doubling it again later to return to the original playback delay, for the second one, a modified version of the startup enhancement algorithm is used.

4.16.1 Dynamic Channel Addition Algorithm

Due to its binary construction, the Fast Broadcasting Protocol supports a bandwidth change in a simple way [YCTY+01]: To add a channel, all segments are halved and the first half of the lowest segment of each channel is moved to the next channel, i. e. the first half of segment #1 is moved to a new channel, the first half of segment #2 to the previous position of the first half of segment #1, the first half of segment #4 to the previous position of the first half of segment #2, and so on.

For other broadcasting protocols, a more complex algorithm has to be used, e. g. the one which has been proposed in [PL03]: Similar to the Fast Broadcasting channel addition algorithm, this algorithm assumes that the worst case playback delay equals the slot interval, that a constant-bit-rate media stream is transmitted and that one channel is added to halve the playback delay. The basic idea behind this algorithm comes from the observation that it is possible to halve the playback delay by halving the slot interval with only one exception: The first half of all segments

which had been scheduled at their required period are transmitted too seldom when the playback delay is halved, their utilized period in the obtained transmission schedule is one too high. For example, if the required and the utilized period of segment #i were $\pi_i^+ = \pi_i = i$, the period of the first half of this segment is ${\pi_{i_1}}' = 2 \cdot i$ after halving the slot interval[12], but for a continuous playback with a halved playback delay, ${\pi_{i_1}^+}' = 2 \cdot i - 1$ is required. The authors of the aforementioned article solved this problem by moving these segments: The first of these segments (i. e. segment #1_1) is moved to the added channel, the second one to the position of the first segment (which provides at least a period two smaller than the old position and therefore a period which is low enough), and so on until all necessary segments have been moved.

The complete algorithm can be described as follows:

1. A new channel is allocated and all segments of the media stream transmission are divided into two parts of equal size, i. e. segment #i is divided into segments #i_1 and #i_2.
2. All segments are examined in ascending order:
   - If the utilized period of the segment is equal to the required period of the segment, the first half of the segment has to be "shifted" to a new position: Segment #1_1 is moved to the new channel (where it is transmitted with the same period as before), the next segment is moved to the previous position of segment #1_1, and so on. The previous position of the last shifted segment is left unused.
   - If the utilized period of the segment is lower than the required period of the segment, it is not modified.
3. If a segment is moved to a position where it would be sent later than in its current position (or exactly: if it is moved to a position so it is not transmitted for one period after its last transmission), this may interrupt the continuity of ongoing playbacks. To prevent this, three solutions are proposed:
   - Additional channels can be used where these segments are transmitted once.
   - As the added channel of step 1 is only used for half of the time yet, the unused positions in it can be used for the transmission of an additional repetition of the moved segment.
   - The shifting of the segment (and consequently the shifting of all following ones) can be delayed until the segment has been transmitted the next time or until one of the above solutions can be applied.

   The first solution is the simplest one but requires temporarily additional sender bandwidth and increases the bandwidth requirements of the receiver systems. The second solution is

[12] Periods are measured in multiples of the slot interval, so the period doubles if the slot interval is halved. For clearness, the variables which refer to values after halving are marked with a tick $'$.

the most elegant one, but it cannot be used in all cases as it only allows a transmission of one further segment every slot interval. The third proposed solution has the advantage that it does not require additional bandwidth, but it will prolong the duration of the transition and therefore retard the halving of the playback delay.

4. When the shifting process completed, the transmission period of segment $\#1_1$ is halved, so it is transmitted permanently on the added channel thereafter. Starting from this moment, the transmission can be joined with halved maximum playback delay.

Figure 4.35 shows an example for a dynamic addition of one channel to halve the playback delay. It also illustrates some of the problems described above and that the change may take some time to complete. During this time, intermediate versions of the transmission schedule are used and the playback delay is still unmodified. Only after the change has completed, the new playback delay can be utilized.

It is also possible to apply the bandwidth addition algorithm several times: Each time when one channel is added, the playback delay can be halved.

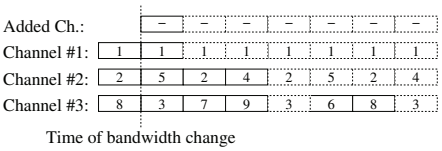
4.16.2 Dynamic Channel Releasing Algorithm

To release one channel, the algorithm is applied in reverse order: Firstly, the period of segment $\#1_1$ is doubled and then all segments are moved back to their original positions in reverse order (taking the same precautions as above when a segment is moved to a later position). When all (halved) segments have been moved to their original positions, they can be merged with their counterparts and the (in between unused) first channel can be released.

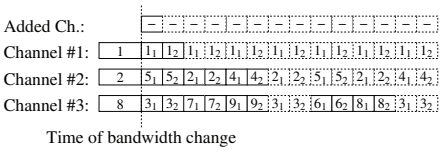
As the releasing process only reverses the adding process, it is obvious that channels can only be released if they have been added previously. Therefore, if a transmission scheme is needed where k out of n channels can be released at a later time, a transmission scheme for $n-k$ channels has to be created and k channels have to be added using the channel addition algorithm. If this step is performed before the transmission of the media stream is started, no ongoing transmissions exist which can be interrupted. Therefore, the segments can be moved in any direction by the algorithm at this time, making step 3 of the channel adding algorithm obsolete.

4.16.3 Efficiency of Channel Adding Algorithm

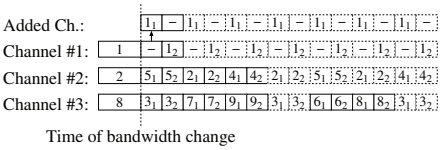
The efficiency of a transmission scheme where a channel has been added is unfortunately much lower than the efficiency of a transmission scheme which has been created for the total number of channels: Adding k channels using the channel adding algorithm increases the bandwidth by k times the channel bandwidth and halves the playback delay k times, i. e. the efficiency η' can be



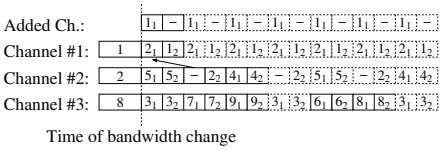
(a) Original transmission and time of bandwidth increase



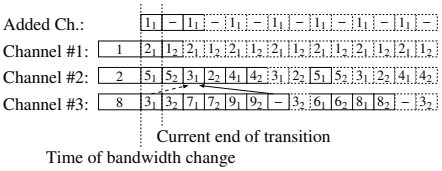
(b) Segments are halved



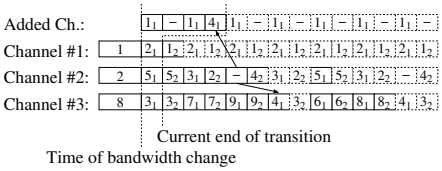
(c) Segment #1_1 is moved to the added channel



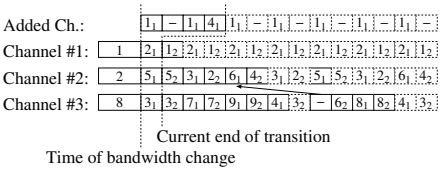
(d) Segment #2_1 is moved to the previous position of segment #1_1



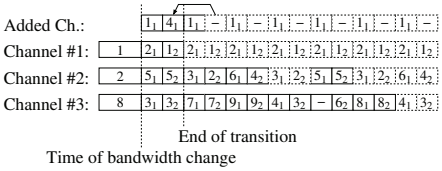
(e) Segment #3_1 would have to be moved forward to fill the previous position of segment #2_1 which would infringe ongoing playbacks; therefore it is transmitted once in its old position, which allows to move the next transmission of segment #3_1



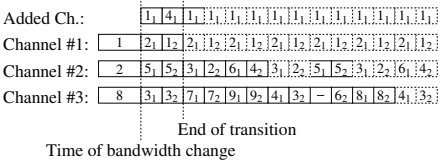
(f) Segment #4_1 would have to be moved forward, too, but here another solution is possible: It is moved to its new position and is additionally transmitted in an unused slot of the added channel



(g) Segment #6_1 is moved to the previous position of segment #4_1



(h) Segment #4_1 is moved to an earlier unused position on the added channel



(i) The transmission frequency of segment #1_1 is doubled after the transition has finished

Figure 4.35: Example showing the functioning of the dynamic channel addition algorithm

calculated from the efficiency of the original transmission schedule η by

$$\begin{aligned} \eta' &= \eta \cdot \frac{\Psi_0\left(\frac{d+W^+\cdot 2^{-k}}{\delta}\right)-\Psi_0\left(\frac{W^+\cdot 2^{-k}}{\delta}\right)}{\Psi_0\left(\frac{d+W^+}{\delta}\right)-\Psi_0\left(\frac{W^+}{\delta}\right)+k} \\ &\approx \eta \cdot \frac{\ln\left(1+\frac{d}{W^+\cdot 2^{-k}}\right)}{\ln\left(1+\frac{d}{W^+}\right)+k} \end{aligned} \tag{4.25}$$

Table 4.4 shows the efficiency for different playback delays. This table shows that the efficiency of the transmission schedule is lowered by a factor of approximately 0.903 if the playback delay is halved from ten to five minutes this way. On the other hand, the efficiency stays nearly unchanged if the playback delay is very short (e. g. near to the slot interval).

Original playback delay	New playback delay	Efficiency quotient $\frac{\eta'}{\eta}$
10 min	5 min	0.903
5 min	2.5 min	0.923
2 min	1 min	0.939
1 min	30 s	0.948
20 s	10 s	0.959
10 s	5 s	0.967
6 s	3 s	0.973
4 s	2 s	0.981
2 s	1 s	0.999985

Table 4.4: Efficiency effects of dynamic channel addition algorithm for adding one channel to a two hour media stream with one second slot intervals

4.16.4 Application to Generalized Greedy Broadcasting Scheme

To apply this algorithm to the Generalized Greedy Broadcasting scheme, it has to be generalized as the Generalized Greedy Broadcasting scheme allows to increase the playback delay independently of the segment size (see section 4.5) and even allows to create transmission schemes which adapt to a given playback function of a media stream (see sections 4.7 and 4.8). The effects of these generalizations to the bandwidth changing algorithms are proposed in this section in several steps: Firstly, the mechanisms of the above algorithm are described separately and are generalized. This is necessary as they are actually independent from each other and are not needed all at the same time in every setup. In a second step, one special case is examined: Transmission schemes which have a linear playback function (i. e. for a constant-bit-rate media stream with playback delay and partial preloading, but no other extensions). In a third step, proposals for improvements are given, e. g. ways to reduce the transition time from one schedule to another.

The algorithm which has been proposed above is comprised out of several mechanisms:

- Preparing to change the playback delay:

This step seems to be the most simply one: The playback delay is changed internally to the desired value. But as the transition to the new schedule may take some time, the receiver systems must not use the new playback delay until the transition completed.

The algorithm proposed in [PL03] only examined the process of halving the playback delay by adding one channel. This can be generalized: In the following, any reduction of the playback delay is examined. The additionally needed bandwidth in this case may be higher or lower than one channel and is also influenced by some other conditions (e. g. the amount of preloading or the playback function in case of variable-bit-rate media streams).

- Splitting segments and the slot interval:
  In the algorithm proposed in [PL03], all segments are halved in this step. This was necessary as the playback delay is equal to the slot interval in the supported transmission schemes of the described algorithm and the playback delay was to be halved.

  For the generalized case, any type of splitting is to be examined: A split into more than one subsegment may be interesting if the playback delay is reduced to a value shorter than half of the slot interval or if the desired playback delay cannot be reached accurately enough using the halved playback delay (i. e. if it is not close to a multiple of the halved slot interval).

  Instead of splitting segments into two or even more parts, it may be unnecessary to perform any splitting in other cases. For example, if the desired playback delay can be expressed as a multiple of the slot interval or if the segment size is already as small as practically possible or desired (as described in section 4.5, some bandwidth can be saved if segments are made as small as possible), further splitting is actually unnecessary or impossible.

  Strictly speaking, the bandwidth savings by splitting are the same that are gained by decoupling the playback delay and the slot interval (see section 4.5). As described there, the bandwidth requirements can be reduced by using as small segments as possible (always keeping the overhead in mind which is necessary at the receiver to identify the segments as well as the storage seek overhead which grows if the segments become smaller). Therefore, if the results of decoupling the playback delay have been considered when the transmission scheme has been generated, further splitting is neither required nor advisable.

- Moving segments:
  To allow a continuous playback with the new playback delay, some segments have to be moved to a position with a lower period. In the algorithm proposed in [PL03], the first half of all segments which had been scheduled to a position where their utilized period equals the required period had to be moved. The algorithm moved the first segment to the added channel and moved the latter segments to the released positions of the preceding moved segments. This way a queue of segments is build which are shifted by one position.

In the generalized case, it may be necessary to move more segments than described above and to move the segments in the queue by more than one position: For the most generalized case, a playback function describes the playback time for any part of the media stream (see section 4.1.2), and equation (4.22) defines how the required segment period can be calculated from a cumulative bandwidth function. As the cumulative bandwidth function includes the playback delay, a change of the playback delay results in a new cumulative bandwidth function f' and a new sequence of required segment periods $(\pi_i^{+\prime})_{i\in\{1,\dots,n\}}$. Using this sequence of required periods and the sequence of utilized periods of the old schedule, it is possible to calculate the number of positions k by which the segments have to be shifted in the queue using the following equation:

$$k = \min\left\{ j \in \{1,\dots,n\} \mid \forall i \in \{j+1,\dots,n\} : \pi_i^{+\prime} \geq \pi_{i-j} \right\} \tag{4.26}$$

The bandwidth which is additionally required in this moving process is equal to the bandwidth of the first k segment transmissions. To schedule these segments efficiently, the Generalized Greedy Broadcasting algorithm can be used again.

- Utilizing the changed playback delay:
  After the transition completed, the new playback delay can be used by the recipients.

4.16.5 Simplified Application for Basic Transmissions

As a special case, a transmission scheme which uses increased playback delay and partial preloading but no other of the previously described extensions is examined below. These schemes have two benefits: They are very simple (e. g. in contrary to variable-bit-rate transmission schemes where the required periods of the segments depend on the playback function of the media stream) and very popular (because constant-bit-rate media streams are still very common).

Additionally, it is assumed that the schedule already uses the smallest utilizable segment size to save the bandwidth which is gained by this procedure (see section 4.5) so the splitting step for segments can be skipped. (Otherwise splitting can simply be performed beforehand and independently of the playback delay and bandwidth modification.)

If an increased playback delay and/or partial preloading is used for a constant-bit-rate media stream, the required period of segment #i can be calculated by $\pi_i^+ = i - 1 + \left\lfloor \frac{W^+ + f^{-1}(P)}{\delta} \right\rfloor$ (see equation (4.17)). Similarly, to allow a continuous playback using a new playback delay $W^{+\prime}$, the period of segment #i must be less or equal to $\pi_i^{+\prime} = i - 1 + \left\lfloor \frac{W^{+\prime} + f^{-1}(P)}{\delta} \right\rfloor$. According to this, all segments #i with $\pi_i > \pi_i^{+\prime}$ have to be moved. In case of a perfect schedule, where $\pi_i = \pi_i^+$ is fulfilled for all segments #i (which is impracticable, see section 3.7.1), this means that all segments

would have to be moved by

$$\begin{aligned}
k &= \min\{j \in \{1,\dots,n\} \mid \forall i \in \{j+1,\dots,n\} : \pi_i^{+\prime} \geq \pi_{i-j}\} \\
&= \min\{j \in \{1,\dots,n\} \mid \forall i \in \{j+1,\dots,n\} : \pi_i^{+\prime} \geq \pi_{i-j}^{+}\} \\
&= \min\{j \in \{1,\dots,n\} \mid \forall i \in \{j+1,\dots,n\} : \\
&\quad i-1+\left\lfloor \tfrac{W^{+\prime}+f^{-1}(P)}{\delta} \right\rfloor \geq i-j-1+\left\lfloor \tfrac{W^{+}+f^{-1}(P)}{\delta} \right\rfloor\} \\
&= \min\{j \in \{1,\dots,n\} \mid \left\lfloor \tfrac{W^{+\prime}+f^{-1}(P)}{\delta} \right\rfloor \geq \left\lfloor \tfrac{W^{+}+f^{-1}(P)}{\delta} \right\rfloor - j\} \\
&= \left\lfloor \tfrac{W^{+}+f^{-1}(P)}{\delta} \right\rfloor - \left\lfloor \tfrac{W^{+\prime}+f^{-1}(P)}{\delta} \right\rfloor
\end{aligned} \tag{4.27}$$

positions.

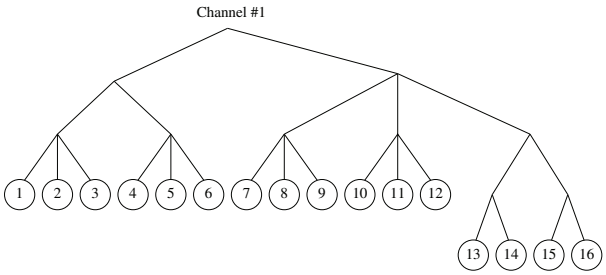
In a realistic transmission schedule, π_i is very near but often smaller than π_i^+, so only few segments have to be moved if k is very small. On the other side, if k is huge (i. e. if the playback delay is reduced considerably), most segments have to be moved.

Figure 4.36 illustrates this in an example where one channel is added to a transmission schedule which has a maximum playback delay of twelve slot intervals: Adding one channel allows to halve the maximum playback delay, so the period of all segments must be reduced by six in the resulting schedule. As no segment has been scheduled with a period six higher than needed, all segments have to be moved in this example.

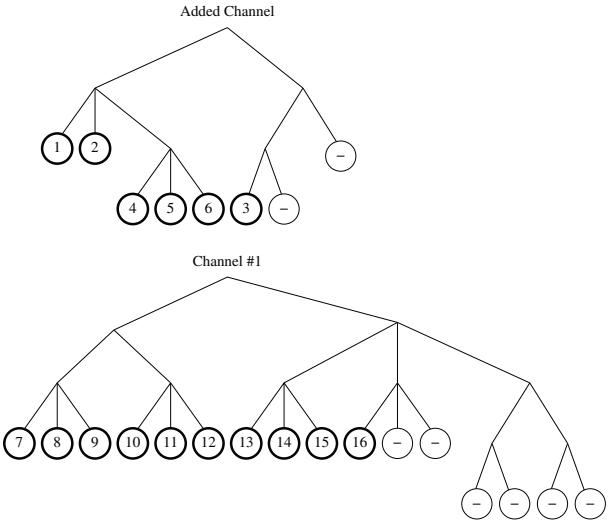
At first glance, this generalization seems to be much worse than the algorithm proposed by the authors of [PL03] where only few segments had to be moved by $k = 1$ positions in the queue. The reason for this is hidden in the slot interval: The original algorithm is based on the fact that the slot interval is as long as the (worst case) playback delay while the above proposed algorithm allows to use a playback delay of several slot intervals. In other words: The original algorithm benefited from the fact that the schedule was indeed very inefficient compared to the schedule using the short slot interval.

This insight can be used to lower the number of segments which have to be moved: If segments are intentionally scheduled in such a way that their utilized period is k lower than their required period, these segments are not involved in the moving process. The bandwidth loss can be controlled much more precisely than by using a longer slot interval: As the first segments need the most bandwidth, they can be scheduled as good as possible, while only the latter segments are excluded from the moving process by lowering their period requirements artificially for the scheduling process. If several bandwidth changes are intended, it is possible to adjust the periods for the shortest of the playback delays. Alternatively, it is even possible to reduce the period of different numbers of segments for each planned playback delay.

The efficiency loss by decreasing the period of the latter part of a media stream is exactly the same as the efficiency gain that has been described for the insertion of breaks into a stream



(a) Original transmission schedule before bandwidth change



(b) Resulting transmission schedule after halving the playback delay

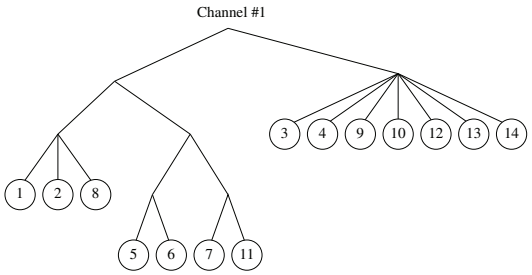
Figure 4.36: Example of a dynamic bandwidth change for a transmission scheme using a playback delay of twelve slot intervals

transmission. This is evident as decreasing the period of the latter part equals to moving this part to an earlier position in the playback, thus it can be contemplated as a negative break.

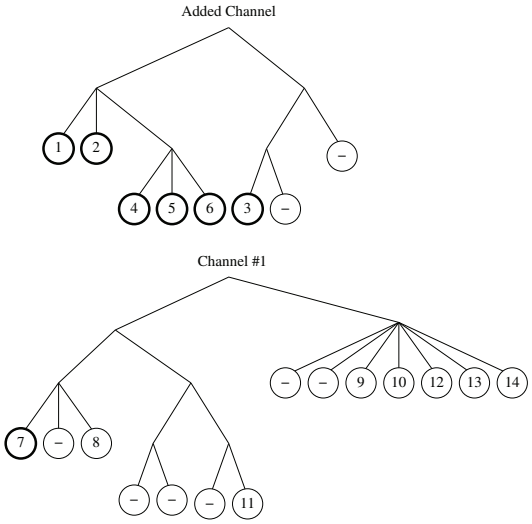
Figure 4.37 illustrates this in a similar setup as of the previous example: The original maximum playback delay is twelve slot intervals and it should be halved by adding a channel. In contrast to the previous example, segments #8 and higher have been scheduled with a period six lower than necessary. (Therefore, only fourteen instead of sixteen segments fit into the schedule.) When the playback delay is halved in this case, only the first seven segments have to be moved.

4.16.6 Alternative Solution: Transition between different Transmission Schedules

Instead of adding one channel and moving segments in a queue-like fashion to transit from the old schedule to the new one, another solution is presented in this subsection. This approach is based on the simple idea to transmit data for two different transmission schemes (one utilizing the old playback delay and one for the new playback delay) in parallel, using the enhanced termination



(a) Original transmission schedule before bandwidth change



(b) Resulting transmission schedule after halving the playback delay

Figure 4.37: Example of a dynamic bandwidth change for a transmission scheme using a playback delay of twelve slot intervals, but with a reduced period for segments #8 and further

algorithm (see section 4.13) for the transmission using the old playback delay and a modified version of the enhanced startup algorithm (see section 4.12) for the transmission which provides the new playback delay:

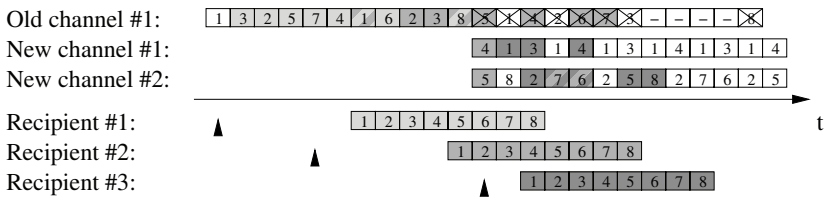
As described in section 4.12 for enhanced startup, the new schedule (i. e. the schedule which utilizes the new playback delay) is disassembled and reassembled afterwards to get a transmission scheme where segment transmissions can be left out. In case of a transition from one transmission schedule to another, the upcoming transmission can even benefit from the old schedule and omit some more segment transmissions. Thus, the rule for selecting the position for the segments can be changed in this case:

1. If the list contains positions where the next transmission of the segment does not occur later than the time at which it has to be sent the next time according to the old schedule, the position with the latest of these transmissions is selected.
2. Otherwise the position with the latest transmission at all is used and the segment has to be

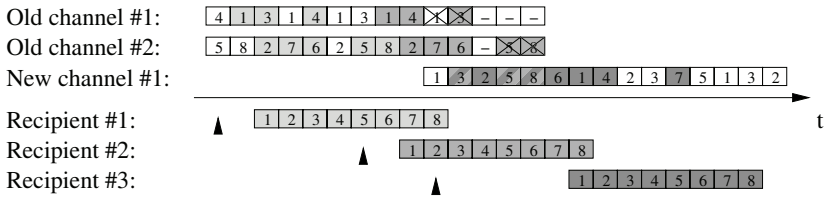
transmitted once additionally not later than the time at which the segment has to be sent according to the old schedule.

Similar to the enhanced startup, the results can be improved by using two rounds for the reassembly.

Figure 4.38 shows examples of a playback delay change of a media stream with eight segments, from six slot intervals to two intervals and vice versa. For the change from six to two intervals, a transition from a one-channel transmission to a two-channel transmission is performed, and due to the modified enhanced startup, the old transmission can be terminated immediately. For the reverse direction, two segments have to be transmitted once before the old transmission can be terminated.



(a) Transition from a schedule with a playback delay of six slot intervals to a schedule with two slot intervals



(b) Transition from a schedule with a playback delay of two slot intervals to a schedule with six slot intervals

Figure 4.38: Examples of a bandwidth change using enhanced transitions between two schedules

4.16.7 Comparison

Both of these procedures, the channel adding algorithm and the transition between different transmission schemes, have their own benefits: The advantage of the former channel adding algorithm is that the bandwidth change can be performed without any further additional bandwidth resources. As a disadvantage, the transitions may take a long time. During these transition times, the transmission consumes the higher of the bandwidths of the two schedules without granting the lower playback delay. Thus the schedule is very inefficient while the transition takes place (e. g. the efficiency is lowered to about $89,88\,\%$ if the halved playback delay is $\frac{1}{7200}$ of the media stream duration and about $71,95\,\%$ if the halved playback delay is $\frac{1}{12}$ of the media stream duration[13]).

[13]These fractions equal to a playback delay of one second and 10 minutes of a two hour media stream.

Additionally, the transition to a different transmission scheme has the advantage that it is based on more efficient transmission schedules.

To make the schemes comparable, additional channels are provided for both algorithms which are used to transmit segments when they do not fit in the transition schedule. Providing these additional channels, both algorithms can perform the playback delay change immediately and their efficiency can be measured by examining the amount of additionally transmitted segments and the efficiency of the used schedules. For the comparison, a constant-bit-rate media stream of 7 200 segments is assumed.

Table 4.5 shows the fraction of the media stream which has to be transmitted on additional channels by the two algorithms when performing a change from one playback delay to another. This fraction varies between 35.47 % and 64.63 % in this example when the playback delay is changed using the segment shifting algorithm and between 3.47 % and 28.17 % when the media transition algorithm is used.

In addition to the amount of additionally needed bandwidth during transition, the efficiency of schedule after changing the playback delay is also important because it has long-term effects on the needed bandwidth. Table 4.6 shows the efficiency of the resulting transmission schemes after reducing the playback delay from 900 slot intervals to value between 1 and 240 slot intervals for both algorithms. While the efficiency of the schedules of the segment shifting algorithm drops to values below 80 % when performing significant changes of the playback delay, the efficiency for the media transition algorithm stays always above 99 %.

Due to these results, the media transition algorithm should be preferred in any case as it needs to transmit much less segments on the additional channel (in some cases about only one tenth of the amount of the shifting algorithm) and grants a much better transmission schedules at the same time.

As described earlier, the number of segments which have to be moved by the shifting algorithm can be reduced by decreasing the periods artificially. But besides lowering the number of segments to move, this reduces the efficiency of the transmission schedules further, so the gap between the two algorithms cannot be closed this way. (Lowering the periods shows also positive effects for the adjusting algorithm.)

This means that the adjusting algorithm should be preferred in all cases where possible (i. e. where additional bandwidth for the time of the transition is available). The only disadvantage of this procedure is that the receiver systems must be able to receive the additional channels, i. e. the necessary receiver bandwidth is increased for the duration of the transition for all ongoing receptions, too.

Another advantage of the above proposed adjusting transmission schedule transition algorithm is that this procedure allows to transit between arbitrary schedules for the same media stream:

New Playback Delay	Old Playback Delay (in slot intervals)					
	1	4	15	60	240	900
1	—	35.47 %	45.60 %	46.19 %	43.32 %	36.33 %
4	64.51 %	—	44.68 %	46.22 %	42.38 %	35.29 %
15	54.13 %	55.24 %	—	44.08 %	46.18 %	35.53 %
60	53.75 %	52.82 %	55.69 %	—	44.56 %	35.94 %
240	56.65 %	57.57 %	53.56 %	54.58 %	—	39.85 %
900	63.63 %	64.63 %	64.46 %	63.96 %	60.11 %	—

(a) Amount of a 7 200 segment media stream which has to be transmitted on additional channels when the playback delay is changed using the shifting algorithm

New Playback Delay	Old Playback Delay (in slot intervals)					
	1	4	15	60	240	900
1	—	10.08 %	9.01 %	8.31 %	4.81 %	3.47 %
4	8.13 %	—	9.01 %	7.14 %	4.31 %	3.69 %
15	9.06 %	11.44 %	—	9.65 %	5.40 %	3.58 %
60	10.81 %	12.42 %	11.44 %	—	5.85 %	4.50 %
240	14.46 %	16.19 %	15.14 %	14.17 %	—	5.56 %
900	26.89 %	28.17 %	26.58 %	26.83 %	21.01 %	—

(b) Amount of a 7 200 segment media stream which has to be transmitted on additional channels when the playback delay is changed using the adjusting algorithm

Table 4.5: Comparison of the two proposed algorithms for changing the bandwidth of an ongoing transmission

Playback Delay	1	4	15	60	240	900
Efficiency	79.85 %	77.46 %	77.96 %	80.46 %	86.00 %	99.63 %

(a) Efficiency of transmission schemes which are gained by shifting algorithm, starting with the transmission scheme for a playback delay of 900 slot intervals

Playback Delay	1	4	15	60	240	900
Efficiency	99.66 %	99.54 %	99.48 %	99.27 %	99.52 %	99.63 %

(b) Efficiency of transmission schemes which are gained by adjusting algorithm, starting with the transmission scheme for a playback delay of 900 slot intervals

Table 4.6: Comparison of the efficiency of the transmission schemes used by the two proposed algorithms

Besides of using the algorithm for changing the playback delay, it is also possible to insert, remove or modify breaks in the transmission or to change the amount of preloading.

4.17 Live Transmissions

Live streaming means the ability to capture a media stream and transmit it immediately after capturing. This imposes additional requirements to the transmission scheme as no data can be transmitted in advance, i. e. before it has been captured.

Unfortunately, nearly all of the broadcasting schemes assume that the media stream is completely available at the beginning of the transmission, thus most of them cannot be used for live transmissions originally.

4.17.1 Intention and Effects

In order to modify transmission schemes for live transmission support, it is important to examine the characteristics of a broadcasting scheme which supports live streaming. The authors of [YCTY+01, YYT03] summarized these characteristics in form of three requirements, but as one of these requirements (the transfer rate for the segments must not be higher than the media production rate) cannot be applied to variable-bit-rate media streams, a modified version of these requirements is presented here:

R1. The transmission schedule must not demand to broadcast media data before it has been captured.
If a transmission schedule requires transmitting data which has not yet been captured, it cannot be sent at this moment. Delaying the transmission until it has been captured may exceed the available bandwidth and sending it at its next planned transmission time may be too late for a continuous playback at the receiver system.

R2. The transmission scheme must tolerate the varying length of the live program.
As the available bandwidth is limited in most cases, the transmission schedule cannot request more bandwidth if the media becomes longer than expected. Instead, other parameters (e. g. playback delay) should be changed in this case.

The first requirement is satisfied only by Harmonic Broadcasting, Staircase Broadcasting and Fast Broadcasting (with minor changes as shown in [YCTY+01]), but by increasing the bandwidth temporarily (for the duration of one transmission of the media stream) for one additional transmission of the live media stream as described in [YYT03], it is possible to fulfill this requirement for any frequency-based transmission scheme, too.

Figure 4.39 shows an example of a frequency-based transmission where an additional live channel has been added. In this figure, segments which cannot be transmitted in the original schedule (as they have not yet been captured) have been marked with a cross and segment transmissions which can be omitted (because they have already been sent on the live channel) have been marked with a slash.

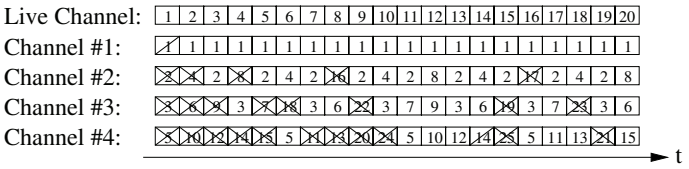


Figure 4.39: Example of a live media transmission using the Greedy Broadcasting scheme and an additional live channel

To fulfill requirement R2, it is necessary to reduce the total bandwidth requirements of an ongoing transmission dynamically, so that the gained bandwidth can be used for a prolongation of the transmission as necessary. This can be achieved by increasing the playback delay as described in section 4.16. Unfortunately, this either needs additional bandwidth for the time of the transition to the longer playback delay or takes some time, thus a bandwidth change has to be considered timely in advance.

If additional bandwidth for an extension is available, a new channel can be allocated and the next segments scheduled onto it. The transmission schedule created this way is expected to be slightly less efficient than one which has been created initially for the increased bandwidth but provides still the same playback delay.

Similar but not as strict requirements apply for near-live streaming, where a short delay between capture and transmission is used. This delay increases the number of duplicate segments of the live channel and the ordinary transmission schedule and therefore can be used to lower the additionally needed bandwidth slightly.

4.17.2 Application to Generalized Greedy Broadcasting Scheme

Using a live channel to fulfill requirement R1 can also be used for the general case in conjunction with the Generalized Greedy Broadcasting scheme with the following constrictions:

- Similar to above, the media stream duration is not always known in advance. Therefore, the transmission schedule may use too much channels if the duration is overestimated and may be less efficient if an additional channel is added in case of an underestimation.
- If the cumulative bandwidth function is not known in advance (e. g. if the media stream uses a variable-bit-rate encoding or if the location of breaks is not known in advance), the schedule cannot be generated in advance at all. Instead, each segment has to be added in real-time into the transmission schedule using the Generalized Greedy Broadcasting scheduler algorithm. This means that it is not possible to try different weights in the Generalized Greedy Broadcasting scheduler algorithm to improve the efficiency of the schedule. Another consequence is that it is not possible to calculate the needed bandwidth for a given

media stream duration in advance as the bandwidth depends on the cumulative bandwidth functions.

- In case of a variable-bit-rate media stream, the live channel must be able to transmit the variable-bit-rate media stream. Due to the live constraint, no smoothing is possible to eliminate bandwidth peaks.

To reduce the impacts of these problems, the following enhancements can be used:

- When the live transmission ends (i. e. the media stream is completely available at the sender system), it is possible to create a better transmission schedule and migrate to this one using the transition algorithm for different playback delays which has been presented in section 4.16.6. This may be esp. beneficial if the efficiency of the transmission schedule is low as a result of using a single weight combination for the Generalized Greedy Broadcasting schedule algorithm when generating a schedule for variable-bit-rate live media streams.
- To reduce the number of segments which have to be transmitted additionally due to the live channel transmission, another modified version of the startup enhancement algorithm (see section 4.12) can be used. The rule for selecting the best node N for a segment #i from the list of all possible nodes $\mathcal{N}$ is changed as follows:

  Let $\mathcal{P} = \left\{ N \in \mathcal{N} \;\middle|\; \Phi_N + 2 \cdot \Pi_N < 2 \cdot \pi_i^+ \right\}$ be the list of segment positions so that segment #i can be placed in such a way that its first *two* transmissions can be omitted.

  1. If $\mathcal{P} \neq \varnothing$, select (one of) the node(s) of this set which provides the latest segment transmission, i. e. $N \in \underset{M \in \mathcal{P}}{\operatorname{argmax}}(\Phi_M + 2 \cdot \Pi_M)$.
  2. If $\mathcal{P} = \varnothing$, select the node with the latest segment transmission of all possible nodes, i. e. $N \in \underset{M \in \mathcal{N}}{\operatorname{argmax}}(\Phi_M + 2 \cdot \Pi_M)$.

  These rules are similarly designed as the ones in section 4.12 with the exception that they exploit the segment transmission on the live channel: As each segment is transmitted on the live channel at its latest possible time (i. e. at $t = \delta \cdot \pi_i^+$), the next segment transmission is not needed before $2 \cdot \delta \cdot \pi_i^+$. This allows to omit the first segment transmission in all cases and the second one if it occurs before slot $2 \cdot \pi_i^+$ (which is the case if the first of the above two rules applies).
- If a near-live transmission is sufficient (i. e. a delay between capturing and transmission can be accepted), a limited smoothing is possible which reduces the bandwidth peaks of the live channel a bit.

Summarizing the above, the Generalized Greedy Broadcasting scheme allows live transmissions if an additional live channel is temporarily added. Problems like an unknown media stream

duration, a minor efficiency loss when generating the schedule at real-time, the additional amount of bandwidth for the live channel and the varying of the bandwidth of the live channel can be solved very efficiently with the Generalized Greedy Broadcasting scheme.

4.18 Support for Receiver Systems with Limited Bandwidth

Most of the proposed segmenting transmission schemes benefit from the fact that segment transmissions can be received by several receiver systems if the systems are already listening to the transmission. But this requires that all receiver systems are able to receive many segments at once and that they are able to store them temporarily. As this causes high bandwidth and storage requirements for the receiver systems, it is important to examine these requirements and search for alternative solutions. In the following of this section, the bandwidth requirements are examined and, in the next section, an inspection of the storage requirements is given.

4.18.1 Intention and Effects

For most of the presented transmission schemes (and nearly all of the frequency-based schemes), the receiver systems must be able to receive all data which is sent by the sender system, i. e. the receive bandwidth requirements of the receiver systems are equal to the send bandwidth requirements of a sender system for one media stream. But as all segments must only be received once, the bandwidth requirements of the receiver system decrease throughout the transmission. Figure 4.40 shows how long the channels of a transmission have to be joined by two recipients which joined the reception at different times and thus the bandwidth requirements of the two recipients. Because of the different join times, the bandwidth requirements of the two recipients differ slightly. Figure 4.41 shows this for a more simple example for two receivers: The maximum bandwidth requirements for recipient #1 is three times the media bit rate while recipient #2 only has to receive data of twice the media bit rate, and the bandwidth requirements differ mainly during the transmission for these recipients.

If the bandwidth of the receiver system does not allow a reception of all channels of a media transmission at once, the sender system can accommodate the transmission schedule to allow a media reception by these recipients:

The basic idea here is that the receiver system uses a sliding window for the channels it receives: If the receiver system can only receive R out of N channels at once, it first receives the first R channels. When the first of these channels is not needed any more, it is dropped and the next channel is joined. This is continued until all channels have been joined. In other words, the receiver system first receives channels $\#1, \ldots, \#R$, then $\#2, \ldots, \#(R+1)$, and so on, until finally the channels $\#(N-R+1), \ldots, \#N$ are received.

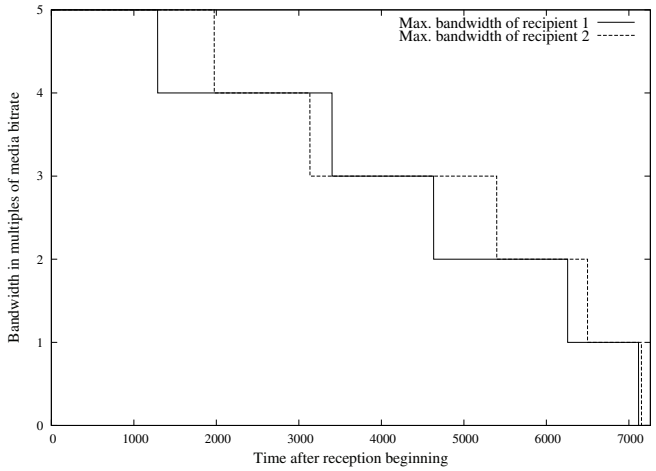


Figure 4.40: Max. bandwidth requirements of two recipients of a Generalized Greedy Broadcasting scheme transmission with 7 200 segments and a playback delay of 60 slot intervals

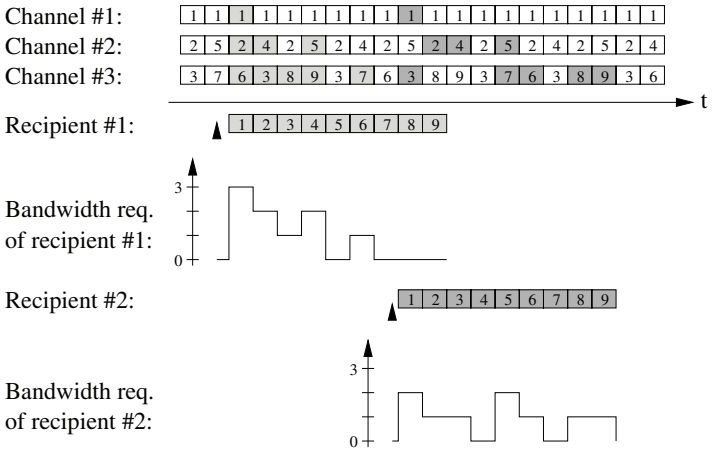


Figure 4.41: Max. bandwidth requirements of two recipients of a Generalized Greedy Broadcasting scheme transmission

When the transmission schedule for a channel $\#i > R$ for such a system is created, one has to consider that the channel is joined with the latency J_i by the recipients. This latency can be calculated from the join latency J_{i-R} of the left channel $\#(i-R)$, the (worst case) channel reception duration (which is equal to the maximum segment period P_{i-R} of this channel times the slot interval δ) and the time which is required for leaving and joining channels J:

$$
\begin{aligned}
J_i &= \begin{cases} J_{i-R} + P_{i-R} \cdot \delta + J & \text{if } i > R, \\ 0 & \text{if } i \leq R \end{cases} \\
&= \sum_{j=1}^{\lfloor \frac{i-1}{R} \rfloor} \left(P_{i-j \cdot R} \cdot \delta + J \right) \\
P_i &= \max \{ \pi_j | 1 \leq j \leq n \wedge \kappa_j = i \}
\end{aligned}
\tag{4.28}
$$

This idea (without an extra channel leave/join latency) has been studied in [PL00] for the Fast Broadcasting and New Pagoda Broadcasting schemes and gives good results in many cases.

4.18.2 Application to Generalized Greedy Broadcasting Scheme

This approach can be applied nearly unmodified to the Generalized Greedy Broadcasting scheme. Similar to the procedure in [PL00], best results are achieved by this approach for the Generalized Greedy Broadcasting scheme if the segments on the first $N - R$ channels are scheduled on one channel after another. That is the first segments are scheduled on the first channel, when it is full the next channel is filled, and so on until $N - R$ channels have been filled. The last R channels can be filled at once.

This means that the scheduling algorithm of the Generalized Greedy Broadcasting scheme has to be adjusted slightly:

- To fill one channel after another, the Generalized Greedy scheduler algorithm is applied to a single channel only. When the channel is full, the Generalized Greedy scheduler algorithm is called again for the next channel, supplying the list of remaining segments to the scheduler. This is continued until $N - R$ channels have been filled. For the remaining R channels, the Generalized Greedy scheduler algorithm is called a last time and the channels are filled at once with the remaining segments.
- When searching for a node in the transmission trees which can be used for the transmission of a segment, the join delay of the channel has to be taken into account: To allow a continuous playback at the receiver side, the join delay has to be subtracted from the playback time when calculating the required period of the segment, i. e. the needed period of segment #i on channel #j can be calculated by

$$
\pi^+_{i,j} = \left\lfloor \frac{F^{-1}((i-1) \cdot \sigma) - J_j}{\delta} \right\rfloor
\tag{4.29}
$$

  if an inverse playback function F^{-1} is used and immediate segment reception is possible (see section 4.9). Additionally, the efficiency of the resulting transmission schedule is increased if the quality function for the search of the best fitting node for a segment takes the join

latency of the node into account:

$$\begin{aligned} Q^{I}_{i,N} &= \frac{\pi^{+}_{i,K_N} \bmod \Pi_N}{\Pi_N} \\ Q^{II}_{i,N} &= \frac{\Pi_N}{\pi^{+}_{i,K_N}} \\ Q^{III}_{i,N} &= \frac{J_{K_N}}{\pi^{+}_{i,K_N} \cdot \delta} \\ Q_{i,N} &= (1 - W^{II} - W^{III}) \cdot Q^{I}_{i,N} + W^{II} \cdot Q^{II}_{i,N} + W^{III} \cdot Q^{III}_{i,N} \end{aligned} \tag{4.30}$$

But when this approach is applied to the Generalized Greedy Broadcasting scheme, the results sometimes lag behind the ones of the New Pagoda Broadcasting scheme, and in other cases they are much behind the theoretical limit for this setup. The reason for this penalty is that the Generalized Greedy Broadcasting scheme is too efficient: The Generalized Greedy Broadcasting scheme manages to put more segments in the first channels than the New Pagoda Broadcasting scheme as it uses higher segment periods. As a result, the former channels must be joined for a longer time due to the higher periods. This increases the channel join latencies of the later channels which thereby loose much of their efficiency.

The solution is to lower the efficiency of the Generalized Greedy Broadcasting scheme for the first channels to improve the overall efficiency: If a segment period limit P^{+}_{i} is introduced for channel #i, the duration for reception of this channel can be shortened. As the receiver is able to receive R channels at once, this limit is only needed for the first $N - R$ channels, for the last R channels no limit is required nor beneficial.

To calculate the optimal segment period limit for a channel, the remainder of this section has to make the additional assumption that each segment can always be scheduled with its maximum period (or the segment period limit of the channel, whichever is lower). This is normally not true as nearly any transmission scheme schedules some segments at a lower segment period than they require in order to fit them into the transmission schedule. As a minor compensation, the channel bandwidth is lowered by 1 % for these calculations, which should reflect that the Generalized Greedy Broadcasting scheme operates at an efficiency around 99 %.

Using these assumptions, the consumed bandwidth of a segment #i on channel #κ_i can be calculated as $\max\left\{\frac{1}{\pi^{+}_{i}}, \frac{1}{P^{+}_{\kappa_i}}\right\}$. As the bandwidth of each channel is limited, the following relation has to be fulfilled at the same time:

$$0.99 \geq \begin{cases} \sum\limits_{i=x_{k-1}+1}^{x_k} \max\left\{\frac{1}{\pi^{+}_{i}}, \frac{1}{P^{+}_{k}}\right\} & \text{if } k \leq N - R \\ \sum\limits_{i=x_{k-1}+1}^{x_k} \frac{1}{\pi^{+}_{i}} & \text{if } k > N - R \end{cases} \tag{4.31}$$

where x_k is the index of the last segment which can be transmitted on channel #k. (x_0 is assumed to be 0 to omit an extra case for $k = 1$.)

To get the best efficiency, the total number of segments $n = x_N$ has to be maximized by selecting the optimal set of channel period limits P_i^+ for all channels $\#i \leq N - R$. The search for the best set of channel period limits can be performed using a brute-force algorithm which tries different limits and calculates $n = x_N$ using the above relation.

For example, for a constant-bit-rate media stream or 7 200 segments, a playback delay of 60 slot intervals and a receiver bandwidth limit of twice the media bit rate, the Fixed Delay Pagoda Broadcasting scheme needs a total bandwidth of 6.759 times the media bit rate. Without application of period limits, the Generalized Greedy Broadcasting scheme needs 6.596 times the media bit rate (i. e. about 2.41 % less than the Fixed Delay Pagoda Broadcasting scheme) and 6.282 times the media bit rate if the period limits 111, 302, 575, 1201 and 2716 are applied to the first five channels, respectively (about 7.05 % less than the Fixed Delay Pagoda Broadcasting scheme).

4.19 Support for Receiver Systems with Limited Storage

Similar to a limitation of the receiver bandwidth, receiver systems may be short of storage for buffering segments for later playback. This leads to two new topics which needs further examination: How much storage is required for a successful reception and playback of a media stream for a given transmission schedule and how can the memory requirements be lowered for poorly equipped receiver systems? In the following of this section, two algorithms for calculating the memory requirements are proposed, followed by a mechanism to lower the storage requirements.

4.19.1 Exact Calculation of Storage Requirements for Frequency-Based Transmission Schemes

The calculation of the storage requirements for a frequency-based transmission schedule has turned out to be very complicated: As the storage requirements depend on the exact time when the recipient joined the transmission, the brute-force approach would demand to calculate the requirements for every distinguishable join time. Although it is very simple to calculate the exact storage requirements for a given joining time (by simulating the transmission), the number of simulations which have to be performed this way equals the period of the whole transmission schedule. The period of the whole transmission schedule equals the least common multiple of all segment periods of the transmission schedule and is very high for most transmission schedules[14]: For example, the period of a transmission schedule of the Generalized Greedy Broadcasting scheme for 7 200 segments and a maximum playback delay of 60 slot intervals is about $5 \cdot 10^{44}$.

The following algorithm circumvents this brute-force examination by splitting the transmission schedule into independent sub-schedules: Two (or several) sub-schedules are considered to be

[14]The most prominent exceptions is Fast Broadcasting where the transmission schedule has a period of 2^{N-1}.

independent if the set of prime factors of all periods of the segments for each of the sub-schedules are pairwise disjoint. For example, if the schedule contains the segments with periods from 2 to 9, three independent sub-schedules can be created: One sub-schedule contains the segment with the period 5, one sub-schedule the segment with the period 7 and one sub-schedule the remaining segments 2, 3, 4, 6, 8 and 9.

The advantage of independent sub-schedules is that their storage requirements can be calculated independently from each other and be added afterwards: As the periods of all sub-schedules have no common factors, the worst case of all sub-schedules can be reached at the same time and the storage requirements must therefore be added to find the maximum storage requirements of the whole schedule. The other point is that these sub-schedules have much lower total periods than the complete schedule: As the total period equals the least common multiple of the segment periods of the schedule, the product of the total periods of the sub-schedules equals the total period of the complete schedule.

Unfortunately, a large schedule can only be dissected very seldom into independent sub-schedules: Even the simple 3-channel Greedy schedule contains the periods 1, 2, 3, 4 and 6 which cannot be dissected into disjoint sets of prime factors. For this reason, a third mechanism is required: The schedule needs to be de-factorized.

De-factorization means that a prime factor p of a segment period of a schedule is selected and all possible values $q \in \{0, \ldots, p-1\}$ for this factor are examined consecutively. For example, if one segment of the transmission schedule has a period which is 3 or a multiple of 3, the transmission schedule is examined once for receiver systems with $s \bmod 3 = 0$, once for receiver systems with $s \bmod 3 = 1$ and once for system with $s \bmod 3 = 2$, where s denotes the time when the receiver systems joined the transmission, measured in slot intervals relative to the beginning of the media stream transmission. For each value of the prime factor, each segment falls into one of the following categories:

1. If the segment is never transmitted for the selected de-factorization combination, the storage requirement is left unchanged and the segment needs not to be considered in any further examinations, too.
2. If the segment is always transmitted for the selected de-factorization combination, the storage requirement is increased by one and the segment must not be considered in any further examinations.
3. Otherwise the segment is left in the schedule for further examination.

This de-factorization fulfills two purposes: The schedule is shortened as some of the segments can be dropped (first two cases) and for the remaining segments, fewer prime factors have to be considered for further examination if the segment period used this prime factor. In total, it is

more probable that the schedule can be dissected into independent sub-schedules after some prime factors have been removed this way.

De-factorization itself does not reduce the number of combinations which need to be examined as all possible values for each prime factor would have to be gone over. But if the above two procedures, de-factorization and dissection into independent sub-schedules, are used alternatingly, the size of the examined sub-schedules is reduced further and further because de-factorization removes the dependencies between segments step-by-step so the dissection can split the schedules into smaller sub-schedules. This process is continued until the sub-schedules only contain a single segment (in which case the worst case storage requirement of the sub-schedule is one) or are empty (and therefore need no storage).

To implement the algorithm described above, some additional formulas are needed which are provided here:

- Let t be the time within the playback (measured in slot intervals) and i be the index of the examined segment. To check if segment #i has already been played, the following relation can be checked:

$$t \geq \pi_i^+ \tag{4.32}$$

- Let additionally s be the time of joining the transmission after the schedule has been started (measured in slot intervals, too). To check which recipients (designated by their joining time s) were able to receive segment #i, the following relation can be checked:

$$(s+t-\phi_i) \bmod \pi_i < t+1 \tag{4.33}$$

- To check which recipients were able to receive segment #i and will not be able to receive it later in due time (i. e. must have the segment in their buffer), the following relation can be checked:

$$(s+t-\phi_i) \bmod \pi_i < t+1+\pi_i-\pi_i^+ \tag{4.34}$$

  For conciseness, let $r_i = t+1+\pi_i-\pi_i^+$.

- Let p be the prime number which has been selected for de-factorization and k be the number how often p has already been selected for de-factorization. To check if segment #i is independent of this prime factor (i. e. if it has to be examined further as it is not affected by the performed de-factorization), the following equation can be checked:

$$\pi_i \bmod p^k \neq 0 \tag{4.35}$$

- Let $p_1^{q_1}, \ldots, p_n^{q_n}$ be the prime factors of the period of segment #i, $k_1, \ldots, k_n$ be the expo-

nents of the currently selected de-factorization combination for the same prime factors (the remaining factors of the de-factorization combination are not relevant for the examined segment), $m_j = \min\{q_j, k_j\}$ and $d_1, \ldots, d_n$ be the values of the currently examined de-factorization combination, i. e. $\forall j \in \{1, \ldots, n\} : 0 \leq d_j < p_j^{k_j}$.

To determine the period π_i' and the phase ϕ_i' which have been selected by the de-factorization relative to the period of the segment (i. e. only regarding the prime factors of the period of the segment), π_i' can be calculated by

$$\pi_i' = \prod_{1 \leq j \leq n} p_i^{m_i} \tag{4.36}$$

and a value for ϕ_i' has to be found which satisfies

$$\forall j \in \{1, \ldots, n\} : \phi_i' \bmod p_j^{m_j} = d_j \bmod p_j^{m_j} \tag{4.37}$$

Finding ϕ_i' so that equation (4.37) is fulfilled is a well known problem which is mostly cited as Chinese Remainder Problem [HSR98b]. This problem can be solved by repeatedly applying the extended Euclidean algorithm [Knu97]: The extended Euclidean algorithm finds values u, v for given numbers f, g so that $\gcd(f, g) = f \cdot u + g \cdot v$. To apply it to this problem, let $f_0 = 1$, $x_0 = 0$, $f_j = f_{j-1} \cdot p_j^{m_j}$, $g_j = p_j^{m_j}$ and $x_j = f_{j-1} \cdot u_j \cdot q_j + g_j \cdot v_j \cdot x_{j-1}$ successively for all $j \in \{1, \ldots, n\}$ where u_j and v_j are the values found by the Euclidean algorithm for the numbers f_j and g_j. Then $\phi_i' = x_n$ fulfills the above condition.

- For a segment #i which has been received under the condition $(s + t - \phi_i) \bmod \pi_i < r_i$ (see above), one of the following three cases is valid:

  1. If
  $$r_i < \pi_i' \wedge (\phi_i' + t - \phi_i) \bmod \pi_i' \geq r_i \tag{4.38}$$
  the segment is not received under the selected de-factorization combination (i. e. the storage requirement must be left unchanged and the segment needs no further examination for this de-factorization).
  2. If this condition is not true but $\pi_i = \pi_i'$, the segment is received under the selected de-factorization (i. e. the storage requirement has to be increased and the segment needs no further examination for this de-factorization).
  3. Otherwise the segment has to be examined further for this de-factorization.

The complete algorithm for storage calculation is given in appendix A.6.

The storage requirement for a 720 segment transmission schedule of the Generalized Greedy

Broadcasting scheme using a playback delay of 6 slot intervals is shown in figure 4.42. The maximum storage requirements in this case are 324 out of 720 segments (45.0 %).

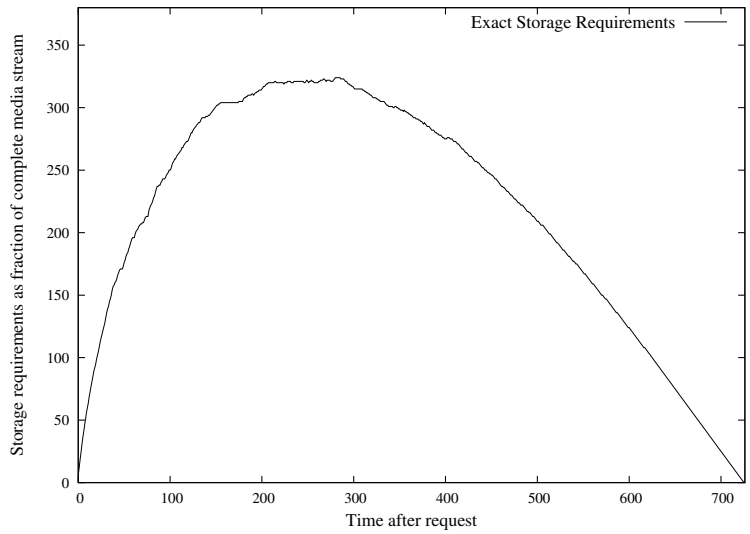


Figure 4.42: Storage requirements of a 720 segment transmission schedule with a 6 slot interval playback delay

4.19.2 Approximation of Storage Requirements for Frequency-Based Transmission Schemes

The above proposed algorithm still needs very much time if the schedule is large[15]. If only an approximation is needed, it is possible to use a much faster algorithm: This algorithm is based on the tree-based representation of the transmission schedule and assumes that the worst case storage requirements of sibling-subtrees can be added, i. e. it may overestimate the storage requirements sometimes. In no case a too low storage requirement is calculated by this algorithm, so if the storage of a receiver system suffices according to the value given by this algorithm, the media stream can be received without risking running out of storage.

Similar to the algorithm above, the storage requirement is calculated for each time of the playback separately. But instead of examining which receiver systems have to store which of the segments depending on the joining time, this algorithm examines the trees recursively to determine the worst case storage requirement.

The basic idea behind the algorithm is that if a node of a tree has been transmitted k times, each of the child nodes of the node have been sent either $\lceil \frac{k}{l} \rceil$ or $\lfloor \frac{k}{l} \rfloor$ times where l is the number of child nodes of the node. The former formula has to be applied $k \bmod l$ times and the latter one

[15] For example, the calculation of the storage requirements given in figure 4.42 took a total of roughly 1 200 hours on a farm of PC systems.

$l - (k \bmod l)$ times. The selection of the child nodes where the former or latter formula has to be used is unknown as this depends on the joining time of the receiver. But independent of the joining time, the nodes where either of the formula has to be applied to are consecutive (in a cyclic manner) as the child nodes of a node are transmitted consecutively.

For example, if a node has been transmitted eight times and the node has five child nodes, three of them have been transmitted twice and the other two of them have been transmitted only once. Five possible combinations have to be calculated in this case: Child nodes #1, #2, #3 have been transmitted twice (and the remaining child nodes once), child nodes #2, #3, #4 have been transmitted twice, child nodes #3, #4, #5, child nodes #4, #5, #1 or child nodes #5, #1, #2.

This leads to the following algorithm: If t is the time after the media request in slot intervals, N a node of one of the trees of the the transmission schedule and k the number of times for which N has been transmitted, then:

- If $k = 0$ or if N is an unused leaf node or if N is a used leaf node which transmits segment i but segment i has already been played, return the storage requirement 0 for node N.
- Otherwise: If N is a leaf node, return the storage requirement 1.
- Otherwise: Let l be the number of child nodes of N. Call this algorithm for each of the child nodes of N, using once $\left\lceil \frac{k}{l} \right\rceil$ and once $\left\lfloor \frac{k}{l} \right\rfloor$ as number of times how often each of the child nodes has been transmitted. Then add the results of $k \bmod l$ consecutive nodes (in a circular manner) which used the up-rounding formula and the results of the remaining nodes which used the down-rounding formula and return the highest of these sums. (If $k \bmod l = 0$, i. e. if $\left\lceil \frac{k}{l} \right\rceil = \left\lfloor \frac{k}{l} \right\rfloor$, this step can be simplified to call this algorithm for each of the child nodes of N using $\frac{k}{l}$ as number of times how often each of the child nodes has been transmitted and returning the sum of these values.)

If this algorithm is applied to each of the root nodes of the transmission schedule, using $t + 1$ as number of times how often the root nodes have been transmitted (including the current transmission), and the results from applying the algorithm to each root node are added, an upper estimation for the storage requirement is gained.

The approximate storage requirement for a 720 segment transmission schedule of the Generalized Greedy Broadcasting scheme using a playback delay of six slot intervals is given in figure 4.43 together with the exactly calculated storage requirements which have been gained in the previous subsection. In this case, the maximum worst case storage requirements of the fast algorithm are around 7.4 % higher than the corresponding value of the exact algorithm.

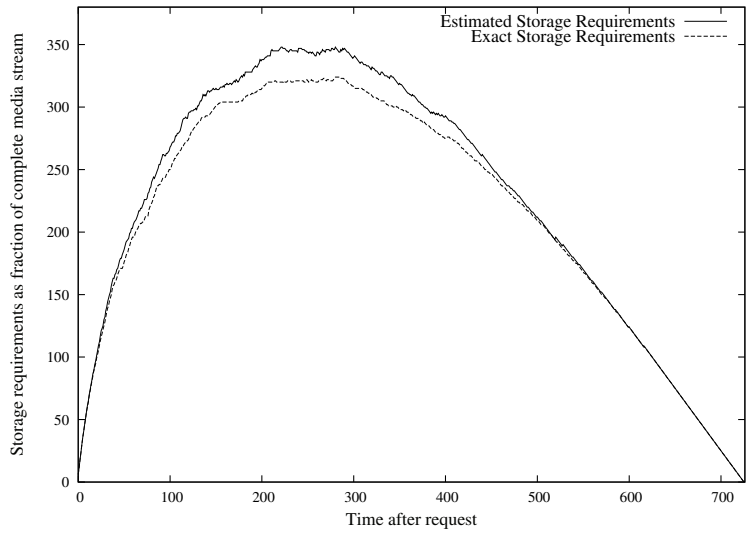


Figure 4.43: Exact and approximated storage requirement calculation of a 720 segment transmission schedule with a 6 slot interval playback delay

4.19.3 Lowering the Storage Requirements

To lower the memory requirements, the authors of [Pâr01a] proposed a simple solution: A limit is put on the required period for the segments. When such a limit is in effect, no segment must be stored longer than this limit specifies (in units of slot intervals) at the receiver system.

Unfortunately, limiting the required period increases the bandwidth requirements for a transmission: Segments which would normally be scheduled with a period higher than the period limit require the same bandwidth as segments of the period limit. This means that the bandwidth grows linearly for segments above the period limit instead of nearly logarithmically. The minimum required bandwidth can be calculated for this case using the following equation:

$$\frac{B^-}{b} = \sum_{i=1}^{n} \min\{\pi_i^+, \pi^+\} \tag{4.39}$$

where π^+ is the proposed period limit. For constant-bit-rate media streams, this equals

$$\frac{B^-}{b} = \Psi_0(\pi^+) - \Psi_0\left(\frac{P+W^+}{\delta}\right) + \frac{\frac{d+W^+}{\delta} - \pi^+ - 1}{\pi^+} \tag{4.40}$$

4.19.4 Application to Generalized Greedy Broadcasting Scheme

The algorithms described above for determining or estimating the storage requirements can be used for all frequency-based transmission schedules, including the Generalized Greedy Broad-

casting scheme. Likewise, a bound for the segment periods can be used to limit the storage requirements of receiver systems when the Generalized Greedy Broadcasting scheme is used.

Figure 4.44 shows the effect of a period limit on the minimum bandwidth requirement in an example. This figure illustrates that the bandwidth requirements grow rapidly if the period limit is below about 20 % of the media stream duration, i. e. if the receiver system can only store less than 20 % of the media stream. Thus, receiver systems should be equipped with at least as much memory as needed to store 20 % of the media stream.

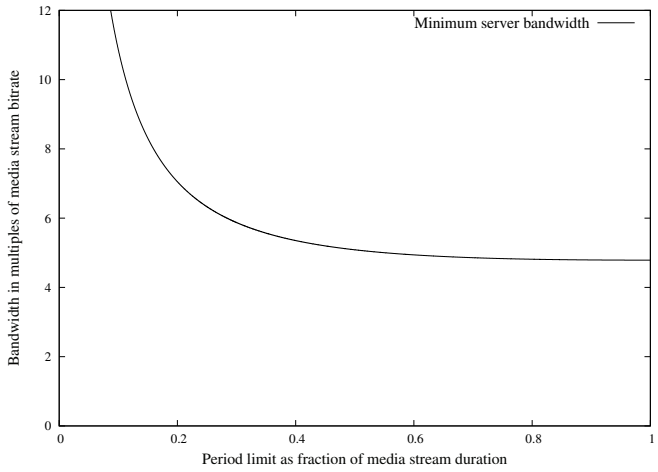


Figure 4.44: Minimum bandwidth requirements for a Generalized Greedy Broadcasting transmission with 7 200 segments and a playback delay of 60 slot intervals using a segment period limit

Figure 4.45 shows the resulting storage requirements for a period limit of 25 % of the media stream duration.

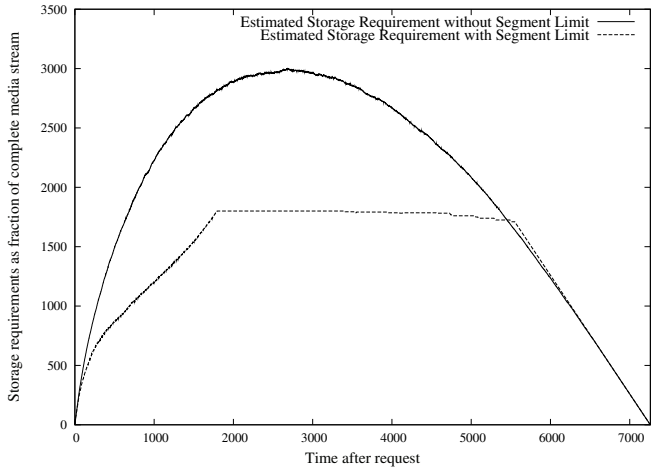


Figure 4.45: Storage requirements for a Generalized Greedy Broadcasting transmission with 7 200 segments and a playback delay of 60 slot intervals using a segment period limit of 1 800

4.20 Enhancing Interactivity

In section 2.4.2, three different types of interactivity have been proposed:

- The Media Scheduling classifies how the media stream has to be requested from the sender system (prearranged, pro-active, reactive, ...).
- The Playback Control specifies how far navigation within a continuous media stream is supported by the transmission scheme (rewind, fast forward, pause, jumping to scenes ...).
- Content Navigation describes which non-continuous navigation requests are supported by the transmission scheme (branches, switching between branches, virtual reality, ...).

This section discusses how a pro-active transmission scheme can be used in a reactive manner and how the interactivity levels of a transmission scheme can be increased (e. g. adding support for rewind and pause).

4.20.1 Using Explicit Media Requests for Pro-Active Transmission Schemes

In [PCL00b], the authors describe how a pro-active transmission scheme can be used in a reactive manner: When recipient systems send their playback requests to the sender system, the sender system can detect which segment transmissions are really needed by the receiver systems and which of the transmissions can be left out as they are not needed by any receiver. This way, every pro-active transmission scheme can be used in a reactive manner.

The advantage of using a pro-active scheme in a reactive environment is that the better of the two worlds is combined: As only segments are transmitted which are needed for at least one recipient, no bandwidth is wasted by transmitting segments unnecessarily. In particular, nothing is transmitted if no recipient is present. On the other side, as segments are transmitted according to a pro-active schedule, the maximum server bandwidth and service delay are determined by the schedule and are never exceeded.

4.20.2 Increasing Supported Level of Playback Control

It is obvious that every transmission scheme is able to navigate within the received part of the media stream without any interaction with the sender system: The receiver system saves the received segments in the same way as it would do when a playback is active and plays the segments repeatedly (if using rewind) or with a delay (when the user has rewound the media stream). The disadvantage of this enhancement is an increase of the storage requirements at the receiver system: If it is allowed to rewind the media stream from the end to a position in the very beginning, the whole media stream has to be present on the receiver system.

To lower the storage requirements, a limited version of this type of playback control enhancement can be implemented which limits rewinding to a specific amount of time (e. g. up to 5 minutes).

If only support for pause and/or slow motion is needed, a solution without additional storage requirements is possible, too: If the receiver system gets knowledge of the schedule (e. g. by adding the next transmission time to each segment), the receiver system can release received segments as soon as it determined that it will receive the segment again in due time. But unfortunately, this is not possible for all transmission schemes:

- When the media stream is terminated, no further transmissions will occur.
- For reactive schemes, no general statement can be made about the next transmission of segments.
- When a pro-active transmission scheme is used in a reactive manner, the sender system has to be informed about the delay which is introduced by using pause or slow motion so it can reschedule the affected segments.
- When the media stream is terminated, the sender system may not be willing to reschedule segments because the channel may be needed for other transmissions after the expected delay from termination.

For these reasons, adding this type of playback control is generally not advisable even if possible.

To support full playback control (including fast forward or jumping forward within the playback), the sender system has to be involved, too: For this, the authors of [Pâr01b] proposed two approaches:

- One additional contingency stream can be used for each receiver system where a fast forward is applied: On this channel, all segments can be transmitted which the receiver system will not receive in due time after it has performed the fast forward operation. These contingency streams can even be shared between several receiver systems to lower the additional bandwidth requirements at the sender system.
- If a period limit is applied in such a way that all segments $\#i$ are scheduled with a period $\pi_i \leq \pi^+$ and the receiver system saves all segments $\#i$ when receiving them (regardless of the fact that they may be received again in due time), the receiver system is able to perform fast forward actions as soon as the time of the period limit has been reached. This approach is additionally interesting as a similar type of period limit has been proposed in section 4.19 to limit the required receiver storage. In other words, the period limit can be either used to lower the receiver storage requirements (if the receiver system releases segments which are retransmitted in due time) or to allow fast forward actions in the latter part of the media stream (if the receiver system is able to store all these segments).

Combining these two approaches, additional contingency streams are only needed to support fast forward within the first part of the media stream while the second approach can be used to support fast forward after a given playback time, providing (regular playback) support for poorly equipped systems at the same time.

The authors of [THS02] proposed that an additional transmission is added where a condensed version of the media stream is transmitted to support fast forward and fast rewind interactivity. Although this allows to perform a fast forward with displaying the skipped parts, this approach does not solve the problem that the media stream must have been received when the playback is resumed. Nevertheless, this approach can be used to lower the amount of data which has to be transmitted on contingency streams while fast forwarding or rewinding, but it is questionable if this compensates for the additional amount of bandwidth for the condensed transmission.

The authors of [LKCC02] incorporated the condensed stream in the normal transmission. Therefore, the media stream is transmitted at twice the playback speed and displaying just I-frames (assuming MPEG encoding) for fast forward. Although this idea can generally be applied to any transmission scheme, the doubling of the bandwidth requirements degrade the efficiency of the transmission scheme by a huge amount.

Summarizing the above, playback control within the received media stream can be supported without any server interaction if enough storage (up to the whole media stream) is available at the receiver system, but for any playback interactivity level above this, additional server interaction and/or additional bandwidth is required. Thus depending on the receiver systems equipment, playback control within the received media stream part may be implemented but for large media-on-demand systems, a higher level of playback interactivity than this is not recommended.

4.20.3 Increasing Supported Level of Content Navigation

To support branch and multi-branch content navigation, the transmission scheme just has to support an increased playback delay: As the media stream can be split into several continuous streams (e. g. one main part and two endings for a simple media stream with two alternative endings), each of them can be scheduled as a separate transmission with an appropriate playback delay. The increase in bandwidth requirements for media streams with branches depends very much on the location and duration of the branches: If alternative beginnings are used, the two beginning media streams will require a lot of bandwidth, if just alternative endings are used, the bandwidth requirements are only slightly increased.

If multi-branch interactivity is not needed and branch interactivity is sufficient (i. e. switching between branches is not required), it is even possible to join just one of the beginning streams in case of alternative beginnings, so the number of alternative beginnings does not influence the receiver bandwidth requirements. In case of alternative endings, this approach cannot be applied

if the selection of the desired ending is not known at the beginning of the transmission. For multi-branch interactivity, the bandwidth requirements of both the receiver and sender system are increased because of the transmissions (and receptions) of all branches.

Full content navigation (virtual reality and similar) are only possible with sender system support. As it is expected that no two receiver systems navigate identically in this case, this type of interactivity should only be used with the Point-to-Point Transmission scheme.

4.20.4 Application to the Generalized Greedy Broadcasting Scheme

All the described approaches can be applied to the Generalized Greedy Broadcasting scheme, with the aforementioned limitations:

For using the Generalized Greedy Broadcasting scheme in a reactive environment, the only difference to the simple cases described above is that an increased playback delay, partial preloading and/or breaks or variable-bit-rate have to be taken into account when examining which segments are needed by the receivers. Additionally, the media startup can be delayed to the first request to benefit from the enhanced startup algorithm and the media termination can be enhanced slightly as not all segments have to be transmitted once when the stream is terminated (see sections 4.12 and 4.13) but only the segments which are needed by active recipients. Esp. if the last request for a media stream was long ago, this shortens the time of the media termination.

When increasing the supported level of playback control, playback control within the received part of the media stream can be supported in the same way as described above, assuming that the receiver system provides enough storage. If full playback control (including fast forward advancing the current playback time) is required, the proposed mechanisms from [Pâr01b] have to be applied.

To support content navigation up to multi-branch interactivity, the continuous parts of the media stream can even be scheduled into the main transmission scheme for the Generalized Greedy Broadcasting scheme by assigning appropriate needed segment periods[16].

4.21 Conclusion

The Generalized Greedy Broadcasting scheme which has been proposed in this chapter has many properties which delimit it from any other known transmission scheme: It supports nearly any enhancement of the transmission schemes which have been examined in chapter 3 (e. g. support for decoupled playback delay, (optional) partial preloading, variable-bit-rate media streams, live transmissions, dynamic changes of parameters, support for poorly equipped receiver systems)

[16] As the algorithm for generating Generalized Greedy Broadcasting transmission schemes assumes that segments are ordered by their needed segment period, the segments have to be sorted once before applying the algorithm.

while performing at the same time at an efficiency very near to the theoretical optimum for all particular cases. At the same time, it is based on the simple idea of frequency-based transmission schemes which allows detailed examinations (e. g. of receiver storage requirements or bandwidth consumption) and simple proves of correctness.

Technical classification	Functional classification
Communication type: One-to-many **Back-channel:** ! Optional (prevents unneeded transmissions if available) **Media-on-demand topology:** ! Star or tree **Sender bandwidth:** Multiple of media stream bit rate **Sender storage seeks:** Single position per slot **Receiver bandwidth:** ! Configurable multiple of media stream bit rate **Receiver storage size:** ! Configurable media stream buffering and optional partial preloading **Receiver storage seeks:** Single position per slot	**Media scheduling:** ! Pro-active or combined reactive/pro-active media-on-demand **Playback control:** ! Limited playback control within received part, playback control within received part or full playback control (with different impacts to receiver storage requirements and sender bandwidth requirements) **Content navigation:** ! Branch navigation, multi-branch navigation (with different impacts to receiver storage and bandwidth requirements) **Dynamic schedule changes:** ! Full dynamic scheduling change support **Dynamic content changes:** ! Full dynamic content support **Variable-bit-rate media support:** ! Full dynamic content support **Live streaming capability:** ! Live streaming capable

Table 4.7: Classification for Generalized Greedy Broadcasting transmissions

Altogether, the Generalized Greedy Broadcasting scheme not only makes media-on-demand more feasible, it also gives media-on-demand services the best possible cost-effectiveness by its high efficiency, its capability to scale for large environments and a high level of utilizability through its great flexibility.

The most notable limitations of the Generalized Greedy Broadcasting scheme are that it requires a multicast-capable network and receiver systems which are capable of receiving a media-on-demand transmission at a higher bit rate than a single media stream and that are equipped with storage for temporarily buffering parts of the media stream.

In the next chapter, media-on-demand transmissions are analyzed from a financial point of view. Therefore, the Point-to-Point transmission scheme which is used in many media-on-demand systems and the Generalized Greedy Broadcasting scheme which has been proposed in this chapter are compared.

Chapter 5

Media Transport Enhancements

IN the previous two chapters, many transmission schemes have been proposed, including the Generalized Greedy Broadcasting scheme which has emerged to be one of the most efficient transmission schemes. In this chapter, the focus is moved away from how a media stream can be transmitted most efficiently. Instead, mechanisms are examined which can by applied to many transmission schemes to enhance media-on-demand transmissions:

In section 5.1, problems around error correction are reviewed for media-on-demand transmissions and in section 5.2 the benefits of layered encodings are examined. In section 5.3, security related issues like confidentiality and authenticity are discussed. Last but not least, it is examined how advanced media-on-demand transmissions fit into currently used media transport protocols in sections 5.4 and 5.5, exemplifying this on the Real-time Transport Protocol RTP in combination with the Session Description Protocol SDP.

5.1 Error Detection, Correction and Recovery Schemes

During a media transmission, several transmission errors can occur: Data may get lost, duplicated, corrupted or packets may be re-ordered (assuming a packetized transport protocol). Most of these errors are very simple to detect: Duplications and out-of-order transmissions can be detected and repaired by adding sequence numbers to the packets, corrupted data can be detected with a very high probability if a checksum is added to the packet (e. g. CRC [PB61, Koo02], Adler32 [PD96] or others). But in case data has been lost or has been discarded because of data corruption, a treatment has to be determined.

The simplest reaction in case of lost data is to leave a hole in the media stream. As it is not possible to guarantee that a lost packet can be recovered by any mechanism in due time, a receiver system should always be able to deal with partial losses. Today's media encodings (e. g. MPEG [Itu00, Itu04, Itu05b]) can handle intermittent packet losses by means of re-

synchronization, i. e. the media codecs continue to work correctly when they reach the next synchronization point (e. g. the next frame or group of pictures) and other receiver side error concealment techniques (like interpolation for missing frames).

In the following, different ways are addressed to minimize the losses to be dealt with.

5.1.1 ARQ-Based Error Recovery

Another simple reaction to lost data is to request a retransmission, either by requesting the data directly (**negative acknowledge (NAK)-based error recovery**) or by omitting an acknowledgment for proper reception of the data (**acknowledge (ACK)-based error recovery**, together known as **automatic repeat request (ARQ) error recovery**) [CDJC84]. If the Point-to-Point transmission scheme is used, this is in most cases the solution of choice, although ARQ-based error recovery always introduces a delay of at least one round-trip time. To prevent halts of the media stream playback, the playback delay has to be increased.

But for other transmission schemes, ARQ-based error recovery has several additional major drawbacks:

- A back-channel from the receiver system to the sender system is required. Although this channel must only provide a low bandwidth, this means that this solution cannot be used in one-way environments.
- The receiver system needs a mechanism to detect which data has been lost. This can be accomplished by either giving the receiver system knowledge about the used transmission schedule or by adding sequence numbers to enumerate parts of transmitted data.
- The receiver system has to reserve additional bandwidth for retransmissions. Esp. if retransmissions are broadcast, this additional bandwidth may be very high as it is shared by all receiver systems. Distributed repeater systems may be used for retransmissions which serve only a small (possibly regional) group of receiver systems to reduce the amount of globally needed bandwidth for retransmissions.
- Similarly, the sender system needs additional bandwidth for sending retransmissions. Esp. if retransmissions are not broadcast, a transmission error which strikes many receivers (e. g. if the data has been lost near the sender) requires a large amount of retransmissions.
- Error recovery based on retransmissions does not scale for large numbers of recipients: If a packet loss occurs which strikes many receivers, each of them sends a retransmission request for the same part of the media stream to the sender system which becomes flooded with these requests (“NAK implosion”). This effect can be reduced if retransmission requests are broadcast after a random delay by receiver systems (assuming a broadcast/multicast capable back-channel) and if receiver systems suppress their request if they receive a retransmission

request for the same piece of data by another receiver system ("NAK suppression") or if retransmission requests from several receiver systems are handled regionally or aggregated regionally and forwarded in a hierarchical way ("NAK aggregation"), but this still increases the time for error recovery.

- The performance of retransmissions is determined by the receiver system with the worst reception quality as this receiver system sends the most requests for retransmissions. Esp. a failure at one receiver system can occupy most of the bandwidth which is available for retransmissions. As a solution, the amount of retransmissions a receiver system is allowed to request can be made dependent from the reception quality of the receiver system.

5.1.2 FEC-Based Error Correction

A different approach is to add redundancy to the media stream transmission in such a way that it can be used to recover lost data (**forward error correction**, FEC). The most simple form of this type of redundancy (besides duplicate transmissions) is to perform some type of arithmetic or logic operation (e. g. addition or exclusive-or) of the data of two or several data units and transmit the result — together with an identification which data units have been combined — to the receiver systems. If one of the data units has been lost, it can be reconstructed by performing the reverse operation (e. g. subtraction or exclusive-or in case of addition or exclusive-or, respectively) of the received data units and the redundancy.

Better results are obtained if a more advanced **erasure-resilient coding scheme** is used [Riz97]. The group of **maximum distance separable codes** provides the highest value of these coding schemes: For a number of n code words[1] these codes are able to create an arbitrary number of m redundancy code words in such a way that all n code words can be reconstructed if a subset of n of the $n+m$ code words are available.

The most commonly used representative of this group of maximum distance separable erasure-resilient coding schemes are **maximum distance separable Reed-Solomon codes**. Reed-Solomon codes are based on linear systems of equations over finite fields $GF[p^L]$ (also known as Galois fields [vdW91] with p^L elements where p is a prime integer (typically 2) and L is an arbitrary positive integer). If $\vec{d} = (d_{1,\ldots,n})$, $d_{1,\ldots,n} \in GF[2^L]$ are the data code words, the code words to transmit $\vec{e} = (e_{1,\ldots,n+m})$ can be calculated using a matrix multiplication by

$$\vec{e} = G \cdot \vec{d} \tag{5.1}$$

where G is the generator matrix of the code, containing $n+m$ rows and n columns.

[1] A code word constitutes the smallest amount of data the coding scheme operates on, typically one or two octets in case of binary encoded data with octet granularity.

If the upper n rows of G equal to the identity matrix, the erasure-resilient code is called **systematic**. Systematic codes are very beneficial because they keep the original data code words as first n generated code words. This has advantages for both the encoder and the decoder systems: For the encoder system, only m code words have to be calculated instead of $n+m$ code words, and for the decoder system only as many code words have to be reconstructed as have been lost, i. e. no computations are necessary if none of the first n words has been lost.

To recover $\vec{d}$ from a received set of code words $\vec{r} = (r_{1,\ldots,n}) = (e_{k_1,\ldots,k_n})$ with $1 \leq k_1 < k_2 < \cdots < k_n \leq n+m$, the data can be calculated by

$$\vec{d} = G'^{-1} \cdot \vec{r} \tag{5.2}$$

where G' is a matrix which only contains the rows $k_{1,\ldots,n}$ of G and G'^{-1} denotes the inverse of G'. To ensure that the inverse of G' can be calculated, the generator matrix must be selected in such a way that every square sub-matrix of it is regular. Often rows from Vandermond matrixes [Ber05a] are added to the identity matrix to obtain a valid generator matrix which guarantees this property.

One disadvantage of maximum distance separable Reed-Solomon codes is that they require a lot of operations for encoding and recovering: Typical algorithms for inverting a matrix require $O(n^3)$ operations. As an improvement, lines from Cauchy matrixes [Ber05b] can be used instead of lines from Vandermond matrixes as time of inversion can be reduced this way to $O(n^2)$ [BKKK+95][2].

An alternative to maximum distance separable Reed-Solomon codes are low-density parity-check codes (**LDPC-codes** [LMSS01, RSU01], also known as **Tornado codes**) [LMSS+97, LMS98a, LMS98b]. These codes are not maximum distance separable any more (and therefore do not guarantee that the data code words can be reconstructed from any n received code words). But as compensation, these codes are much faster: Instead of adding dense rows to the generator matrix (which are needed by maximum distance separable Reed-Solomon because of the regularity of sub-matrixes), low-density parity codes use sparse rows which contain zeros on most entries and only few ones. As a consequence, both encoding and recovery are much faster: For encoding, each redundancy code word can be generated by adding[3] a small number of code words. And even for decoding, a very simple linear system of equations has to be solved — a result of the simple generator matrix again[4].

[2] Theoretically, it is possible to decode Reed-Solomon based codes in time $O(n \cdot \log^2 n \cdot \log\log n)$ [LMSS+97] with a huge constant factor, which is slightly better than quadratic complexity for huge n, but the huge constant factor eliminates the positive effects in mostly any practical case. For small n and m it is also possible to precalculate all $\binom{n+m}{n}$ inverted matrixes.

[3] Addition is still performed in the field of $GF[2^L]$, i. e. to add two code words they are combined using a bitwise exclusive-or operation.

[4] The inversion of the matrix is often replaced by a graph-based propagation algorithm. This makes implementations of low-density parity codes very efficient even if n and m are large [BLMR98a, BLMR98b, BLM99, BLM98] although it may not recover as many erasures as possible.

Another approach is to use **unequal priority codes/priority encoding transmissions** [Xu03, ABEL94], i. e. codes which provide a different level of protection for subsets of data code words. These codes can be used if parts of the media encoding have different importance for a receiver. For example for an MPEG media stream, a higher recovery rate for I-frames than for P- or B-frames may provide an increased playback quality compared to an equally applied error correction.

5.1.3 Hybrid ARQ/FEC-Based Error Recovery

A hybrid approach combining ARQ-based and FEC-based error recovery is presented in [NBT98]: Similar to the ARQ-based scheme, the sender system reacts on retransmission requests, but instead of retransmitting the lost packets, the sender system broadcasts redundancy packets which combine lost packets of several receiver systems. The advantage of this approach is that the sender system has to perform much less retransmissions as each redundancy packet can treat a loss of any of the packets which have been used to calculate the redundancy packet at any receiver.

Similar to the FEC-based error recovery, the sender system can actually add some redundancy data to all transmissions to keep the number of retransmission requests low. It is even possible to adjust the amount of redundancy dynamically according to the number of reported losses.

Although hybrid error recovery performs much better than ARQ-based or FEC-based error correction alone, most of the aforementioned problems of ARQ- and FEC-based error recovery still apply to hybrid ARQ/FEC-based error correction.

5.1.4 Application to Segmented Media-on-Demand Transmission Scheme

While ARQ-based error recovery does not impose any problems other than the ones described above when applying it to a segmented media-on-demand transmission scheme like the Generalized Greedy Broadcasting scheme, integration of forward error correction leaves many possibilities for the place of application:

Segment-based FEC: Redundancy is calculated for each distinct segment.

Stream-based FEC: Redundancy for error correction is calculated for groups of consecutive segments of the media stream, independent of the arrangement of the segments during transmission.

Channel-based FEC: Redundancy is calculated for sequences of consecutively transmitted segments of each channel.

Transmission-based FEC: Redundancy is calculated for segments of all channels of a transmission together.

Each of these places for application has its own advantages and disadvantages:

- Playback delay and bandwidth efficiency:
  Data can only be recovered if enough segments of the forward error correction group have been successfully received. This means that the time for playback of a segment is determined by the reception time of the last segment of the FEC group.

  For stream-based and channel-based FEC, this means the playback delay for the transmission has to be increased by the duration of a FEC group at the receiver systems. For transmission-based FEC, all channels are combined into a single FEC group, therefore the playback delay needs only be increased by $\frac{1}{N}$ of the above described FEC group duration. For segment-based FEC, the segment is immediately available when it has been completely received, thus the playback delay must only be increased by one slot interval.

  For stream-based FEC, the opposite approach is also possible: All segments of a group can be scheduled at the time of the first segment of the group. In this case the playback delay can be kept unmodified and the efficiency of the transmission schedule is higher than for the approaches described above. Nevertheless, the efficiency of the transmission is lowered by the introduction of FEC in a similar way as larger segments decrease the efficiency (see section 4.5).

- FEC group size:
  While the former item implicitly suggests using small FEC groups, large FEC groups have a benefit, too: Depending on the data loss characteristics of the used network, a large FEC group may be required to handle bursts of data losses.

  Segment-based FEC is not able to recover data if the loss affects a too big part of it, transmission-based FEC has a similar problem if the other channels of the FEC group are affected, too. The FEC groups of stream-based and esp. of channel-based FEC span a long time in transmission and are most robust to burst data losses.

- Sender and receiver bandwidth:
  The redundancy has to be transmitted somehow to the receiver system, thus the required bandwidth increases for both the sender (to provide the FEC) and the receiver system (to consume the FEC). If separate channels are used for FEC, the most flexibility is provided: The receiver system can decide if it joins the additional FEC channels, depending on its capability to receive it, the quality of the received data or the users' preferences (e. g. the user may select if she attaches more importance to a high playback quality or to a short playback delay). It is even possible to split the redundancy data onto several channels: For example, if one channel carries one third of the redundancy data and the other channel two thirds, a receiver system can either join the first, the second or both of the redundancy channels, gaining different levels of FEC.

In case of stream-based FEC, the redundancy data will be needed in the same real-time fashion as the media data, so it can be scheduled with the Generalized Greedy Broadcasting scheme with its own schedule. For segment-based, channel-based and transmission-based FEC, the redundancy data has to be transmitted periodically, so it can be streamed continuously on separate channels.

- Impacts for receiver systems with limited receiver bandwidth (see section 4.18):
  To recover segments, the receiver systems must have received enough segments of each FEC group. In case of transmission-based FEC the segments of each FEC group are distributed across all channels. In case of receiver systems with limited receiver bandwidth, the receiver systems cannot receive all channels at once, so it is not possible to use this type of FEC grouping. An alternative is to create transmission-based FEC for subsets of channels, once for each occurring join combination of channels. But as this increases the amount of redundancy to send by a factor of $N - R + 1$, other types of forward error correction should be preferred if sender bandwidth matters.

- Storage size requirements:
  The storage requirements of the receiver system are increased by FEC, too, as the receiver system must be able to keep the segments of all currently active FEC groups in its storage. For transmission-based FEC, it must be possible to keep one FEC group in storage, and for segment-based and channel-based FEC, one FEC group per channel has to be kept. For stream-based FEC, recovery is not always possible concurrently to media reception: The transmission scheme only assure that segments are available when needed for playback, so all FEC data must be kept in storage until playback for stream-based FEC.

- Storage throughput requirements:
  Besides of the additionally needed amount of storage, the storage throughput is increased by FEC, too, because the receiver system has to access all segments of the FEC group when reconstructing a lost segment. For segment-based, channel-based and transmission-based FEC, recovery is performed concurrently to the reception: Even if the receiver system has enough system memory to keep all segments of active FEC groups in it, the storage throughput will change: When segments are lost, the current throughput will be below the average throughput, and when segments have been recovered, it will be above average. For stream-based FEC, the storage throughput is increased by the bandwidth used for FEC data because the FEC data has to be written to the storage and read shortly prior to playback if it is needed for recovery.

- FEC efficiency:
  In case of channel-based and transmission-based FEC, segments which are sent very often

(e. g. because of low segment periods) are included in many FEC groups. This has several disadvantages: Firstly, the receiver system has to receive these segments at each FEC group (even if they have already been played) or keep them in its storage so it is possible to recover segments of this FEC group. Secondly, the effective size of the FEC groups is lowered during playback because the more the playback advances the more segments are in the FEC groups which are not needed any more, lowering the recovery rate of the added redundancy.

- Media format dependency:
  Some FEC methods perform better if they have knowledge about the data to encode. For example, unequal priority codes can protect important parts of media streams with a higher recovery probability. Unfortunately, this causes varying bandwidths of the redundancy channel, depending on the media data which is included in the FEC group. In case of stream-based FEC, the transmission schedule for the FEC data can be generated according to the bandwidth variations which are caused by media-dependent FEC and therefore provide a constant FEC channel bandwidth. Segment-, channel- and transmission-based FEC cannot handle these bandwidth variations efficiently.

In summary, all types of FEC provide their own advantages and disadvantages: Segment-based FEC has the lowest effects on the transmission and system requirements, but has the smallest FEC group size of the four alternatives. Stream-based FEC can use larger FEC groups and maintains the highest FEC efficiency but has impacts on storage size and bandwidth requirements of receiver systems. Channel-based FEC is very robust to bursts data losses but requires the highest increase of the playback delay of the proposed FEC types. And lastly, transmission-based FEC only requires a minor increase of the playback delay but cannot be implemented efficiently for systems with limited receiver bandwidth. As each of the four alternatives is superior to the other ones in some property, a general recommendation is not possible; the decision for one of the FEC schemes has to be done depending on the system limits and preferences for the capabilities.

5.2 Layered Encoding

Another enhancement for media-on-demand systems are layered encodings [JL02]. For layered media encodings, the media stream is split into several sub-streams in such a way that progressive reconstruction at increasingly higher quality is possible: The first stream (the base layer) contains the most significant data and provides a base quality version of the media stream. If this stream is combined with the next higher one (the first enhancement layer), the quality is increased, until the last stream finishes the media stream to its finest quality.

Many different techniques for creating layered video streams exist [ML05]. The most common types are temporal layering, spatial layering and signal-to-noise-ratio layering: For temporal lay-

ering, individual frames of a video sequence are distributed over a set of layers. Thus, the more temporal layers are available to the receiver system, the higher is the frame rate of the video. In spatial layering, a multi-resolution representation is used to split each frame into a set of layers. When the number of layers is increased for the receiver system, the resolution of the video frames increases, too. In case of signal-to-noise-ratio layering, the amount of lossy compression applied through quantization is progressively adjusted.

5.2.1 Use of Layered Encodings

Layered encodings are very useful when heterogeneous receiver conditions should be supported: Instead of transmitting the media stream several times in parallel at different qualities (e. g. using different bit rates to support various receiver bandwidths), a layered encoding allows transmitting the media stream only once in several layers. This reduces the bandwidth requirements at the sender side as the base and medium quality parts of all transmissions are shared.

Receiver systems can select which of the layers of interest they are capable to receive depending on knowledge of the bandwidth from the sender to the receiver system or by measuring packet losses and estimating the receiver bandwidth therefrom [MJV96, SmNs04]. Another way is to rely on quality of service markers: If the base quality stream is tagged with a higher priority (e. g. utilizing quality-of-service tags of the lower protocol layers) than the high quality streams, the high quality data will be dropped when a bandwidth shortage occurs somewhere on the path from the sender to the receiver. This procedure relies on capabilities of the routers to distinguish the different quality-of-service groups and to drop the low priority data before dropping high priority one.

Another advantage of layered encodings has been mentioned in section 4.15: Using layered encodings, a change of the media bit rate for ongoing transmissions is possible by dynamically adding or removing another layer.

5.2.2 Application to Segmented Media-on-Demand Transmission Scheme

To use a layered encoding with segmented media-on-demand transmission schemes, each layer must be received independent of the other ones (or at least independent of the higher quality layers). This is best obtained if the layers are transmitted independent of each other, i. e. if they are segmented, scheduled and transmitted on distinct sets of channels.

This allows providing the highest flexibility:

- The receiver system can select which quality of the media stream it wants to (and is able to) receive.

- Each set of channels can be assigned a quality of service to assure that the base quality media layers are delivered more reliable than the high quality media layers.
- Bandwidth variations of the layers can be exploited better if transmission schedules are generated for each layer instead of the total stream.

The only disadvantage which is to be expected by layered encoding is that the total media bit rate and the protocol overhead are slightly higher than for a single, non-layered transmission.

5.3 Security

The purposes of security in a media-on-demand system are manifold but the main reason why security techniques are implemented in a media-on-demand system is access control: To prevent that unlicensed viewers receive the media streams, the data of the transmission has to be made confidential. But confidentiality can also be necessary if a limit audience is intended for other reasons, e. g. if a transmission contains company secrets and therefore should not be received by third parties, to prevent that a media stream can be received in regions where no license for distribution has been acquired for, or to exclude underage viewers from adult content.

To achieve confidentiality in a media-on-demand transmission, one of the following three mechanisms is generally used: The most straightforward way is to encrypt the media stream (see section 5.3.1). Another interesting way to keep a transmission confidential is chaffing and winnowing (see section 5.3.4): Instead of using encryption algorithms, chaffing and winnowing only uses digital signing (which can be implemented using a cryptographic one-way hashing function like SHA-2 [Fip02] or MD5 [Riv92]) and therefore can help to circumvent local law restrictions on the use of encryption. A third solution for confidentiality is steganography: Using steganography, the confidential information is embedded into a larger piece of information, making it difficult to detect the information. As this adds a lot of redundancy to the transmission, steganography is not analyzed in more detail for media-on-demand transmissions[5].

Other scopes of security involve authenticity and integrity: For example in case of news transmissions, it may be important that nobody can fake or alter the media stream unnoticed. If the authenticity and integrity should be perusable by every receiver, the best way to achieve this goal is digital signing based on asymmetric encryption. If it is adequate if the check is only performed by a trusted third party (or the producer itself), signing based on symmetric encryption (with a shared, secret key) or watermarking can be used, too.

Another use for watermarking is to mark a media stream invisibly in order to discover or prove the source, e. g. of pirate copies. Depending on the quality of the watermark, this only works as

[5] Steganography is sometimes entitled as insecure because many people expect that it is possible to extract the information if one knows where the information is hidden, but if implemented properly (e. g. using a pseudo random number generator which generates the correct locations depending on a key) it is as secure as Winnowing/Chaffing.

long as nobody detects the watermark or accidentally removed the watermark (e. g. by converting into another media format).

Non-repudiation, the fourth attainment of electronic security, is typically not needed for the media stream. Nevertheless it is automatically achieved if the content is signed by the provider if the provider is the only holder of the private key which has been used for the signature.

The following four subsections describe in more detail how the described mechanisms (encryption, signing, watermarking and winnowing/chaffing) can be used for media-on-demand transmissions. The last subsection describes a follow-up problem of encryption, signing and winnowing/chaffing: As all these mechanisms are based on keys, the task of transferring keys to the receiver systems has to be examined.

5.3.1 Encryption

Encryption provides the most simple (and most common known) way for confidential transmissions. Through **encryption**, a message (typically called **plaintext** or **cleartext** in the context of cryptography) is transformed into an encrypted message (the **ciphertext**) in such a way that its substance is hidden. Conversely, the cyphertext can be converted back into the plaintext by **decrypting** it.

While the mean of security of ancient encryption algorithms was the algorithm itself, modern algorithms are parameterized by a **key**: As the keys are part of the encryption and decryption process, it is not possible to reveal the plaintext from the cyphertext without knowledge of the decryption key, even if the decryption algorithm is known[6]. The introduction of keys to cryprographic algorithms allows relying on encryption algorithms which are quality-proven without loss of security.

Generally, two classes of encryption algorithms are available: Symmetric and asymmetric encryption algorithms. For symmetric encryption algorithms, the encryption and decryption keys are identitical, so they must be shared between the encrypting entity and the decrypting one. Asymmetric encryption algorithms use key pairs, of which one is used for encryption and the other for decryption. This allows publishing one of the keys as long as the other one is kept secret.

To provide confidentiality, either symmetric or asymmetric encryption algorithms can be used, but as symmetric encryption is much faster than asymmetric encryption, symmetric encryption is typically utilized.

The main problem of encryption is that the sender and receiver systems have to use the same keys (or keys of the same key-pair in case of asymmetric encryption) for the transmission. In case of Point-to-Point transmissions, these keys can differ for each receiver, making it very difficult to

[6]Two exceptions must be made to this general clause: If the key space is too small, it is possible to try each key (if it is possible to identify the plaintext if it is revealed). Additionally, some algorithms have “weak” keys which must be avoided because they cause the algorithms to degrade to limited security.

snoop a transmission, but if more receiver systems listen to one transmission the key has to be shared between all these receiver systems. Thus keeping the keys secret is an additional problem which gets the more difficult the more receiver systems share the same key.

Another problem may arise when new receivers enter a transmission or receivers are excluded from an ongoing transmission: To prevent that new receivers are able to decrypt transmitted data before they were allowed to join the transmission or that excluded receivers are able to decrypt transmitted data after they have been excluded, the encryption key must be changed and announced to all licensed receivers.

5.3.2 Signing

Digital signatures are typically based on asymmetric encryption: The signer encrypts the data (or a one-way hash of the data) using its private key. As this encryption can be reversed with the known public key of the signer[7], authenticity can be proven if decoding reveals the correct plain data (or the one-way hash of the data, respectively).

Alternatively, a symmetric encryption procedure or a one-way hashing algorithm can be used if the key is shared between the signer and the verifier of the signature. This means on the other hand that the verifier must be a trusted party as it is the only one which can verify the signature without publishing the key.

5.3.3 Watermarking

Watermarking is a technique where some data (e. g. information about the copyright or the licensee) is hidden invisibly within other data (e. g. the media stream). Typically, high frequency parts of images, movies or audio streams are used for hiding a watermark, but many other types of data within media streams can be used as well (e. g. the lowest bits of time stamps in a media stream, parts of the media stream which are not accessed for decoding or the presence of redundant additional information at specific locations). As these changes are not visible/audible, the viewer does not register that the media stream has been marked[8].

As the watermark is not needed for playing a media stream, this means that it is possible to remove the watermark without negative effects as soon as its location is known. Often it is even sufficient if the media stream is converted into another encoding to destroy the watermark. Thus watermarking is an unstable marking procedure if it is not integrated very well into the media content and if the marking process is not kept secret.

[7] It is also possible to attach the public key to the data together with a certificate from a trusted entity, i. e. a digital signature from a trusted entity that the attached public key is valid and who the owner is. This way only the public key of the trusted entity must be known.

[8] In this sense, a digital watermark differs from physical watermarks which are used as a mean of copy protection: Physical watermarks (e. g. on bank notes) are difficult to reproduce and simple to verify, while digital watermarks are simple to reproduce and difficult to detect.

5.3.4 Chaffing and Winnowing

Chaffing and winnowing [Riv98] is a technique which provides confidentiality without using encryption. The advantage of avoiding encryption is that strong encryption (i. e. encryption with a long key and a secure algorithm) is prohibited in some countries by law.

The basic idea behind winnowing and chaffing is that data must not be encrypted to protect it from eavesdropping. Instead, it can be split into small pieces and mixed with a lot of white noise. To allow the receiver to distinguish which of the pieces constitute the original data and which are garbage, the pieces are signed (e. g. using a one-way hash with a shared key): The receiver system can verify the signature if it has the key and sort out the garbage whereas the eavesdropper has no way to distinguish which parts contain the transmitted data and which parts are simply noise[9].

Instead of mixing the media stream with white noise, it is also possible to mix several transmissions as long as different keys are used for every transmission. This way, no bandwidth of the server is wasted by transmitting white noise while at the other side the level of security is maintained unchanged.

The disadvantage of winnowing and chaffing is that the receiver system must be able to receive data of several transmissions (or additional white noise) and identify the valid data of the correct transmission.

Address-Based Winnowing and Chaffing

This problem can be solved if a multicast address space is used instead of (or in addition to) signatures in the data: The sender system calculates a one-way hash over a sequence number and the secret key of the transmission, converts this hash into a multicast address of the given address range and uses this address for the transmission of the next piece of information. If (and only if) the receiver system has the secret key of the transmission, it can calculate these addresses, too, so it can join the multicast channel accurately timed to receive the transmitted data.

This model makes several additional assumptions about the receiver system:

- The sequence number of the receiver system must be synchronized with the number of the sender system. This can be solved if the server announces its currently used sequence number periodically on a global or transmission specific channel.
- If the multicast address space is too small to ensure that no two transmissions use the same multicast addresses at the same (or nearly the same) time, the receiver systems must be able to receive more than one data stream. As a resort, it is possible to use address adjustments for single transmissions which may be announced together with the current sequence number or to skip sequence numbers which cause conflicts.

[9] There is a low probability that the random signature of noise data accidentally matches the transmission signature. This chance can be reduced to nearly zero if the signature is long enough.

- The join delay must be determined by the receiver system to join the channel accurately in advance. Additionally, the overhead of joining and leaving multicast addresses must be low.
- Similarly, the leave delay must be short because an address must not be reused during this time by other transmissions.
- There must be enough concurrently active media stream transmissions on the selected multicast address space to prevent that all data is received and the streams assembled afterwards using heuristic algorithms (e. g. based on frame content, MPEG time stamps, RTP source identifiers, ...). This can be made difficult if the pieces are very small.
- The receiver system must be able to receive data on a high number of multicast addresses concurrently without switching into promiscuous mode when several transmissions are sent on the network link of the receiver system (e. g. for satellite-based transmissions).

Depending on the lower layer protocols, it may be possible that this address-based approach cannot be used. For example, multicast routing in the Internet is based on IGMP [CDKF+02, Fen97], which causes very high delays for joining and leaving multicast groups and produces a lot of additional routing protocol traffic. Additionally, the old version 1 of IGMP [Dee89] does not support active leave requests for multicast groups so the multicast addresses must not be reused before the IGMP group membership expires. In contrast, join and leave delays within local area networks or single link networks (e. g. satellite-based networks) can be very short and thus can be used without problems.

If the number of concurrent transmissions is really huge (i. e. if the total bandwidth exceeds the receive bit rate of an eavesdropper by a multiple), the data pieces can be made larger and the address-based approach can even be used in the aforementioned problematic cases: As long as an eavesdropper cannot receive all channels at once, she has to select an arbitrary subset of the channels to join, thus she only receives pieces of different media streams.

5.3.5 Key Management

While it is simple to exchange keys secretly for Point-to-Point transmissions (e. g. using the Diffie-Hellman-Merkle key exchange technique [Sch96, Hel02]), distributing and changing keys for a broadcast service is much more difficult: On one hand, all receiver systems have to receive the same key because the sender system can only use a single key for encryption when the media stream is broadcast to all receiver systems. But on the other hand, the transmitted key has to be protected from unlicensed receivers (e. g. customers who never joined the service or who canceled it).

Hierarchical key management schemes [Tan81] are broadly used to handle encryption for broadcast media-on-demand services: The International Telecommunication Union standardized

such a system in [Itu92] and the obsolescent EUROCRYPT system [CMS93, MQ95] utilize a similar approach. For these systems, subscription channels are grouped by the providers according to the subscription types and each group uses one **authorization key** for encryption. These keys are changed frequently and distributed periodically to all receivers, encrypted with a **distribution key** which is stored in a protected part of the receiver system (e. g. a smart card). Although this mechanism uses a temporary authorization key for encryption (which lowers the consequences after discovery of this key), they are based on the secrecy of the distribution key.

The authors of [Lee96] and [HSW00] propose a four-level hierarchy to be used for encryption instead of the two-level hierarchy of the above proposed one. For this scheme, the key of the next upper level is used for encryption of the key of the former lower one:

- A **control word** is used to encrypt the stream content. This key is replaced very frequently (typically every 10 to 120 seconds).
- An **authorization key** is used to encrypt the control word and is updated at short intervals (e. g. daily).
- A **distribution key** is used to encrypt the authorization key and is updated at long intervals (e. g. monthly).
- A **secret key** is saved at the receiver system and is installed during registration.

This scheme distinguishes itself from the previous one because it supports different secret keys for receiver systems. This allows excluding receiver systems from transmissions if their subscription is canceled, a procedure which had been performed by the receiver systems themselves for the previous approach (and thus invited for manipulation of this part of the receiver system). The successor of the EUROCRYPT system, the widely used DVB-CSA [CHIS05] standard, uses such a four-level hierarchy with receiver system specific keys, stored on smart cards.

Nevertheless, even these schemes require that keys are transmitted to all subscribed receiver systems. As this does not scale to a huge number of receiver systems, new multicast-based key distribution schemes have to be utilized. The standard approach to this problem, as described in [WHA99a, WHA99b, WGL00], is to assign several, partially different keys to the receiver systems. These keys can be used to address selected subsets of receiver systems and can be used to efficiently communicate key changes to this subgroup. The solution described in [WHA99b, WGL00] requires the transmission of $2 \cdot \log_2 n$ messages for a key change which has to be performed each time a user leaves or joins the group, and in [WHA99a], a more common solution which requires $k \cdot \log_k n - 1$ messages is proposed where n denotes the size of the group and k the degree of the key management tree which is internally maintained. The authors of [CGIM+99] proposed an improved solution which reduces the number of messages by a fac-

tor of 2, and the authors of [MP04] proved that this is the lower bound for multicast-based key distribution.

Summarizing the above, both approaches should be combined for media-on-demand: Hierarchical keys can be used so that keys can be updated frequently with low overhead and multicast-based key distribution schemes can be used to perform join and leave operations in an efficient and effective manner.

5.3.6 Application to Segmented Media-on-Demand Transmission Scheme

When applying encryption (or chaffing and winnowing) to a segmented media-on-demand transmission scheme, the place of encryption is significant:

If the media stream is encrypted before it is segmented and transmitted, the receiver system will receive segments which have been encrypted with different keys if more than one key is used for the encryption of the media stream. Additionally, several receiver systems will require the same encryption key at different times because the time of decryption depends on the playback time.

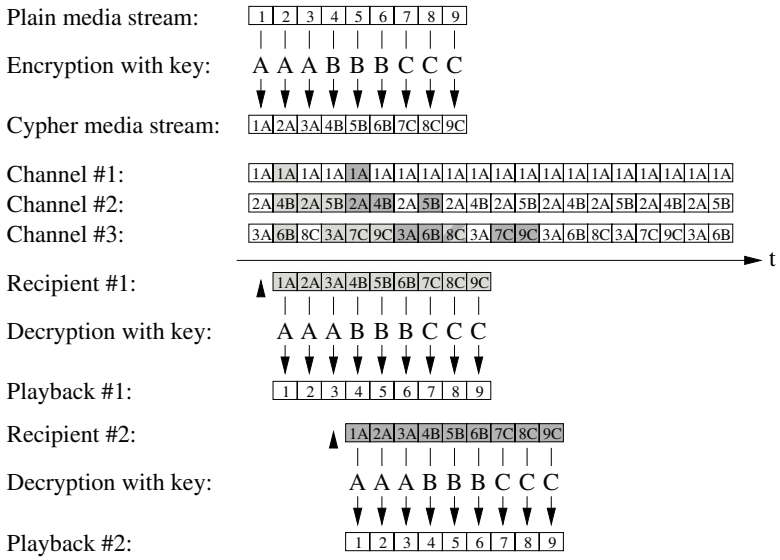
A better way to apply encryption to a segmented media-on-demand transmission scheme is to encrypt segments when they are sent by the transmission scheme. This allows changing the key periodically and distributing them concurrent to the changes to all licensed receiver systems at the same time. Figure 5.1 shows this difference in an example.

For signing and watermarking, it is assumed that the key will not change during transmission, so no additional problems arise when applying it to a segmented media-on-demand transmission scheme.

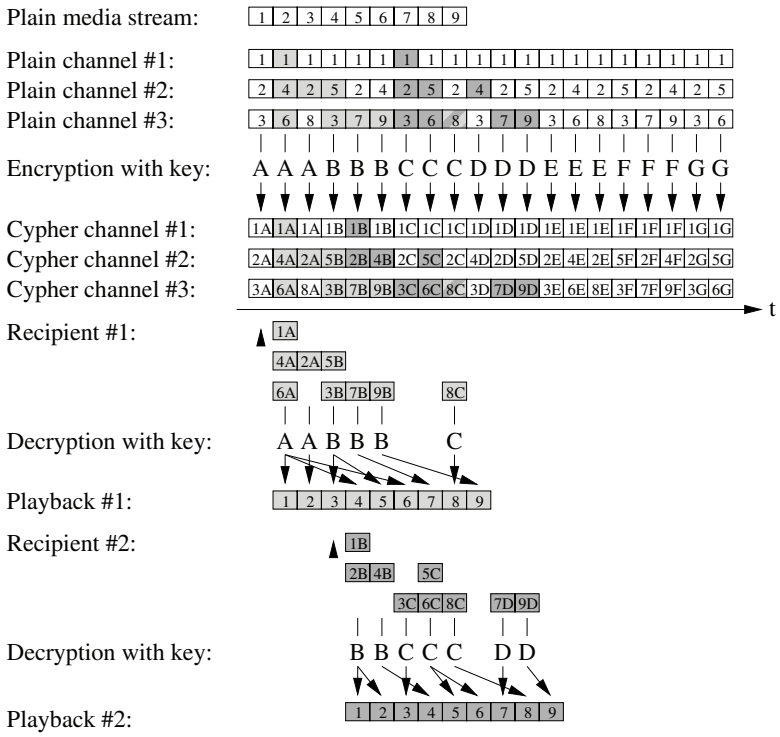
5.4 Embedding Media-on-Demand Transmissions in RTP

Real-time transmissions of media streams impose many requirements to the underlying lower protocol layers: At first, the data has to be transmitted in real-time (i. e. with low delay and jitter), with an acceptable reliability and preserving the sequence. But often, a well timed playback of received data is also required by many receiver systems as the display systems are not capable of storing much data. Additionally, multiplexing of several streams of one transmission (e. g. audio and video or different layers for layered encodings) may also be important. The impact of this demand is the capability to identify different streams of a transmission and to synchronize them for playback. In case of multiple sender systems, source identification may also be important, so that the media streams (e. g. audio streams of a conference) can be identified. Error correction mechanisms, confidentiality, authentication and other enhancing or security related issues are of interest, too.

A solution to these requirements is the **Real-time Transport Protocol (RTP)** [SCFJ03] which



(a) Encryption before segmenting: Recipients need same decryption keys at different time



(b) Encryption before transmission: Recipients need keys at same time

Figure 5.1: Application of encryption at different places to a segmented media-on-demand transmission

addresses many topics of real-time transmissions. RTP is designed as a layer on top of the packet-based Internet User Datagram Protocol (UDP) [Pos80] but it may also be used with other suitable

underlying transport protocols (e. g. on stream-based protocols like the Transmission Control Protocol (TCP) [Pos81]) as defined in [Laz06, SRL98]. Today, it is used for several applications, including multi-participant multimedia conferencing and voice over IP [Itu98, RSCJ+02] besides audio-visual media stream broadcasting.

As described in the standard [SCFJ03], RTP provides the following services:

- End-to-end delivery of data with real-time characteristics,
- payload type identification,
- sequence numbering,
- timestamping, and
- delivery monitoring.

To provide these services, RTP works closely together with the RTP Control Protocol (RTCP): While RTP is responsible for the transmission of data with real-time properties, RTCP is used to monitor the quality of service and to convey information about the participants in an ongoing session.

Many advanced topics are handled in separate RTP profile definitions which describe the interpretation of some RTP header fields for specific environments, and in RTP payload format definitions where the structure of RTP payload data is specified. For audio and video transmissions, the commonly used RTP profile `RTP/AVP` is specified in [SC03] and for different media types and encodings, several RTP payload format definitions have been standardized (e. g. for MPEG-1 and MPEG-2 [HFGC98], MPEG-4 [KNFM+00, vdMMSS+03], H.261 [Eve06], H.263 [OBSW+07], H.264 [WHSW+05], VC-1 [Kle06] and many audio encodings [SCFJ03, LHF05]).

Another purpose of RTP profile and payload format definitions is to add further enhancements to RTP, e. g. by encapsulating the payload of otherwise encoded RTP packets. For example, SRTP/SRTCP [BMNC+04] describes a way how protective measures can be added to an RTP session[10] and [(Ed07] propose mechanisms to add forward error correction to RTP. For better understanding of the remainder of this section, figures 5.2 and 5.3 show the structure of RTP and RTCP packets, respectively, which consist of the following fields:

V: Version

P: Padding indicator

X: Extension header indicator

CC: Contributing source (s. b.) count

[10]The original RTP standard [SCFJ03] also defines methods for encryption of RTP and RTCP data but for [BMNC+04] only the payload data is encrypted, making header compression [CJ99, BBDF+01, KCGT+03] and monitoring on the encrypted streams possible.

M: Marker bit, interpretation is profile dependent

PT: Payload Type

Sequence Number: Sequence number, incremented by one for each RTP packet of an RTP session

Timestamp: Sampling instant of the first octet in the RTP data, used clock frequency is profile dependent

Synchronization Source (SSRC) Identifier: Unambiguous identifier for one source in an RTP session

Contributing Source (CSRC) Identifiers: Identifiers of contributing sources for the payload

Header Extension Id: Identification or parameters of the header extension as defined by the profile, only present if X=1

Header Extension Len: Length of the header extension, only present if X=1

Header Extension: Profile specific extension to the RTP header, only present if X=1

RTP Payload Data: RTP payload data, payload type specific format

Padding: Padding octets, only present if P=1

Padding Len: Number of Padding octets (including the Padding Len octet), only present if P=1

PT specific: Value with Payload Type specific meaning

Length: Length of RTCP packet

RTCP Payload Data: RTCP payload data, payload type specific format

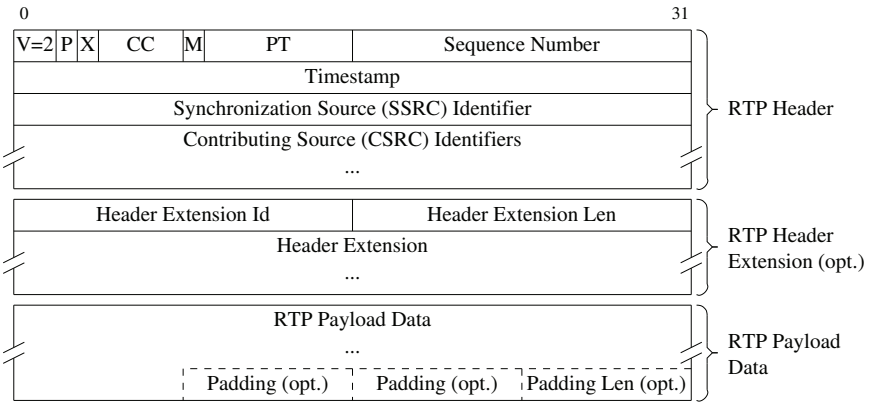


Figure 5.2: RTP packet format

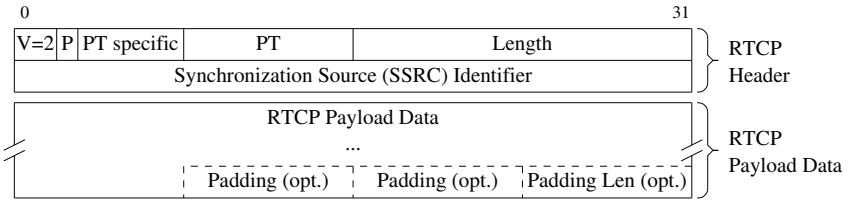


Figure 5.3: RTCP packet format

As media-on-demand transmissions involve a real-time transmission of media streams with similar requirements for enhancements, RTP seems to be a suitable transport protocol. Unfortunately, RTP has not been designed to handle media stream transmissions where packets are sent disordered as required by most of the proposed media-on-demand transmission schemes: RTP provides only a 16 bit sequence number in its header and this number is assumed to wrap around for long transmissions as explicitly stated in the protocol definition, therefore it does not provide a way for identifying the absolute position of a packet in a whole media stream. Packets which have sequence numbers differing significantly from expected ones are even rejected by RTP data header validity checks which conform to the RTP standard. Additionally, sequence numbers are used to detect packet losses during reception which is only possible if the sequence numbers of the transmission are strictly consecutive.

Another problem which results from the disordered transmission concerns the timestamps of RTP packets. RTP receivers and monitors use the timestamps for synchronization of several RTP sessions (together with information which is gained from RTCP sender reports which correlate the timestamps of all RTP transmissions to a common wall clock) and for jitter estimation. If RTP packets are intentionally transmitted disordered, the receivers and monitors would not be able to synchronize these media streams (as they do not have any corresponding wall clock information for timestamps of disordered packets) and they would not be able to determine the jitter.

As a solution, a new RTP payload format[11] has to be defined which encapsulates the payload of an arbitrary other RTP payload format for transmission.

5.4.1 Requirements for an RTP Payload Format for Segmented Media-on-Demand Transmissions

An RTP payload format for segmented media-on-demand transmissions has to fulfill the following requirements:

1. The payload format should comply fully to the RTP standard.

[11] An alternative to defining a new RTP payload format would be a new RTP profile. But unfortunately, RTP profiles are not “stackable” like RTP payload formats, so one new RTP profile per existing profile would have to be defined.

2. Packet-losses have to be determined just-in-time to give the sender immediate feedback if desired.
3. Duplications and disorders which are caused by the network should be resolvable.
4. Receiver systems and monitors should be able to estimate the jitter.
5. Receiver systems should be able to synchronize several segmented media-on-demand RTP streams.
6. Receiver systems should be able to rearrange RTP content to the correct media stream order.
7. Receiver systems should be able to detect duplicate packets which are caused by intentional repeated transmissions from a sender system (as requested by the transmission schedule).
8. RTP receiver systems should be able to determine the playback time for RTP packets without analysis of the media data.
9. A conversion between an RTP media stream (of any payload format, even encrypted) and a segmented media-on-demand transmission should be possible either in the sender and/or receiver system or by using RTP translators for either direction.
10. Other RTP profiles and payload formats, e. g. forward error correction and security, should be combinable with segmented media-on-demand transmissions as far as possible.

Essentially, to fulfill these requirements, the plain media representation is transformed at the sender into the segmented media-on-demand RTP payload format system and back into the plain representation at the receiver system.

But these requirements also have a wide influence on the design of the new segmented media-on-demand RTP payload format: The first three requirements demand that the sequence number of the RTP header is used as described in the RTP specification, i. e. it should be incremented by one for each transmitted packet (independent of the absolute position of the contained media data of the packet).

Determining a jitter is not as important for media-on-demand as for interactive transmissions (e. g. phone calls or multimedia conferences): For interactive transmissions, a delay above 100 ms is noticed slightly and delays beyond 300 ms disturb human communication [Iec05]. As the decoder has to delay the playback to compensate for the jitter (so the decoder does not run out of data), the jitter has to be kept low to keep the total delay short. For media-on-demand transmissions, it is feasible to overestimate the jitter because a delay of some hundred milliseconds increases the playback delay only insignificantly. Nevertheless, to allow an estimation of the jitter, the timestamp field in the RTP packets must not contain the capture timestamp of the media content as this timestamp would vary independent of the transmission time of the RTP packets. Instead, the timestamp field is filled by the sender with the designated transmission time of the

RTP packet. (The designated transmission time should be preferred to the real transmission time if possible to include operating system dependent jitter for sending packets.)

For rearranging all packets and for detection of duplicate transmissions, absolute sequence numbers are additionally required which identify the position of RTP payload uniquely in the media stream.

To transfer the playback time of an RTP packet, the original timestamp of the RTP media stream has to be preserved. Similarly, the marker bit and the padding bit have to survive the segmented media data transfer process.

To allow synchronization of several segmented media-on-demand RTP streams (e. g. video and audio), the receiver system has to get access to the RTCP sender reports which accompany the media stream transmission. This may also apply to some other RTCP messages (e. g. source description RTCP packets or goodbye RTCP packets).

SRTP protects RTP data and RTP headers by including both in the signing process when authentication is requested. Thus to support SRTP authentication, the complete header (including the original sequence number) must be preserved through the transfer process. As a consequence, in case of packet losses before transforming RTP media stream packets into segmented media-on-demand RTP packets, these losses are still visible after the unpacking process without the possibility to determine the location where the packets have been lost[12].

5.4.2 RTP Payload Format for Segmented Media-on-Demand Transmissions

The following RTP payload format proposal can be seen as a layer on the RTP processing stack between the RTP application and the transport layer: At the sender side, RTP packets from the application are converted into segmented media-on-demand RTP packets and (possibly at later time and more often than once) passed to the transport layer and at the receiver side, RTP packets are received from the transport layer and — at correct playback time — passed to the RTP-aware application. Figure 5.4 shows this setup using a segmented media-on-demand layer in the processing stack.

As an alternative to the insertion of a layer into the processing stack of the sender and receiver applications, it is also possible to use RTP translators for one or both of the conversions: As an RTP translator is an intermediate system that forwards RTP packets from one RTP session into another, probably performing conversions, an RTP translator can convert an RTP session into a segmented media-on-demand RTP session or vice versa. By this approach, existing RTP applications can be used without any changes for RTP-based segmented media-on-demand. Figure 5.5 shows this approach, using segmented media-on-demand translators for both the sending and receiving side.

As the segmented media-on-demand payload format encoding increases the length of the gen-

[12] This problem has already been noted in the RTP standard in the context of RTP translators.

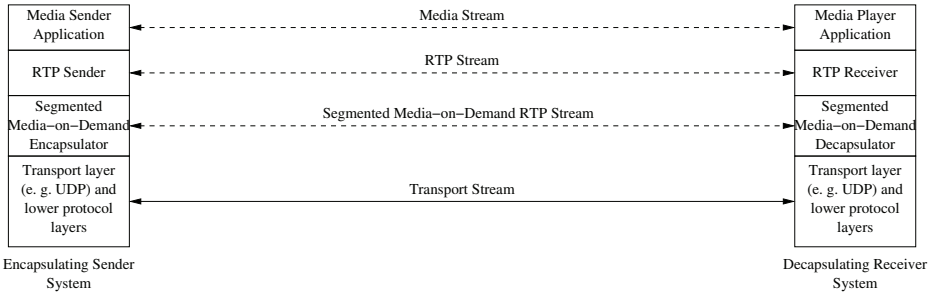


Figure 5.4: Segmented media-on-demand transmissions using inserting a layer in the RTP stack

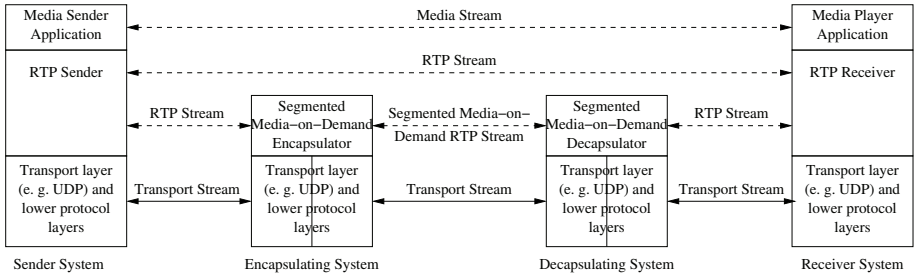


Figure 5.5: Segmented media-on-demand transmissions using RTP translators

erated RTP packets by 16 bytes, the encoding process in the RTP application has to be signaled a correspondingly lower maximum transmission unit size. Similarly, if other processing is applied to the generated packets afterwards which further increase the packet length (e. g. by adding security as defined in [BMNC+04]), the maximum transmission unit (MTU) size has to be set to a value which accommodates all post-processing steps. Additionally, the bandwidth requirements have to be adjusted to allow the transmission of additional segmented media-on-demand headers in RTP and RTCP packets.

Segmented Media-on-Demand RTP Packet Layout

The packet layout for segmented media-on-demand RTP packets is shown in figure 5.6. It consists of an RTP header, optionally followed by an RTP header extension (the header extension field is not allocated by the segmented media-on-demand RTP payload format as suggested in the guidelines for writers of payload format specifications [HP99] but is left for the RTP profile), a segmented media-on-demand header and the encapsulated RTP packet header and payload. As the synchronization source (SSRC) identifier, the contributing source (CSRC) identifiers and the RTP header extension of the encapsulated RTP packet are identical for the segmented media-on-demand RTP packet, these parts are left out for the encapsulated RTP packet header.

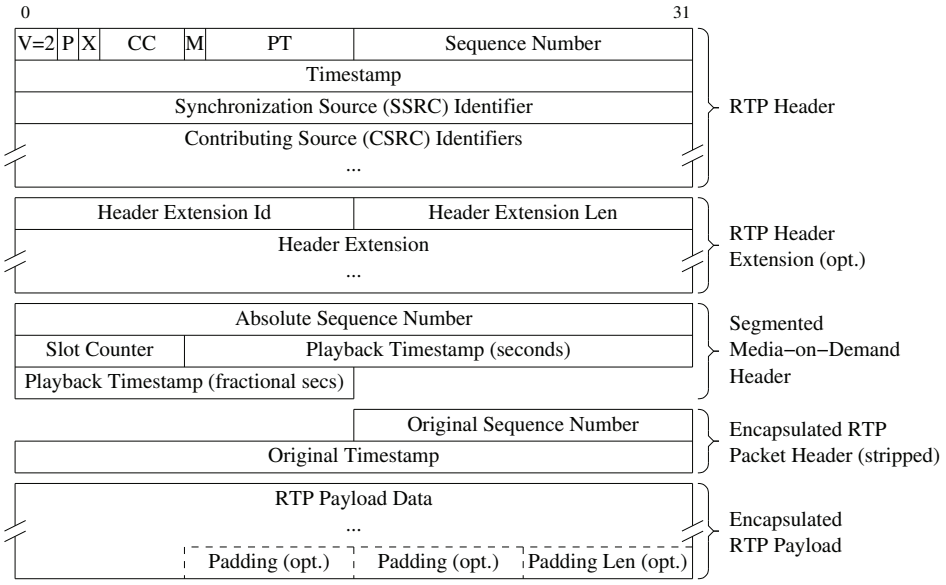


Figure 5.6: Segmented Media-on-Demand RTP packet format

RTP Header for Segmented Media-on-Demand Packets

The version field (V) is set to `2`. The padding bit (P), the extension bit (X), the marker bit (M), the synchronization source (SSRC) identifier, the contributing source identifier count (CC), the contributing source (CSRC) identifiers and the RTP header extension are identical for segmented media-on-demand RTP packets and the encapsulated RTP packets. The sequence number is set to a random value for the first packet and is incremented by one (wrapping around on overflow) for any further packet.

The assignment of an RTP payload type has to be performed in a dynamic way. The proposed segmented media-on-demand payload format does not preserve the payload type of the encapsulated media RTP packet (esp. in case of statically assigned payload types this is not recommendable), therefore a one-to-one mapping between the used payload type and the payload type of the media RTP packet has to be set-up and signaled outside of RTP (e. g. using SDP, see section 5.5).

The timestamp is set to the absolute transmission time of the packet plus an initial startup offset which is chosen randomly. Either the clock of the encapsulated media stream or a clock with sufficient precision (e. g. 90 kHz) if the clock of the encapsulated data is unknown should be used[13].

Segmented Media-on-Demand Header

The segmented media-on-demand header contains the following fields:

[13]The used clock frequency must be signaled outside of RTP, e. g. using SDP (see section 5.5).

Absolute Sequence Number: The absolute sequence number of the encapsulated RTP packet in the original media stream transmission. The absolute sequence number must be independent from the sequence number in the RTP header as the media-on-demand transmission scheme may require repeated transmissions of segments. The absolute sequence number should start with 0 at the beginning of the complete media stream and has to be incremented by one for any further RTP packet[14]. If packet losses are detected at the encapsulation side, the absolute packet number should reflect the losses if possible.

Slot Counter: This is a counter which is increased by one for each slot interval (and wraps around on overflow). It can be used by the receiver system to locate slot boundaries which may be necessary if the used transmission scheme requires that playbacks are delayed to slot boundaries. The slot counter should start with a random value.

Playback Timestamp: As the unpacking process may have no knowledge of the used media clock (and may have no access to sender reports to estimate it in case of encryption), this timestamp provides the instant when the encapsulated packet has to be passed to the receiver system. This timestamp uses a 65 536 Hz clock[15] independent of the payload format of the encapsulated RTP packet and starts with 0 at the beginning of the complete media stream. Due to its size, the Playback Timestamp should not wrap around for a media-on-demand transmission.

Segmented Media-on-Demand RTCP Packet Layout

The service provided by RTCP can be divided in two types for segmented media-on-demand transmissions: Some of the services should be provided end-to-end (e. g. sender reports which can be used to synchronize several RTP streams to a common wall clock), others only within each of the RTP sessions between sender, encapsulator, decapsulator and/or receiver (e. g. receiver reports).

While session-specific RTCP packets like receiver reports can be evaluated and handled separately for each of the RTP sessions (except if encrypted), end-to-end RTCP packets like sender reports (esp. if encrypted) must be carried to the receivers without any change. To transfer these RTCP packets from the sender to the receiver without interfering with the segmented RTP communication between the encapsulator and decapsulator, the RTCP packets have to be encapsulated in a similar way as the RTP packets[16]. Figure 5.7 shows the layout of segmented media-on-demand

[14] This means that the absolute sequence numbers of a segmented media-on-demand transmission which is preceded by a preloaded part does not start with 0 but continues the sequence numbers of the preloaded part. The same applies to the playback timestamp.

[15] A 65 536 Hz clock corresponds to an NTP [Mil92] clock as described in the RTP standard for wall clock timestamps in RTCP sender report packets (omitting the least significant 16 bit). If the sender process can access the wall clock timestamps of sender reports, this clock allows a simple interpolation.

[16] Although sender reports can partly be reconstructed by the decapsulator from segmented media-on-demand RTP

packets which are used for this purpose. As requested by the RTP standard, an RTCP packet of type “APP” is used for this purpose.

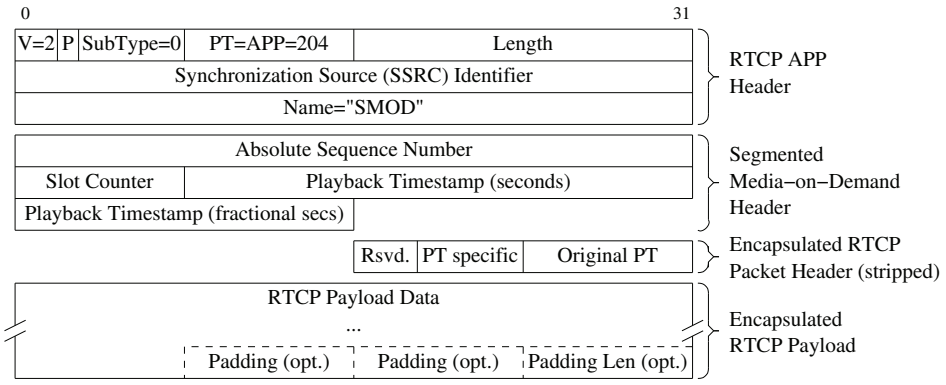


Figure 5.7: RTCP packet format for encapsulated transmission of RTCP messages

RTCP Header for Segmented Media-on-Demand Packets

The version field (V) is set to 2. The padding bit (P) and the synchronization source (SSRC) identifier are identical for segmented media-on-demand RTCP packets and encapsulated RTCP packets. The subtype (SubType) is set to 0 and the packet type (PT) to APP=204. The length field of the segmented media-on-demand RTCP packet is 4 higher than the length field of the encapsulated RTCP packet. (The length field measures in units of 32 bit.) The Name field is set to the ASCII representation of “SMOD” (as abbreviation of segmented media-on-demand).

Segmented Media-on-Demand Header

The segmented media-on-demand RTCP header contains the following fields:

Absolute Sequence Number: The absolute sequence number of the encapsulated RTCP packet in the original media stream control transmission. The absolute sequence number has to start with 0 and has to be incremented by one for any further segmented media-on-demand RTCP packet.

Slot Counter: This is a counter which is increased by one for each slot interval (and wraps around on overflow). It can be used by the receiver system to locate slot boundaries (even if no RTP traffic is sent) which may be necessary if the used transmission scheme requires that playbacks are delayed to slot boundaries. The slot counter should start with the same value as the slot counter in the RTP data stream and should be increased synchronously with it.

packets by evaluating the playback timestamp, this reconstruction is incomplete (e. g. packet losses at the encapsulator cannot be recorded in the sender reports) and impossible in case of encryption.

Playback Timestamp: As the decapsulating process may have no knowledge of the used media clock (and may have no access to sender reports to estimate it in case of encryption), this timestamp provides the instant when the encapsulated packet has to be passed to the playback system. This timestamp uses a 65 536 Hz clock independent of the payload format of the encapsulated RTCP packet and starts with 0 at the beginning of the whole media stream.

RTP/RTCP Packet Encapsulation Process

The sender shall perform the following steps to construct a segmented media-on-demand RTP packet from an RTP packet of one RTP stream:

- Copy the RTP header, RTP extension header and payload data into the appropriate positions of the segmented media-on-demand RTP packet. Thereby copy the sequence number and timestamp of the RTP header and the payload data into the corresponding fields of the encapsulated packet position.
- Replace the sequence number in the RTP header with the number one higher than the RTP header sequence number of the last packet (or a random value for the first packet). Replace the timestamp in the RTP header with the calculated instant when the RTP packet should be transmitted, in units of the clock of the media stream if known or a 90 kHz clock otherwise. At the beginning of the transmission, the timestamp should start with a random value.
- Calculate the playback time which corresponds to the timestamp of the RTP packet from sender reports (if available and not encrypted), from the media timestamp (if the media clock is known) or from the capturing instant and place it in the playback timestamp field of the segmented media-on-demand header of the RTP packet. Place the least significant bits of the current slot interval number in the slot counter field.

RTCP packets are processed in the same way.

RTP/RTCP Packet Decapsulation Process

The receiver has to keep the following context for the playback:

- The absolute sequence number of the next RTP packet to play,
- the absolute sequence number of the next RTCP packet to play,
- the current slot counter,
- a storage for RTP packets and
- a storage for RTCP packets.

The receiver shall perform the following steps when it receives a segmented media-on-demand RTP packet:

- When a higher slot counter value than stored in the context is received (considering wrap around on overflow), signal the start of a new slot boundary if the playback awaits this.
- If the absolute sequence number of the RTP packet is lower than the next absolute sequence number to play or if a packet with the same absolute sequence number already exists in the storage, ignore the packet.
- Reconstruct the encapsulated RTP packet from fields of the RTP packet and the encapsulated RTP packet.
- Save the packet together with the absolute sequence number and playback timestamp in the RTP packet storage.

Concurrently, it has to perform the following steps for playback:

- Check if the playback time for any unplayed packet in the storage has been reached[17]: Therefore, calculate the playback time for all packets in the storage with an absolute sequence number greater or equal to the next absolute sequence number of the context by adding the request time of the playback, the playback delay of the transmission (which has to be transmitted outside of the scope of RTP, e. g. through the session description protocol [HJP06], see section 5.5) and the playback timestamp. If the playback time for a packet has been reached, play all RTP packets in the storage up to this packet and update the absolute sequence number of the next packet to play in the context.
- If rewinding interactivity is not supported by the playback system, all played RTP packets can be removed from the storage.

RTCP packets are processed in the same way.

Encryption and Forward Error Correction

Authentication and encryption [BMNC+04] as well as forward error correction [(Ed07] can be applied to the segmented media-on-demand transmission to protect it. If forward error correction is applied to the segmented media-on-demand transmission using [(Ed07], segment-based, channel-based or transmission-based FEC can be achieved.

[17] The calculated playback time of RTP packet may be non-monotonic for some profiles if the synchronization timestamp is derived from the RTP media timestamp. For example in H.264, frames which are required to calculate interpolated frames are transmitted in advance although their RTP timestamp is higher than the timestamp of the interpolated frames. Therefore, it is not sufficient to examine only the playback time of the next packet to play. But in most cases it should suffice to examine the next few packets instead of all unplayed packets.

But it is also possible to tunnel an authenticated, encrypted and/or FEC-enhanced media stream transparently through the segmented media-on-demand transmission encapsulation. This has been made possible by a transparent transfer of media data independent of the payload type and by preserving all fields of the RTP/RTCP packets. If forward error correction is applied to the original media stream, the result corresponds to the stream-based FEC.

For encryption, it is more beneficial if it is applied after the transformation into a segmented media-on-demand transmission, so the encapsulator has access to RTCP sender reports. Another advantage if encryption is applied after the encapsulation (or a second time after encapsulation) is that the encryption key can be changed for long-term segmented media-on-demand transmissions.

Extended RTP Profiles for RTCP-based Feedback

In [OWSB+06] and [OC08], RTP profiles are defined which allow receiver systems to give timely feedback to sender systems. For segmented media-on-demand transmissions, this feedback cannot be provided end-to-end because the lifetime of communication between sender system, encapsulator, decapsulator and receiver system is independent from each other: For example, the encapsulator may perform a segmented media-on-demand transmission for a long time, although the sender system has finished the media stream transmission, and the decapsulator may continue streaming the media stream data to the receiver system even if the encapsulator has already finished the transmission.

For this reason, each of the translators should replace the "`RTP/AVPF`" or "`RTP/SAVPF`" profile with "`RTP/AVP`" or "`RTP/SAVP`", respectively, or vice versa, corresponding to the support of feedback and security for their session.

5.4.3 Summary

This section proposed a simple but effective way how segmented media-on-demand transmissions can be performed using the standardized real-time transport protocol RTP. Using a similar approach as SRTP, the plain media stream data is converted into a new format at the sender system and back into the original stream at the receiver system. Nearly all of the benefits and extensions which are available for RTP can be used for segmented media-on-demand transmissions, too, if the proposed encapsulation scheme is used.

Besides of this, the proposed RTP payload format allows performing the conversion between the original RTP media stream and the segmented media-on-demand RTP stream in separate systems. This gives the possibility for a smooth transition to segmented media-on-demand systems as changes in existing applications are not necessary.

5.5 Segmented Media-on-Demand Session Descriptions with SDP

In the previous section, the real-time transport protocol RTP has been extended by a new payload format so it is able to transport segmented media-on-demand content. But RTP provides no means to signal the used payload format, it is only capable of transporting the media content. To describe the payload format (and other session metadata), RTP is typically used in conjunction with the **Session Description Protocol (SDP)** [HJP06].

Each SDP session description may consist of one or several SDP media stream descriptions. For each SDP media stream, a transport protocol can be specified (e. g. the RTP audio/video profile [SC03] via `RTP/AVP`, the secured version of this profile [BMNC+04] via `SRTP/AVP`, and the versions with extended RTCP feedback [OWSB+06, OC08] via `RTP/AVPF` or `SRTP/AVPF`, respectively.). In case of one of the RTP audio/video profiles, each SDP media stream may consist of one or several RTP streams, identified by their payload type number. Figure 5.8 shows the dependencies between SDP sessions, SDP media streams and — in case of one of the RTP audio/video profiles — RTP streams.

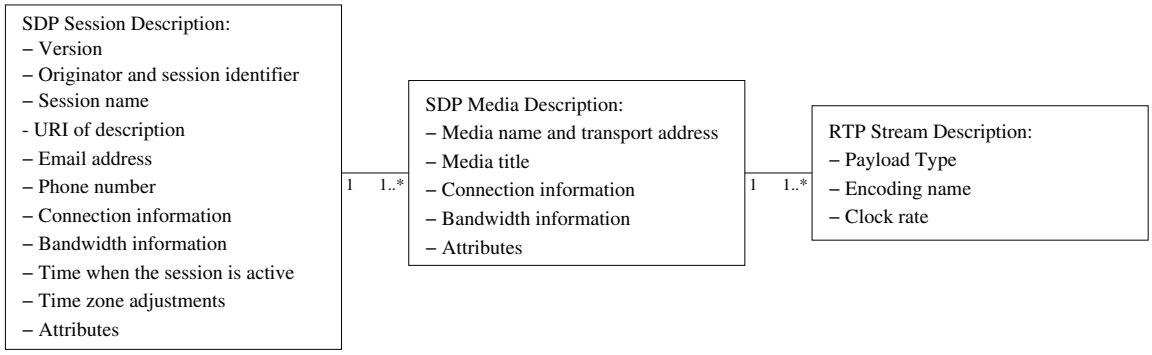


Figure 5.8: Relationship between SDP session, SDP media streams and RTP streams

SDP uses a textual notation to describe multimedia sessions, consisting of key-value pairs for the whole multimedia session (see table 5.1) and for each media stream in the session (see table 5.2).

While SDP only provides a standardized description for metadata of multimedia sessions, it is utilized by many other protocols to announce these session parameters (e. g. by the Session Announcement Protocol [HPW00]), to signal session parameters when initiating a session (e. g. in the Session Initiation Protocol [RSCJ+02]) or to indicate the session parameters in streams (e. g. in the Real Time Streaming Protocol [SRL98]). So defining an SDP media stream format for the proposed RTP payload format renders support for segmented media-on-demand transmissions in a wide range of media streaming software possible.

Key	Meaning of Value
v	Version
o	Originator and session identifier
s	Session name
i	Session information
u	URI of description
e	Email address
p	Phone number
c	Connection information for all media
b	Bandwidth information for all media
t	Time when the session is active
r	Times of repetitions of the session
z	Time zone adjustments
k	Encryption key for all media
a	Attributes for all media

Table 5.1: Session description parameters in SDP

Key	Meaning of Value
m	Media name and transport address
i	Media title
c	Connection information
b	Bandwidth information
k	Encryption key
a	Attributes

Table 5.2: Media description parameters in SDP

5.5.1 Requirements for an SDP Media Stream Format Description for Segmented Media-on-Demand

The SDP media stream format description for the segmented media-on-demand RTP payload format has to fulfill the following requirements:

1. The media stream format description should comply fully to the SDP standard.
2. The media stream format description should cover all media-on-demand specific transmission parameters (e. g. for playback delay, partial preloading).
3. The original payload format should be encapsulated in the new media stream format description.
4. Other RTP profiles and payload formats, e. g. forward error correction and security, should be combinable with segmented media-on-demand transmissions.

The ability to convert a media stream description is very important as only this allows translating an RTP media stream together with its description into the segmented media-on-demand RTP payload format and back into its native payload format.

5.5.2 SDP Media Stream Format Description for Segmented Media-on-Demand

To signal the payload format of media streams in SDP using the `RTP/AVP` profile or one of its variants, either a statically assigned payload type number can be used or the assignment can be done dynamically using an "`a=rtpmap:`" attribute. The "`a=rtpmap:`" attribute thereby is assembled in the following way:

`a=rtpmap:` ⟨*payload type*⟩ ⟨*encoding name*⟩/⟨*clock rate*⟩
[/⟨*encoding parameters*⟩]

⟨***payload type***⟩**:** The payload type number for which to specify a mapping.

⟨***encoding name***⟩**:** The name of the payload format used for the encoding.

⟨***clock rate***⟩**:** The clock rate of the encoding.

⟨***encoding parameters***⟩**:** Arbitrary encoding parameters.

Examples (extended excerpt from [HJP06]) for this are given in figure 5.9.

Session level information:
```
v=0
o=bn 3417930927 3417930927 IN IP4 83.246.72.241
s=SDP Example
i=Example showing SDP format
e=bn@example.com
t=3417930927 3417930927
a=recvonly
```
Statically assigned payload format:
```
m=audio 49232 RTP/AVP 0
c=IN IP4 224.2.3.4/127
```
Dynamically assigned payload format:
```
m=audio 49233 RTP/AVP 96
c=IN IP4 224.2.3.5/127
a=rtpmap:96 L16/16000/2
```
Several payload formats:
```
m=audio 49232 RTP/AVP 97 98 99
c=IN IP4 224.2.3.6/127
a=rtpmap:97 L8/8000
a=rtpmap:98 L16/8000
a=rtpmap:99 L16/11025/2
```

Figure 5.9: Example media specifications in SDP

To use the segmented media-on-demand payload format, only the dynamically assigned payload type specification can be used because the segmented media-on-demand payload format has not yet been statically assigned in any RTP profile. To translate a media description with a statically assigned payload format into one using the segmented media-on-demand payload format, an unused payload type number must be searched and substituted for the statically assigned payload type.

As a next step, the payload format has to be changed to "`SMOD`" (segmented media-on-demand) and the clock rate should be set to 90 kHz if it is unknown. Thereby the original payload format must be preserved so the decapsulator can signal the original payload format to the application later. The standard states that

> *encoding parameters added to an "`a=rtpmap:`" attribute SHOULD only be those required for a session directory to make the choice of appropriate media to participate in a session*

and that other parameters should be added in other attributes. As the original payload format is important for the choice of the appropriate media, it must be added as encoding parameter to the "`a=rtpmap:`" line. For this, the following format will be used for the ⟨*encoding parameters*⟩ of the "`a=rtpmap:`" line:

- If a statically assigned payload type had been used (which has been converted into a dynamic one as described above):

  `static/`⟨*original payload type*⟩

- If a dynamically assigned payload type had been used:

  `dynamic/`⟨*original encoding name*⟩`/`⟨*original clock rate*⟩[`/`⟨*original encoding parameters*⟩]

Some other fields of the session description must be updated when translating a session description into one using segmented media-on-demand transmissions at the encapsulator or back to the original format at the decapsulator:

- The "`c=`" connection field should be updated by the encapsulator and decapsulator: Firstly, for the segmented media-on-demand transmission, multicast addresses should be used, so several decapsulators can receive the transmission. But even if multicast addresses have already been used by the sender, these addresses should be exchanged by other multicast addresses to prevent that the original media streams are routed to the decapsulators[18]. The

[18] If the sender, encapsulator and decapsulator all use different IP addresses, source-specific multicast routing could also be used if utilized by all participants.

decapsulator should replace the multicast addresses either by one other set of addresses per receiver or use its own unicast address. Additionally, the port numbers in the "m=" media formats must be updated so they are (at least for equal connection addresses) unique.

- The "t=" timing and "r=" repeat times fields should be adjusted to reflect the segmented media-on-demand transmission: The timing field should be updated to indicate the start and end of the segmented media transmission and the repeat times field should signal any scheduled repetitions of the segmented media-on-demand transmission. Any existing "z=" zone offset must be replaced according to time zone adjustments which occur during the segmented media-on-demand transmission (which may take place at a different time than the original transmission). The decapsulator will derive the "t=" timing for the playback using the media playback time as start time and calculating the end time from the duration in the "r=" repeat time field.
- The "b=" bit rate field should be updated to reflect the overhead of the segmented media-on-demand RTP encapsulation. To allow reconstruction of the original values after decapsulation, the original values should be transmitted in "a=smod-b:" attributes.
- The "o=" origin field contains a globally unique identification of the session and a version number for the session. As each translator forwards data from one RTP session into another one, the sessions must be named differently.
- An "a=recvonly" attribute should be added if not already present. "a=sendrecv" attributes should be deleted. (If an "a=sendonly" attribute is present, this should be considered as an error.)

Playback Delays

In addition to the above described modifications to standard parameters, some media-on-demand specific attributes must be added to the media description.

To indicate the playback delay to the decapsulator, the minimum and maximum playback delay are encoded using an "a=smod-delay:" media stream attribute. This attribute is defined as follows:

a=smod-delay: ⟨*minimum delay*⟩ [⟨*maximum delay*⟩]

⟨***minimum delay***⟩**:** The minimum playback delay in clocks of the segmented media-on-demand media stream.

⟨***maximum delay***⟩**:** The maximum playback delay in clocks of the segmented media-on-demand media stream. The maximum playback delay actually works as a flag, indicating if the playback should start with ⟨*minimum delay*⟩ after a slot boundary has been reached. The

signaled value can be used to indicate the maximum playback delay to the user or to implement a timeout. If minimum and maximum playback delay equal in the used transmission scheme (see section 4.9), the maximum value must be omitted or set to the same value as ⟨*minimum delay*⟩.

Note that the `Playback Timestamps` in the segmented media-on-demand RTP packets are always starting with 0 at the beginning of a media stream, any delays which are specific to the scheduling parameters are signaled in SDP. This has the advantage that the scheduling parameters can be changed dynamically (see section 4.15 and 4.16) without updates in the segmented media stream, a prerequisite for not interrupting ongoing transmissions.

Partial Preloading

Partial preloading consists of two processes: The preloading of a part of the media stream and the playback of a preloaded media stream as part of a segmented media-on-demand transmission.

Although preloading does not require real-time transmission and can be performed e. g. using content distribution systems, this section describes a way to transfer the preloaded part of a media stream using RTP for convenience[19].

For the preloading process, a means to signal the preloading (and thus storing the media stream) is required. This is indicated by an "`a=smod-preload:`" attribute in the media description:

`a=smod-preload:` ⟨*payload type*⟩ ⟨*expiry time*⟩ ⟨*username*⟩ ⟨*preload-id*⟩ ⟨*preload-version*⟩ ⟨*nettype*⟩ ⟨*addrtype*⟩ ⟨*unicast-address*⟩

⟨***payload type***⟩**:** The payload type number of the media stream which is to be preloaded. The content is named separately for each payload type during preloading so it is possible to associate multiple streams with their continuation at playback.

⟨***expiry time***⟩**:** The time when the preloaded data can be dropped because it is not needed any more. This value is the decimal representation of a Network Time Protocol (NTP) [Mil92] time value in seconds since 1900.

⟨***username***⟩**:** The user's login on the originating host, or "-" if the originating host does not support the concept of user IDs.

⟨***preload-id***⟩**:** A numeric string such that the tuple of ⟨*username*⟩, ⟨*preload-id*⟩, ⟨*nettype*⟩, ⟨*addrtype*⟩, and ⟨*unicast-address*⟩ forms a globally unique identifier for the preloaded data. The method of ⟨*preload-id*⟩ allocation is up to the creating tool, but Network Time Protocol (NTP) [Mil92] format timestamps are suggested to ensure uniqueness.

[19] If preloaded parts are transmitted using other means, the proposed naming scheme must be followed, so receiver systems are able to insert the media stream on playback.

⟨***preload-version***⟩**:** A version number for the preloaded data. Its usage is up to the creating tool, so long as ⟨*preload-version*⟩ is increased when a modification is made to the preloaded data. Again, it is recommended that an NTP format timestamp is used.

⟨***nettype***⟩**:** A text string giving the type of network, e. g. "IN" for "Internet".

⟨***addrtype***⟩**:** A text string giving the type of address that follows, e. g. "IP4" or "IP6".

⟨***unicast-address***⟩**:** The address of the machine on which the preloaded data was created.

The above attributes assure that the preloaded media stream data is uniquely named (using the same approach as SDP uses for the "`o=`" origin field), has a version number (so the decapsulator can replace old versions with new ones) and an expiry date (to enable the decapsulator to delete the preloaded data after some time). When the decapsulator receives a media stream with the "`a=smod-preload:`" present, it should save the media stream together with the media description for later playback.

To request playback of a preloaded media stream at the beginning of a segmented media-on-demand transmission[20], an "`a=smod-insert:`" attribute is used:

`a=smod-insert:` ⟨*payload type*⟩ ⟨*username*⟩ ⟨*preload-id*⟩ ⟨*preload-version*⟩ ⟨*nettype*⟩ ⟨*addrtype*⟩ ⟨*unicast-address*⟩

⟨***payload type***⟩**:** The payload type number of the media stream where to insert the preloaded part of the media stream.

⟨***username***⟩**,** ⟨***preload-id***⟩**, . . . :** Identification of the preloaded part.

Example

Figure 5.10 shows the result of this procedure for the above given examples, using a (minimum and maximum) playback delay of 10 seconds, and figure 5.11 shows an example using a 10 second partial preloading.

5.5.3 Summary

SDP is a commonly used specification to describe session parameters, esp. in combination with RTP. Specifying a media format description for the previously proposed RTP payload format completes the payload format specification and supports the use of the segmented media-on-demand RTP payload format in a wide range of media streaming applications.

[20] Only preloading to the beginning of a media stream is supported with this attribute; introducing breaks into a media stream which are filled with preloaded data is out of the scope of SDP.

Session level information:
```
v=0
o=bn 3417930927 3417930927
  IN IP4 83.246.72.241
s=SDP Example
i=Example showing SDP translation
e=bn@example.com
t=3417930927 3417938127
a=recvonly
```

Statically assigned payload format:
```
m=audio 49232 RTP/AVP 0
c=IN IP4 224.2.3.4/127
```

Dynamically assigned payload format:
```
m=audio 49232 RTP/AVP 96
c=IN IP4 224.2.3.5/127
a=rtpmap:96 L16/16000/2
```

Several payload formats:
```
m=audio 49232 RTP/AVP 97 98 99
c=IN IP4 224.2.3.6/127
a=rtpmap:97 L8/8000
a=rtpmap:98 L16/8000
a=rtpmap:99 L16/11025/2
```

(a) Example media specifications in SDP

Session level information:
```
v=0
o=bn 3417930937 3417930937
  IN IP4 83.246.72.242
s=SDP Example
i=Example showing SDP translation
e=bn@example.com
t=3417930927 3420522927
r=10 7200 0
a=recvonly
```

Statically assigned payload format:
```
m=audio 49232 RTP/AVP 100
c=IN IP4 224.2.4.4/127
a=rtpmap:100 SMOD/90000/static/0
a=smod-delay:100 900000
```

Dynamically assigned payload format:
```
m=audio 49232 RTP/AVP 95
c=IN IP4 224.2.4.5/127
a=rtpmap:96 SMOD/16000/dynamic/L16/16000/2
a=smod-delay:96 160000
```

Several payload formats:
```
m=audio 49232 RTP/AVP 97 98 99
c=IN IP4 224.2.4.6/127
a=rtpmap:97 SMOD/8000/dynamic/L8/8000
a=rtpmap:98 SMOD/8000/dynamic/L16/8000
a=rtpmap:99 SMOD/11025/dynamic/L16/11025/2
a=smod-delay:97 80000
a=smod-delay:98 80000
a=smod-delay:99 110250
```

(b) Translated segmented media-on-demand media specifications in SDP

Figure 5.10: Media specifications in SDP

5.6 Conclusion

In this chapter, several enhancements have been proposed which can be applied to media-on-demand transmissions independent of the used transmission scheme:

To handle partial loss of data of a transmission, several treatments have been described, one reactive (ARQ-based error recovery), one pro-active (FEC) and the combination of these two approaches (HARQ-based error recovery). It has been analyzed that it is more efficient to apply forward error correction to the media stream directly instead of applying it to the sequence of segments which are sent on one or all channels.

To support systems with low receiver bandwidth, modifications to the Generalized Greedy Broadcasting scheme have been proposed in the previous chapter. Another approach is to use layered encodings where the media stream is split into a common, base-quality part and one or

Preloading:

```
v=0
o=bn 3417930927 3417930927 IN IP4 83.246.72.242
s=SDP Example
i=Example showing SDP format
e=bn@example.com
t=3417930927 3417930987
a=recvonly

m=video 49152 RTP/AVP 96
c=IN IP4 224.2.4.4/127
a=rtpmap:96 SMOD/90000/dynamic/H264/90000
a=smod-preload:96 3420522927 bn 3417930927 3417930927 IN IP4 83.246.72.241
```

Playback inserting preloaded data into a transmissions:

```
v=0
o=bn 3417934537 3417934537 IN IP4 83.246.72.242
s=SDP Example
i=Example showing SDP format
e=bn@example.com
t=3417934587 3420522987
r=1 7200 0
a=recvonly

m=video 49153 RTP/AVP 97
c=IN IP4 224.2.4.5/127
a=rtpmap:97 SMOD/90000/dynamic/H264/90000
a=smod-delay:97 0
a=smod-insert:97 bn 3417930927 3417930927 IN IP4 83.246.72.241
```

Figure 5.11: Segmented media-on-demand media specifications using partial preloading

several enhancing parts. Depending on the available bandwidth, the receiver system can decide how much of these streams are received.

To protect the media stream transmission, several security related issues have been addressed: Encryption for confidentiality, signing for authenticity and integrity, and watermarking for recognition of unlicensed copies. As an alternative approach to encryption, chaffing and winnowing has been proposed and extended to lower the bandwidth requirements of receiver systems.

To assist embedding of segmented media-on-demand transmissions in current environments, a payload format for the real-time transport protocol RTP and a media format specification for the proposed payload format in SDP have been defined: Using the segmented media-on-demand payload format, it is possible to transmit segmented media-on-demand transmissions of all size-based and frequency-based transmission schemes while benefiting from the capabilities of the real-time transport protocol at the same time.

Additionally, the proposed RTP payload format can be implemented in separate systems. This allows extending existing RTP-based media-on-demand system to support segmented media-on-demand transmissions using any size- or frequency-based segmented transmission scheme (e. g.

the Generalized Greedy Broadcasting scheme proposed in chapter 4) including the other above described enhancements without changing the existing system.

Chapter 6

Analysis of Media-on-Demand Application Cases

THIS chapter presents an analysis of the cost-effectiveness of different media-on-demand approaches in various application cases. Instead of differentiating the proposed media transmission schemes by their requirements or efficiency as done in chapter 3, their costs and benefits are balanced against each other in this chapter.

After starting with some basic calculations about cost analysis for media-on-demand in the next section, three application cases of different size are examined in sections 6.2 to 6.4, resp. This examination in three groups of different size is commendable due to the effects on the number of receiver systems and bandwidth availability:

- In small environments, media-on-demand systems have to serve only a low number of receiver systems while typically a high bandwidth is available. Typical small environments for media-on-demand systems can be found in hotels, air planes, trains, ships or blocks of houses.
- Larger installations, e. g. hotel chains, large companies or small villages, are examined in the group of medium-sized environments. They differ from the small environments by a higher number of receiver systems and less available (or at least more expensive) bandwidth.
- Distribution of media streams to a huge number of receiver systems, e. g. to cities or even whole countries, with low bandwidth availability, are examined in the large environment case.

The last section of this chapter gives a summary of the used media-on-demand transmission schemes which have been evaluated to be favored in each of the cases.

6.1 Cost Analysis of Media-on-Demand Systems

As stated above, the comparison in this chapter is driven purely by financial aspects. Therefore, it is most important to analyze the business model for media-on-demand and evaluate costs and revenue. In the following subsections, the foundation for the cost analysis are observed, starting with the value chain for media-on-demand in section 6.1.1, continuing with a layout of preconditions which are assumed in the analysis in section 6.1.2 and finishing with a composition of costs and revenue for media-on-demand providers in section 6.1.3.

6.1.1 Value Chain

The value chain for media-on-demand involves a series of players [Poi04]: Starting with the content providers who originate the media streams, aggregators negotiate distribution rights and market packages of media streams to service providers. The service providers then distribute the media streams to the end users using a network infrastructure which may be owned by yet another party. Table 6.1 gives a more detailed overview of the participants in the value chain, their function and some well-known examples.

Player	Role	Examples
Content Provider (movie studio, broadcaster, independents)	Commission, production, marketing; own and develop the brand, own the copyright	Disney, Time Warner, BBC, MTV, Star, CNN, Fox Kids, ...
Content Aggregator	Encoding and managing media-on-demand content, negotiate rights with multiple content providers, market complete package and deliver it to e. g. video service providers	FastWeb, Anytime, Arrivo, ODG
Service Providers	Manage and deliver media-on-demand (and other media services) to end-users, perform billing with end-users	FastWeb, Broadmedia, KIT, Video Networks
Broadband network operators	Provide network infrastructure to transmit media streams	FastWeb, Yahoo! Japan, France Telecom, BT, Comcast
End-user	Consumes the media streams	

Table 6.1: Value chain for media-on-demand

The splitting of revenue across the players of the value chain is still contentious. For traditional video broadcast, often a split of 50:25:25 for content provider, content aggregator and video service provider (where the service provider pays for the broadband network) is quoted [Poi04], but the actual split varies greatly from country to country. For the most valuable media streams, content providers will often demand 60 % or even more of the revenue. Additionally, some play-

ers choose to occupy multiple positions in the value chain, making these estimations even more difficult.

For media-on-demand, the division may change as the network costs are significantly higher. The authors of [Moo05] state that a split of 60:40 for content provider and content distributor is yet commonly used in the context of video-on-demand but this change of the established split is one of the reasons which causes the major content providers to be concerned about media-on-demand. New media-on-demand transmission schemes may be a way to lower the transmission costs, both increasing the revenue for the content provider and the service provider and preventing a failure of this type of service due to discordance of the split to apply.

6.1.2 Preconditions

In the cost analysis of this chapter, the following preconditions are assumed:

Number of connected receiver systems: As mentioned above, this analysis covers three application cases of different size: For the small environment, it is assumed that 200 receiver systems are connected to the media-on-demand system, for medium-sized installations, 20 000 receiver systems are assumed and for the large installation, the analysis is based on 2 000 000 connected media-on-demand receiver systems.

Peak usage: In lifelike environments, it is very improbable that all connected media-on-demand receiver systems are receiving a media stream at the same time. But as the bandwidth of reactive media-on-demand transmission schemes has to be calculated depending on the number of concurrently active systems, some assumptions have to be made about:

For small environments (e. g. in hotels, planes or trains), it is assumed that 160 receiver systems are receiving data from the media-on-demand system concurrently, for medium-sized environments 5 000 concurrently active receiver systems are assumed and for large environments 500 000 receiver systems can be concurrently active. This corresponds to a concurrent usage rate of 80 % for small environments and 25 % for medium-sized and large environments [Poi04].

Please note that the number of active receiver systems may also differ depending on the place of installation (e. g. in planes the offer may be used more intensive and more simultaneously than in hotels) and on the popularity of the service.

Request pattern: It is assumed that media stream requests follow the Zipf distribution with a parameter of 0.729 (see section 3.1.1).

Media Type: This analysis only examines the most popular type of media-on-demand: The on-demand transmission of audio-visual content, essentially movies.

Media Quality: As more and more customers have been indulged by the media quality of DVD and CD, they expect a high video and audio quality for media-on-demand, too. But as bandwidth requirements are an important cost factor, a video quality slightly below DVD and a moderate audio quality are presumed to fulfill the customers' demands.

Media Bit Rate: The media bit rate depends on the video compression scheme and the compression rate which determine the video quality. Using MPEG-2 video encoding, 1.5 Mbit/s corresponds nearly to VHS quality while DVDs typically use 3–10 Mbit/s in variable-bit-rate encoding. If H.264/MPEG-4 Part 10/AVC is used instead, the bit rate can be reduced by a factor of two to three [GC01], resulting in an average bit rate of roughly 1.5–3 Mbit/s for DVD video quality. Thus for this analysis, an average media bit rate of 2 Mbit/s is assumed.

Reactivity: Both pro-active and reactive environments are examined.

Multicasting: For pro-active environments, a multicast-capable network is assumed, for reactive environments, multicasting is not considered in this analysis.

Transmission Schemes: This analysis focuses on two transmission schemes: Point-to-Point transmission scheme and the Generalized Greedy Broadcasting Scheme. Point-to-Point transmissions have been selected for several reasons: On the one hand, they are very common in the context of media-on-demand (nearly all media-on-demand services currently available are based on Point-to-Point transmissions, see appendix E), on the other hand, they place very few requirements on the receiver systems. As alternative to this simple transmission scheme, the newly proposed Generalized Greedy Broadcasting scheme has been selected because of its high efficiency and general utilizability.

Although it is known that multicast-based transmission schemes perform better than unicast-based ones if the multicast data is received by several receiver systems at once [CC01], the sender and receiver system requirements are higher for multicast-based systems. Through this analysis, the threshold when the bandwidth savings of a multicast-based system overweight the increase in complexity and costs of multicast-based transmission schemes are examined.

Additionally, a hybrid setup is examined where the seldom requested media streams are sent using Point-to-Point transmissions and often requested media streams using the Generalized Greedy Broadcasting Scheme.

Media assortment: In environments where the customers only stay relatively short (e. g. hotels, air planes, trains or ships), a small media assortment (around ten media streams) is most probably sufficient. In other installations (e. g. media-on-demand systems for home entertainment), a larger assortment (e. g. with several thousands of media streams) may be necessary to accomplish customer demands.

This analysis compares the results for media assortments of the different sizes between 10 and (where applicable) 1 000 media streams, detailing the calculations for 20 titles for the small environment, 50 titles for the medium-sized setup and 200 titles for the large installation.

Partial preloading: Partial preloading can be used to reduce bandwidth requirements as described in section 4.6. For this analysis, the case that no, a fraction of $\frac{1}{60}$ or a fraction of $\frac{1}{12}$ of the media streams are preloaded are examined. For the latter two cases, 30 GB or 150 GB of additional storage are required if the beginnings of 1 000 media streams of an average media bit rate of 2 Mbit/s are preloaded, resp.

Playback Delay: Many of the more efficient transmission schemes (and all pro-active schemes which do not use partial preloading) cannot provide immediate service; the customers have to wait for a playback to commence. As customers may refuse the service if they have to wait too long for the media playback and as they may have been indulged from media-on-demand services with immediate service, a new service must accomplish a short playback delay to be competitive.

Thus, this analysis covers the results for three different playback delays: The results for immediate service (if possible), for playback delays of $\frac{1}{120}$ and for delays of $\frac{1}{60}$ of the media stream duration are examined (where the latter two correspond to playback delays of one and two minutes for a two hour media stream, resp.).

As it is assumed that media streams are typically requested in a Zipf-distribution like manner (see above), it may be reasonable to accommodate the playback delay to the request rate: Therefore, an alternative setting is evaluated, too, where the most popular 5 % of all media streams are served with immediate service, the next 20 % of the media streams in popularity with a playback delay of $\frac{1}{120}$ and the remaining media streams with a playback delay of $\frac{1}{60}$. Using this partitioning, about 31 % of all requests are served with a playback delay of $\frac{1}{120}$ of the media stream duration, another 30 % with a playback delay of $\frac{1}{60}$ and the remainder with a delay of $\frac{1}{24}$, resulting in an average playback delay of $0.009 \cdot d$ (which equals approximately 64.8 seconds for a two hour media stream).

Breaks: No breaks are introduced to the media stream transmissions.

Error Correction: It is assumed that no ARQ-based error recovery but only forward error correction is performed. Therefore, 5 % redundancy data is added to the transmission.

Protocol Overhead: When transmitting media data, each protocol in the used protocol stack adds some data and thus increases the bandwidth requirements. For example, to send media stream data using segmented media-on-demand RTP (see section 5.4) over UDP and IP, 56 octets overhead per RTP packet are produced. For 1 500 octet IP packets, this results in

an overhead of 3.73 %, thus a gross media bit rate (including FEC and protocol overhead) of 2.178 Mbit/s is used for the following calculations.

Bandwidth Requirements: To analyze the costs of a transmission, the bandwidth requirements have to be determined beforehand: According to the examination of transmission schemes in chapter 3 and 4, the media bit rate, the bandwidth requirements of the used transmission scheme and the desired playback function influence the required bandwidth.

Depending on the transmission scheme, a multiple of the media bit rate is needed to get the transmission bandwidth:

- In environments which do not support multicasting, only Point-to-Point transmissions can be used. In this case, the maximum needed bandwidth at the sender system is the product of the number of concurrently active receiver systems and the gross media bit rate (2.178 Mbit/s). Table 6.2 shows the resulting bandwidth requirements for the Point-to-Point Transmission scheme for the described setups.
- Table 6.3 shows the data for the Generalized Greedy Broadcasting scheme for all setups specified above. In all cases, the sender bandwidth is much lower compared to the Point-to-Point transmission scheme because of the high efficiency of the Generalized Greedy Broadcasting scheme, but the requirements on the receiver systems are higher.

Storage size of receiver systems: For the Generalized Greedy Broadcasting scheme, it is assumed that receiver systems provide space for storing up to about 43 % of the media stream (which corresponds to about 774 MB in case of a two hour media stream of an average bit rate of 2 Mbit/s). In case of partial preloading, additional storage space is necessary as described above.

Network type: Depending on the size of the environment, different types of networks are used. In small environments, a local area network (e. g. based on Ethernet) provides enough bandwidth: If Point-to-Point transmissions are used to transmit media streams of 2.178 Mbit/s,

Size of environment	Number of active rcv.	Total bw. in Mbit/s
Small	160	347.7
Medium	7500	16297.5
Large	200000	434600.0

Table 6.2: Bandwidth requirements for Point-to-Point transmissions

Playback delay	Amount of preloading	Req. ch. per stream	Avg. bw. per stream. in Mbit/s
0	$\frac{d}{60}$	4.12	8.98
0	$\frac{d}{12}$	2.50	5.45
$\frac{d}{120}$	0	4.84	10.54
$\frac{d}{120}$	$\frac{d}{60}$	3.72	8.10
$\frac{d}{120}$	$\frac{d}{12}$	2.41	5.25
$\frac{d}{60}$	0	4.14	9.02
$\frac{d}{60}$	$\frac{d}{60}$	3.44	7.49
$\frac{d}{60}$	$\frac{d}{12}$	2.33	5.07
varying	$\frac{d}{60}$	varying	7.75
varying	$\frac{d}{12}$	varying	5.12

Table 6.3: Bandwidth requirements for Generalized Greedy Broadcasting transmissions

100 Mbit/s Ethernet can accomplish 34 transmissions and 1 Gbit/s Ethernet allows transmitting 344 media streams (assuming a network utilization of 75 % in both cases[1]).

For transmissions using the Generalized Greedy Broadcasting scheme, a bandwidth of up to 10.52 Mbit/s is required for each media stream (see table 6.3). This limits the number of broadcasts which can be sent on 100 Mbit/s Ethernet to seven and on 1 Gbit/s Ethernet to 71 for this case (again assuming a network utilization of 75 %). The network layout is more complicated, too, but if no media-on-demand sender system provides more than 71 media streams and no switch where receiver systems are connected to serves more than 71 receiver systems, it is sufficient to connect all media-on-demand sender systems and switches to a fast "root" switch via 1 Gbit/s which has a backplane of at least 3 Gbit/s to support 200 receiver systems. Figure 6.1 illustrates this setup.

Wireless local area networks which provide enough bandwidth for the described media-on-demand application cases are not yet available and thus not included in this analysis: Currently, IEEE 802.11g only provides three and IEEE 802.11a eight non-overlapping 20 MHz bands, each providing a gross bit rate of 54 Mbit/s and a net bit rate (after removing physical and data link layer overhead) of 37 Mbit/s[2]. Assuming a media stream of 2.178 Mbit/s and a network utilization limit of 75 %, 12 concurrent Point-to-Point transmissions or 2 Generalized Greedy-based stream transmissions are possible on each band. Unfortunately, these bit rates are not achievable in practical environments, esp. inside build-

[1] A network utilization of 75 % should be possible in this setup if switched Ethernet is used, but if other traffic than media-on-demand is also served through this network, precautions have to be taken.

[2] IEEE 802.11n (standardization not finished yet) will allow wireless local networks reach bandwidths of up to 300 Mbit/s with a net throughput of up to 100 Mbit/s, which may become more interesting for media-on-demand.

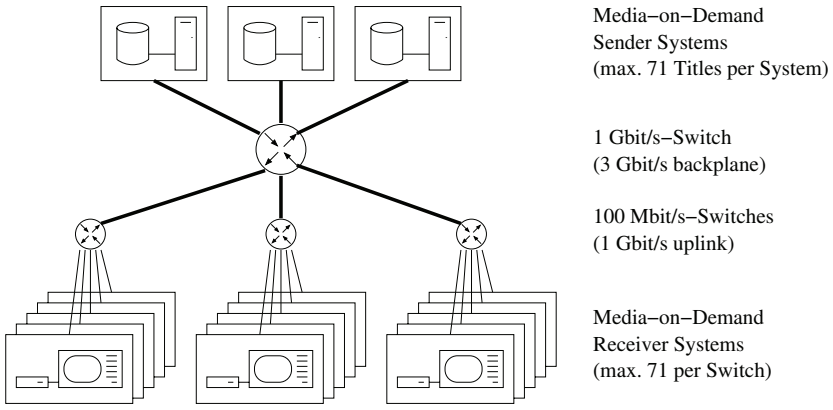


Figure 6.1: Structure for a setup where the Generalized Greedy Broadcasting scheme can be used to serve 200 receiver systems using 100 Mbit/s and 1 Gbit/s Ethernet only

ings, where this bit rate decreases rapidly for distances above 25 meters. Therefore, the use of wireless local area networks is not applicable for the analyzed environments.

Medium-sized environments require larger types of networks, e. g. metropolitan to wide area networks or several interconnected small networks. Examples for suitable metropolitan and wide area networks are based on Asynchronous Transfer Mode (ATM) networks [Rig98, Gin96a] (currently limited to 2 488 Mbit/s if used for IP transmissions due to the technical complexity from segmentation and reassembly for IP packets in ATM cells), Packet over SDH/SONET (POS) [MS99] (up to 10 Gbit/s if used for IP transmission) or Metro Ethernet [San03] (up to 10 Gbit/s, too). Approaches to use ATM without the Internet layer for media-on-demand [Gin96b] — either in point-to-point or in point-to-multipoint mode — have not spread widely due to missing ATM connectivity of private households.

If households have to be connected to a broadband network, installation is expensive if not performed on top of existing infrastructure: For example, in case of DSL (Digital Subscriber Line)-based Internet access, the high frequency spectrum of coppery telephone lines is used to transmit data between the telephone central office and households. (Other examples for data transmission over existing infrastructure are Internet over cable TV networks or over power lines.)

Major parts of the Internet are not multicast-capable yet, so at a first glance, it looks questionable if multicast-based media-on-demand transmission systems can be used in this scenario. But fortunately, it is not required that the whole Internet is multicast-capable but just the DSL access concentrator: For DSL-based Internet access, the media-on-demand sender system can be placed at the DSL access concentrator[3] and communicate with all receiver

[3]DSL lines are typically not connected to the Internet at the central office: At the central office, the DSL traffic is

systems which are bound to this access concentrator[4]. The number of systems which can be reached this way depends very much on the number of DSL customers and the number of DSL access modulators which are connected to the access concentrator. In case of Deutsche Telecom AG, 74 DSL access concentrators are used to connect around 5 million DSL customers (as of year 2004), so roughly 67 000 systems can be reached in average at each access concentrator which matches roughly the medium-sized environment case. Alternatively, if the media-on-demand server cannot be placed directly at the DSL access concentrator, a multicast tunnel can be established from the media-on-demand sender systems to the DSL access concentrator.

Deutsche Telecom AG also started to install Outdoor-DSLAM systems which provide up to 50 Mbit/s in downstream to the home. These systems are equipped with IPTV-aware VDSL2 hardware which already supports multicast in downstream direction.

As DSL and similar techniques like the aforementioned cable TV or power line Internet access methods provide currently the best affordable high-speed Internet access which is usable for media-on-demand, DSL-based Internet access has been selected for medium-sized environments.

For large environments, it is nearly impossible to circumvent wide area networks. One type of wide area networks which is already popular for traditional media transmissions is built by satellites: Satellites allow to transmit data to all inhabitants of a country or even continent at once and thus fit with ease in the large-sized environment case.

Alternatively, DSL-based Internet access can be used in large environments, too: In this case, media-on-demand sender systems can be placed at several DSL access concentrators or a single media-on-demand service location can be selected which sends the media streams to all access concentrators.

6.1.3 Costs and Revenue of Media-on-Demand

This subsection examines the different types of costs and sources of revenue for media-on-demand providers and estimates their amounts which are used for the calculations in the next three sections.

only split from the telephone lines using DSL access modulators and then sent (e. g. via ATM, sometimes also using fiber GBit-Ethernet) to the access concentrator where routing to the Internet is performed.

[4]Some DSL-based video broadcast systems already utilize multicasting for sending television data over DSL, e. g. the mPhaseTV+ platform, consisting of an mPhase Broadcast Television Switch (mPhase Technologies) and a Stinger DSL Access Concentrator (Lucent Technologies).

Costs for Providing Media-on-Demand

The costs for providing a media-on-demand service can be divided in two parts: Some of the costs are general to any media-on-demand service, others depend on the used transmission technique.

Although the comparison is just based on the costs which depend on the used transmission technique, other costs are given here as found in literature for completeness:

Sender system equipment: For sending media streams to end-users, the service provider has to operate a media-on-demand sender system. Depending on the type of the media transmission scheme, this system can be small (e. g. for pro-active systems which are connected to a multicast/broadcast environment) or a huge server farm (e. g. if Point-to-Point transmissions are used in a medium-sized or large environment), centralized (if all receiver systems can be reached from a single point, e. g. in case of satellite transmissions) or distributed (e. g. if a sender system is required at each DSL access concentrator to provide media-on-demand over DSL).

According to [Poi04], a Point-to-Point media-on-demand sender systems costs roughly USD 200.00 (approximately EUR 126.00) per concurrent playback stream.

In case of pro-active systems, the costs are independent of the number of receiver systems. For simplicity, it is assumed that a sender system costs EUR 630.00 per media title for the Generalized Greedy Broadcasting scheme as Generalized Greedy Broadcasting transmissions use up to five channels in the examined setups (see bandwidth requirements item of section 6.1.2).

Receiver system equipment: A typical media-on-demand receiver system has to perform the following tasks:

- Requesting a media stream (in case of reactive transmission schemes and if not performed by other means, e. g. telephone),
- reception of the media stream,
- buffering (if required by the transmission scheme),
- reassembling (if required by the transmission scheme),
- decryption (if encryption is used),
- error recovery (if supported),
- decoding (for compressed data, at least conversion into the output format),
- presentation on output devices (e. g. video screen, speakers).

To save expenses for presentation, providers often incorporate TV systems into the receiver system equipment for playback of media streams. An estimation of the price of a suitable

set-top box can be performed on the base of existing set-top boxes which have been designed for similar purposes: Table 6.4 shows an overview of several boxes of different type. All set-top boxes in this table are capable of decoding MPEG-2 (and some of them even of other formats with a higher compression ratio) and provide a broadband network interface and a TV outlet. The prices vary much depending on the type of the broadband network, where DVB-S systems seem to be the cheapest ones and DVB-C the most expensive ones. Systems without hard disk in this table start at EUR 29.95, with hard disk at EUR 99.00. Even if these set-top boxes often not contain a processor which is powerful enough to support advanced media-on-demand transmissions, they give a hint about the current level of prices for set-top boxes.

The last four entries in the table are of special interest: The first one is a gaming console which even provides much more processor power and hardware support than needed in media-on-demand environments, the second one is a set-top box which has been developed for Internet-based television (including pay TV and video-on-demand) as well as other Internet-based services like email and chat. Unfortunately, it is not possible to ignore that these two systems were sold under production price: Both the WebTV service and the game console business were deficit branches of Microsoft which were cross-subsidized in 2005 [CNE01, Inq05]. In 2007, Microsoft merged these systems and provides a video-on-demand service on its new Xbox 360 gaming console [Xbo07, CNE07] (second last item). Finally, the last item is a digital video recorder which provides a hard disk and MPEG decoder as needed for media-on-demand.

For the calculations in the following sections, it is assumed that suitable set-top boxes without storage can be produced for EUR 100.00. For pro-active media-on-demand transmission schemes which require a huge amount of storage, hard disk based solutions should be preferred to RAM-based ones: Systems with 150 GB hard disk storage should be available for around EUR 160.00[5], systems with 2 GB RAM or 8 GB Flash for around the same price. As only hard disk-based systems will provide enough storage space for partial preloading, memory based ones will not be included in this analysis. If partial preloading is not used, RAM- or Flash-based systems may be preferred because they are much less sensitive to physical stress.

Another approach is to use the customers' personal computers as media-on-demand receiver systems, eliminating the receiver equipment costs completely.

Sender and receiver system software: Both for the sender and the receiver systems, software is required for providing media-on-demand services. According to [Poi04], USD 150.00 (ap-

[5] Currently, hard disks with a capacity of 150 GB and available for around EUR 35.00.

Set-top box type	Brand	Capabilities	Price
DVB-T receiver box	ComAG SL 25T	Terrestrial DVB tuner, MPEG-2 decoder, TV-out	EUR 24.99
DVB-T receiver box	Skymaster DVRT 1000	Terrestrial DVB tuner, MPEG-2 decoder, TV-out, 80 GB hard disk	EUR 99.95
DVB-S receiver box	Skymaster DX 15	Satellite DVB tuner, MPEG-2 decoder, TV-out	EUR 22.99
DVB-S receiver box	Medion MD24500	Satellite DVB tuner, MPEG-2 decoder, TV-out, 40 GB hard disk	EUR 99.00
DVB-C receiver box	TechnoTrend micro C202	Cable DVB tuner, MPEG-2 decoder, TV-out	EUR 59.30
DVB-C receiver box	Humax PDR 9700C	Cable DVB tuner, MPEG-2 decoder, TV-out, 80 GB hard disk	EUR 248.00
Gaming console	Microsoft Xbox	MPEG-2 decoder, TV-out, DVD drive, 8 GB hard disk, Ethernet	EUR 129.90 (not sold any more)
IPTV receiver box	Microsoft MSN TV2	Windows Media 9 decoder, wireless keyboard, USB 2.0 (e. g. for connecting a hard disk), intended for cable, satellite or DSL use	USD 79.00–179.00 (not sold any more)
Gaming console	Microsoft Xbox 360 Premium System	MPEG-2 decoder, TV-out, DVD drive, 20 GB hard disk, Ethernet	EUR 253.99
Digital video recorder	LG Electronics RH-256	MPEG-2 decoder, TV-out, DVD drive, 160 GB hard disk	EUR 159.00

Table 6.4: Examples of set-top boxes of different type

proximately EUR 94.30) had been accounted for by Kingston Communications when they set up their Point-to-Point-based media-on-demand service. In [Kuh99], the total costs for Point-to-Point media-on-demand sender systems (hardware and software) are estimated as USD 350.00 (approximately EUR 220.00) per receiver system which roughly acknowledges the price estimates for hardware and software.

In case of Generalized Greedy Broadcasting media-on-demand systems, these prices differ much as the media-on-demand sender systems do not depend on the number of active receiver systems but mainly on the number of media streams. Hence, EUR 8 000.00 are assessed for sender software per media title and EUR 40.00 for software for each receiver system for the Generalized Greedy Broadcasting scheme which roughly reflects personal experiences.

To make the comparison more balanced, the software costs of Point-to-Point transmissions are also split into a sender and a receiver system part: For receiver systems, software costs

of EUR 20.00 per receiver system are assumed which should reflect that receiving a media stream on a reactive system is less complicated than for Generalized Greedy Broadcasting scheme systems. For sender systems, costs of EUR 74.00 remain per receiver system according to the above stated total software costs which corresponds to EUR 370.00 per concurrently active transmission if a peak usage of 20 % is assumed (which has been used for the price estimation in [Poi04]).

For the hybrid case, the most popular media streams are sent using the Generalized Greedy Broadcasting scheme and the remaining streams using Point-to-Point transmissions. Thus for the sender systems, there will be both types of software present. For the receiver systems, it is assumed that the Generalized Greedy Broadcasting receiver software is also capable to play Point-to-Point transmissions[6].

Network installation and operation: Media transmissions require a huge amount of bandwidth, and depending on the used transmission scheme, the costs for setting up the network infrastructure and utilizing it for media-on-demand transmissions differ:

For small-sized environments, Ethernet equipment is very cheap today (e. g. managed switches are available starting around EUR 6.00 per port for 100 Mbit/s and around EUR 20.00 per port for 1 Gbit/s). In addition to the hardware, costs from installing cables have to be considered. A study [Cis04] estimated the installation costs for CAT5 Ethernet cables in a hotel to USD 150.00 per port.

For providing media-on-demand via DSL, the authors of [Poi04] listed capital costs of USD 250.00 for ADSL lines and DSL access modulators per household.

For satellite-based transmissions, it is not assumed that the provider owns the network (launching rockets with satellites is very expensive) but that satellites are leased. Unfortunately, costs for satellite transmissions are very high: Leasing a transponder channel of a satellite (with a net capacity of around 47 Mbit/s in case of DVB-S2 using 8PSK 3/5 modulation) causes annual costs between EUR 1 000 000 (which is used in this analysis) up to EUR 6 000 000 for satellites at commonly used sky positions. Additionally, receiver equipment for around EUR 100.00 per receiver system is required.

Media license fees: Content providers and aggregators request a license fee for distribution of media streams. Typically, these fees are negotiated with the content providers or aggregators which request a given percentage from the takings.

Deployment costs: For setting up a media-on-demand service, an initial huge amount of money has to be invested: The hardware has to be installed, the service has to be promoted and

[6] If RTP translators are used at the receiver systems for segmented media-on-demand reception, this assumption should be valid.

service employees have to be trained. In [Kuh02], the deployment costs are estimated as USD 300 000.00 (approximately EUR 188 000.00) for a large environment installation, but smaller installations should be much cheaper.

Operational expenses: To maintain the system and to supply required resources as well as managing customers and company needs, several additional costs have to be considered (e. g. room rental, electric power supply, telephone support for customers, billing of customers, promotion, accounting). These expenses seem to vary much depending on the operator: According to [Kuh99], these expenses have averaged USD 4.00 and are still falling. Some companies already reached USD 1.20 per month and subscriber in year 1999 which is the latest reliable statement.

Capital recovery: As media-on-demand providers like to get their investments returned after some time, a recovery time for the capital costs has to be defined. According to [Poi04], a capital recovery after five years is typical in this class of business.

To compare the two different transmission techniques, only hardware and software costs and for satellite-based environments network costs are relevant. Table 6.5 summarizes these costs in a table.

Sender systems:			
EUR	126.00	per active rcv.	for Point-to-Point transmissions
EUR	630.00	per title	for Generalized Greedy Broadcasting transmissions
Receiver systems:			
EUR	100.00	per receiver	for Point-to-Point transmissions
EUR	160.00	per receiver	for Generalized Greedy Broadcasting transmissions
Sender software:			
EUR	370.00	per active rcv.	for Point-to-Point transmissions
EUR	8000.00	per title	for Generalized Greedy Broadcasting transmissions
Receiver software:			
EUR	20.00	per receiver	for Point-to-Point transmissions
EUR	40.00	per receiver	for Generalized Greedy Broadcasting transmissions
Network bandwidth:			
EUR	1 000 000.00	per transp. and year	for satellite-based environments

Table 6.5: Overview of costs which depend on the used transmission technique

Revenue from Media-on-Demand

The following sources of revenue from media-on-demand services can be opposed to the above described costs:

Revenue from rental or selling of receiver system equipment: If a receiver system equipment is required (i. e. if not an existing system such as a personal computer is used), the provider may rent or sell receiver systems to its customers. On the other hand, it is also possible to subsidize the equipment in order to push the service.

In this analysis, the media-on-demand receiver system costs are allocated completely to the monthly fees, based on a capital recovery of five years.

Pay-per-view charges or periodical customer fees: For using the media-on-demand service, the provider may charge a periodical fee from the customer and/or a rate per received media-stream. Typical prices range from below EUR 1.00 for filmlets or TV episodes up to around EUR 2.50–4.00 for new titles (e. g. for T-Online videoload, Comcast pay-per-view or Microsofts IPTV services).

Setup and registration fees: Initial setup fees are very uncommon for media-on-demand. In cases where costs are caused from the receiver system equipment or the installation of the broadband network access, these costs are typically allocated to monthly fees.

Advertising revenue: Similar to private television broadcasting, advertisements can be used to cover some of the expenses of the media-on-demand service. But depending on the local habits, advertising for a paid service may not be accepted by customers.

A possible solution would be a twofold offer which allows the customer to choose between a cheap, advertisement sponsored transmission and a more expensive transmission without any commercial breaks.

Besides these, several indirect benefits may be achieved (e. g. increase of the company image and value). As none of these benefits depend much on the used transmission technique, they are ignored in the comparison.

6.2 Evaluation of Media-on-Demand in Ethernet-Based Environments

For small environments, high bandwidth networks are typically available for relatively low costs: Switched Ethernet excellently fulfills the requirements for both Point-to-Point media-on-demand transmissions and transmissions using a pro-active scheme like the Generalized Greedy Broadcasting scheme. Thus the efficiency of the transmission scheme plays only a minor role, the evaluation is mainly influenced by the costs of the sender and receiver system equipment and software.

Consequently, this has a major impact on the cost-effectiveness of the different media-on-demand system approaches: The following two subsections present a sample calculation for

a small, Ethernet-based media-on-demand system for both of the two examined transmission schemes.

6.2.1 Point-to-Point Transmissions

This setup allows to use the cheap, storage-free receiver systems, thus the transmission technique-dependent costs are moderately low as shown in table 6.6. If the customers' personal computer systems are used as media-on-demand receiver systems, the costs can be lowered by EUR 20 000.00 (EUR 100.00 per receiver).

Item	Costs			Total	
Sender systems (PtP)	EUR	126.00	per active rcv.	EUR	20 160.00
Receiver systems (PtP)	EUR	100.00	per receiver	EUR	20 000.00
Sender software (PtP)	EUR	370.00	per active rcv.	EUR	59 200.00
Receiver software (PtP)	EUR	20.00	per receiver	EUR	4 000.00
Costs	EUR	516.80	per customer	EUR	103 360.00

Table 6.6: Transmission technique-dependent costs for Ethernet-based Point-to-Point transmissions

6.2.2 Generalized Greedy Broadcasting Transmissions

This setup requires the use of set-top boxes which are capable of storing part of the media stream. For hard disk-based set-top boxes, the transmission technique-dependent costs are shown in table 6.7 for a media assortment of 20 titles. Results for media assortments of other size are shown in table 6.8. Other choices in the examined setups (e. g. playback delay and amount of preloading for Generalized Greedy Broadcasting scheme) do not influence the results in this environment because the used network costs are very low and cannot be reduced this way. If the customers' personal computer systems are used as media-on-demand receiver systems, the costs can be lowered in this setup by EUR 32 000.00 (EUR 160.00 per receiver).

Item	Costs			Total	
Sender systems (GGB)	EUR	630.00	per title	EUR	12 600.00
Receiver systems (GGB)	EUR	160.00	per receiver	EUR	32 000.00
Sender software (GGB)	EUR	8 000.00	per title	EUR	160 000.00
Receiver software (GGB)	EUR	20.00	per receiver	EUR	8 000.00
Costs	EUR	1 063.00	per customer	EUR	212 600.00

Table 6.7: Transmission technique-dependent costs for Ethernet-based Generalized Greedy Broadcasting transmissions

Size of assortment	Costs (per customer)	Costs (total)
10	EUR 631.50	EUR 126300.00
20	EUR 1063.00	EUR 212600.00
50	EUR 2357.50	EUR 471500.00
100	EUR 4515.00	EUR 903000.00

Table 6.8: Transmission technique-dependent costs for Ethernet-based Generalized Greedy Broadcasting transmissions assuming different media assortment sizes

6.2.3 Hybrid Setup

In the hybrid setup, the receiver systems must be capable to receive a media stream using the Generalized Greedy Broadcasting scheme, so the receiver systems will be the same as in the previous setup. For the sender systems, a balance between Point-to-Point transmissions and Generalized Greedy Broadcasting transmissions must be found: For the presumed sender system and sender software costs, the Generalized Greedy Broadcasting scheme is more advantageous than separate Point-to-Point transmissions for titles with at least 18 active receivers. Assuming the Zipf distribution for the popularity of the titles, this is only the case for the most popular two titles in case of a medium assortment of 20 titles. Table 6.9 shows the composition of the costs for this setup and table 6.10 the results for different assortment sizes.

Item	Costs			Total	
Sender systems (PtP)	EUR	126.00	per active rcv.	EUR	13986.00
Sender systems (GGB)	EUR	630.00	per title	EUR	1260.00
Receiver systems (hybrid)	EUR	160.00	per receiver	EUR	32000.00
Sender software (PtP)	EUR	370.00	per active rcv.	EUR	41070.00
Sender software (GGB)	EUR	8000.00	per title	EUR	16000.00
Receiver software (hybrid)	EUR	20.00	per receiver	EUR	8000.00
Costs	EUR	561.58	per customer	EUR	112316.00

Table 6.9: Transmission technique-dependent costs for Ethernet-based hybrid Point-to-Point/Generalized Greedy Broadcasting setups

Size of assortment	Titles PtP/GGB	Active rcv. PtP/GGB	Costs (per customer)		Costs (total)
10	7/3	75/85	EUR 515.45	EUR	103090.00
20	18/2	111/49	EUR 561.58	EUR	112316.00
50	49/1	139/21	EUR 587.87	EUR	117574.00
100	100/0	160/0	EUR 596.80	EUR	119360.00

Table 6.10: Transmission technique-dependent costs for Ethernet-based hybrid Point-to-Point/Generalized Greedy Broadcasting setups assuming different media assortment sizes

6.2.4 Result

For Ethernet-based, small environments, Point-to-Point media-on-demand systems are significantly less expensive than Generalized Greedy Broadcasting transmissions as shown in figure 6.2. Even a hybrid setup of Generalized Greedy Broadcasting for the most popular titles and Point-to-Point transmissions for the remaining ones is more expensive than a pure Point-to-Point streaming setup. The main reason for this is that there are too few recipients for the same media title to compensate for the higher costs caused by Generalized Greedy Broadcasting for systems and software.

Only if the media assortment is very small (consisting of up to seven streams for a Generalized Greedy Broadcasting setup or up to ten streams for a hybrid setup), the Generalized Greedy Broadcasting scheme provides lower costs: If only very few media streams are sent, the lower sender system costs for the Generalized Greedy Broadcasting scheme compensate for the higher receiver system costs.

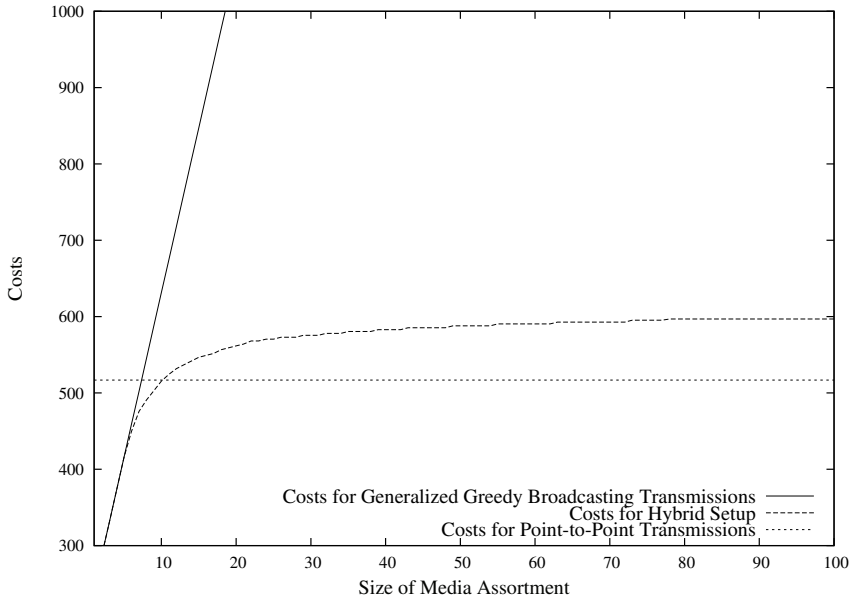


Figure 6.2: Transmission technique-dependent costs for Ethernet-based setups in relation to the media assortment size

6.3 Evaluation of Media-on-Demand in DSL-Based Environments

In environments where customers already have DSL-based Internet access, media-on-demand services can be provided in a very simple way. With the enormous spread of DSL-based Internet accesses, this type of media-on-demand has become very popular in recent years, nearly each Internet service provider offers a video-on-demand service.

The bandwidth requirements of Point-to-Point media-on-demand transmissions are around 2.178 Mbit/s for each active transmission which can be accomplished in many cases where DSL is available if only one media stream per DSL line is consumed concurrently. If pro-active media-on-demand transmission schemes should be used, the bandwidth requirements are much higher because several channels have to be received at once (e. g. up to 10.52 Mbit/s for the examined setups, see table 6.3). Although 6 Mbit/s DSL is often supported and even higher rates are in preparation[7], these bandwidths may not be achievable for all customers. Fortunately, the Generalized Greedy Broadcasting scheme can support the case that less bandwidth is available:

- The Generalized Greedy Broadcasting scheme provides support for limited receiver bandwidth (see section 4.18) which makes it possible to transmit the media stream in such a way that only a fraction of the channels has to be joined at a time.
- The Generalized Greedy Broadcasting scheme supports channels of arbitrary bandwidth (see section 4.10) which allows to select the bandwidth of channels independent from the media stream bit rate.
- The Generalized Greedy Broadcasting scheme supports layered encodings (see section 5.2) so poorly connected receiver systems can receive media streams at reduced quality.

Using these enhancements, it is even possible to send a 2.178 Mbit/s media stream with the Generalized Greedy Broadcasting scheme on 3 Mbit/s or 6 Mbit/s DSL lines in full quality or at lower bit rates in reduced quality.

The disadvantage of a low receiver bandwidth is that the sender bandwidth is increased: For example, the sender bandwidth for media-on-demand transmissions with a playback delay of $\frac{1}{120}$ of the media stream duration and no preloading is increased from 10.52 Mbit/s to 17.69 Mbit/s if the receiver bandwidth is limited to 3 Mbit/s and to 11.32 Mbit/s if limited to 6 Mbit/s. As this only increases the network consumption at the DSL access concentrator where the media-on-demand traffic is routed to the DSL lines, this increase is negligible.

The costs for the two media-on-demand transmission schemes compare as follows:

6.3.1 Point-to-Point Transmissions

As for the Ethernet-based environment, this setup allows to use the less expensive, storage-free receiver systems. Table 6.11 shows the transmission technique-dependent costs in this case.

If the customers' personal computer systems are used as media-on-demand receiver systems in this setup (which is very probably because they may be connected to the Internet via DSL either), costs can be lowered by EUR 2 000 000.00 (EUR 100.00 per receiver).

[7] Deutsche Telecom has installed VDSL2 Internet access in the 50 hugest German cities until 2007, providing up to 50 Mbit/s using outdoor DSL access modulators to shorten the distance to the next access modulator.

Item	Costs			Total	
Sender systems (PtP)	EUR	126.00	per active rcv.	EUR	630000.00
Receiver systems (PtP)	EUR	100.00	per receiver	EUR	2000000.00
Sender software (PtP)	EUR	370.00	per active rcv.	EUR	1850000.00
Receiver software (PtP)	EUR	20.00	per receiver	EUR	400000.00
Costs	EUR	244.00	per customer	EUR	4880000.00

Table 6.11: Transmission technique-dependent costs for DSL-based Point-to-Point transmissions

6.3.2 Generalized Greedy Broadcasting Transmissions

Again, this setup requires the use of set-top boxes which are capable of storing part of the media stream. For hard disk-based set-top boxes, the transmission technique-dependent costs are shown in table 6.12 for a media assortment of 100 titles. Results for media assortments of other size are shown in table 6.13. Other choices in the examined setups (e. g. playback delay and amount of preloading for Generalized Greedy Broadcasting scheme) do not influence the results in this environment because the resulting network traffic has no effects on the costs.

Similar to above, if the customers' personal computer systems are used as media-on-demand receiver systems, costs can be lowered by EUR 3 200 000.00 (EUR 160.00 per receiver).

Item	Costs			Total	
Sender systems (GGB)	EUR	630.00	per title	EUR	63000.00
Receiver systems (GGB)	EUR	160.00	per receiver	EUR	3200000.00
Sender software (GGB)	EUR	8000.00	per title	EUR	800000.00
Receiver software (GGB)	EUR	20.00	per receiver	EUR	800000.00
Costs	EUR	243.15	per customer	EUR	4863000.00

Table 6.12: Transmission technique-dependent costs for DSL-based Generalized Greedy Broadcasting transmissions

Size of assortment	Costs (per customer)		Costs (total)	
10	EUR	204.31	EUR	4086300.00
20	EUR	208.63	EUR	4172600.00
50	EUR	221.57	EUR	4431500.00
100	EUR	243.15	EUR	4863000.00
200	EUR	286.30	EUR	5726000.00
500	EUR	415.75	EUR	8315000.00
1000	EUR	631.50	EUR	12630000.00

Table 6.13: Transmission technique-dependent costs for DSL-based Generalized Greedy Broadcasting transmissions assuming different media assortment sizes

6.3.3 Hybrid Setup

As the sender system costs for this setup are the same as for the Ethernet-based setup, a Generalized Greedy Broadcasting causes lower costs for titles with at least 18 active receivers. But as a much higher number of receivers is assumed in this setup, this number is reached for 99 of the 100 titles.

To give a more practical example for a DSL-based hybrid setup, the size of the media assortment is increased to 200 titles. In this case, the 71 most popular titles (which receive around 69 % of all media requests) have enough active receivers to be transmitted using the Generalized Greedy Broadcasting scheme. The resulting transmission technique-dependent costs for this setup are shown in table 6.14 and for other media assortments in table 6.15.

Item	Costs			Total	
Sender systems (PtP)	EUR	126.00	per active rcv.	EUR	192 402.00
Sender systems (GGB)	EUR	630.00	per title	EUR	44 730.00
Receiver systems (hybrid)	EUR	160.00	per receiver	EUR	3 200 000.00
Sender software (PtP)	EUR	370.00	per active rcv.	EUR	564 990.00
Sender software (GGB)	EUR	8 000.00	per title	EUR	568 000.00
Receiver software (hybrid)	EUR	20.00	per receiver	EUR	800 000.00
Costs	EUR	268.51	per customer	EUR	5 370 122.00

Table 6.14: Transmission technique-dependent costs for DSL-based hybrid Point-to-Point/Generalized Greedy Broadcasting setups

Size of assortment	Titles PtP/GGB	Active rcv. PtP/GGB	Costs (per customer)		Costs (total)	
10	0/10	0/5 000	EUR	204.31	EUR	4 086 300.00
20	0/20	0/5 000	EUR	208.63	EUR	4 172 600.00
50	0/50	0/5 000	EUR	221.57	EUR	4 431 500.00
100	1/99	17/4 983	EUR	243.14	EUR	4 862 802.00
200	129/71	1 527/3 473	EUR	268.51	EUR	5 370 122.00
500	453/47	2 800/2 200	EUR	289.72	EUR	5 794 410.00
1 000	966/34	3 442/1 558	EUR	300.03	EUR	6 000 652.00

Table 6.15: Transmission technique-dependent costs for DSL-based hybrid Point-to-Point/Generalized Greedy Broadcasting setups assuming different media assortment sizes

6.3.4 Results

For the preconditions of the DSL-based setup, media-on-demand using Point-to-Point transmissions and the Generalized Greedy Broadcasting scheme cause nearly identical costs. Although the receiver systems are more expensive for media-on-demand based on pro-active schemes, the

Generalized Greedy Broadcasting scheme has lower costs for sender systems. Graph 6.3 shows this for different sizes of media assortments.

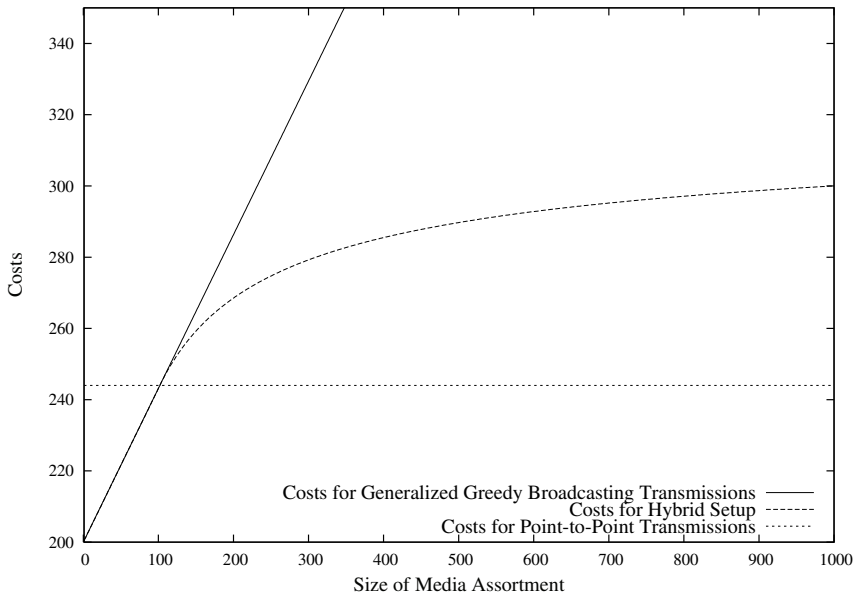


Figure 6.3: Transmission technique-dependent costs for DSL-based setups in relation to the media assortment size

For media assortments with 102 titles or more, the costs of the receiver systems overweigh the savings in the sender systems, so Point-to-Point transmissions are less expensive. But with a growing number of receiver systems, this boundary moves in favor of the Generalized Greedy Broadcasting scheme: For a setup ten times the size of the medium sized setup (so 200 000 recipients and a maximum of 50 000 concurrently active receivers), the Generalized Greedy Broadcasting scheme performs at lower costs for all titles of a media assortment with 1 000 titles as shown in figure 6.4.

6.4 Evaluation of Media-on-Demand in Satellite-Based Environments

Satellite-based networks can be used to access a very large number of customers at once. As bandwidth of satellites is highly valuable, it must not be wasted, thus the efficiency of the transmission scheme gains highest importance. Therefore, it is nearly obvious that Point-to-Point transmissions are ineffective in this setup:

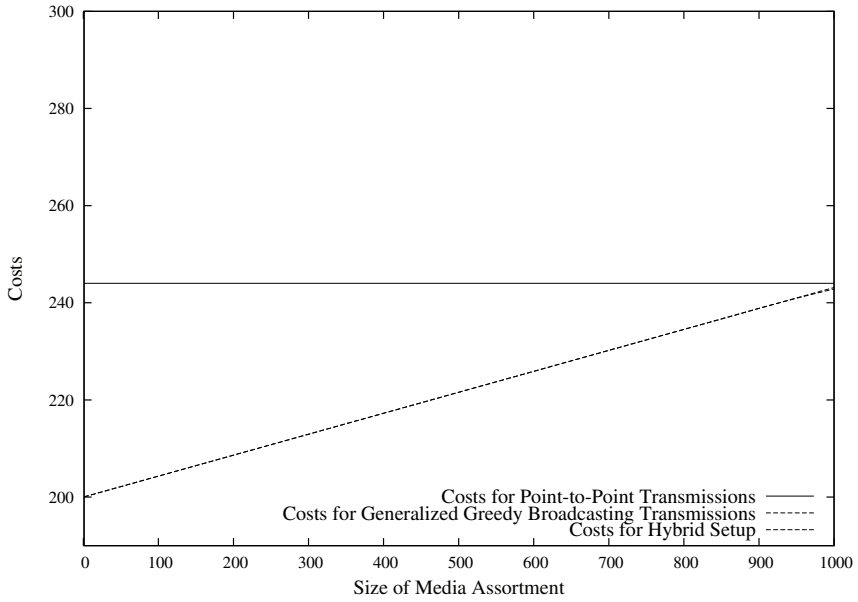


Figure 6.4: Transmission technique-dependent costs for a larger DSL-based setup in relation to the media assortment size

6.4.1 Point-to-Point Transmissions

Table 6.16 shows the transmission technique-dependent costs for the satellite-based environment. The costs are much lower than for the smaller setups because for a lower number of concurrently active receiver systems has been assumed.

Item	Costs			Total
Sender systems (PtP)	EUR	126.00	per active rcv.	EUR 63 000 000.00
Receiver systems (PtP)	EUR	100.00	per receiver	EUR 200 000 000.00
Sender software (PtP)	EUR	370.00	per active rcv.	EUR 185 000 000.00
Receiver software (PtP)	EUR	20.00	per receiver	EUR 40 000 000.00
Costs	EUR	244.00	per customer	EUR 488 000 000.00

Table 6.16: Transmission technique-dependent costs for satellite-based Point-to-Point transmissions

In addition to these costs, satellite bandwidth has to be leased, but serving 500 000 concurrently active receivers using Point-to-Point transmissions would require a bandwidth of more than 1 Tbit/s! Besides of the incomprehensible sum for leasing this amount of satellite bandwidth (more than EUR 20 000 000 000 per year according to the prerequisites), this would allocate the total bandwidth of 465 satellites (assuming 50 transponders per satellite), making this scenario pure theory.

6.4.2 Generalized Greedy Broadcasting Transmissions

While the system costs of the this case do not differ much from the Point-to-Point case (see table 6.17 and 6.18), the bandwidth costs are much lower: Broadcasting 200 titles with a varying playback delay between immediate service and $\frac{d}{60}$ depending on media popularity and utilizing $\frac{d}{12}$ of preloading requires a total bandwidth of 1.02 Gbit/s. This equals nearly 23 transponders, thus EUR 23 000 000 per year or EUR 0.96 per month and customer which is more realistic. Results for other combinations of playback delay, preloading and size of media assortment are given in table 6.19.

Item	Costs			Total	
Sender systems (GGB)	EUR	630.00	per title	EUR	126 000.00
Receiver systems (GGB)	EUR	160.00	per receiver	EUR	320 000 000.00
Sender software (GGB)	EUR	8 000.00	per title	EUR	1 600 000.00
Receiver software (GGB)	EUR	20.00	per receiver	EUR	80 000 000.00
Costs	EUR	200.86	per customer	EUR	401 726 000.00

Table 6.17: Transmission technique-dependent costs for satellite-based Generalized Greedy Broadcasting transmissions

Size of assortment	Costs (per customer)		Costs (total)
10	EUR	200.04	EUR 400 086 300.00
20	EUR	200.09	EUR 400 172 600.00
50	EUR	200.22	EUR 400 431 500.00
100	EUR	200.43	EUR 400 863 000.00
200	EUR	200.86	EUR 401 726 000.00
500	EUR	202.16	EUR 404 315 000.00
1 000	EUR	204.31	EUR 408 630 000.00

Table 6.18: Transmission technique-dependent costs for satellite-based Generalized Greedy Broadcasting transmissions assuming different media assortment sizes

6.4.3 Hybrid Setup

The hybrid setup has no advantages to Generalized Greedy Broadcasting transmissions in a large environment: Due to the high number of active recipients, all titles are requested by enough active receiver systems to give the Generalized Greedy Broadcasting scheme the advantage.

Playback delay	Amount of preloading	No. of titles	No. of transp.	Operational costs (per customer)		Operational costs (total)	
0	$\frac{d}{60}$	200	40	EUR	1.67	EUR	3333333.33
0	$\frac{d}{12}$	200	25	EUR	1.04	EUR	2083333.33
$\frac{d}{120}$	0	200	50	EUR	2.08	EUR	4166666.67
$\frac{d}{120}$	$\frac{d}{60}$	200	40	EUR	1.67	EUR	3333333.33
$\frac{d}{120}$	$\frac{d}{12}$	200	25	EUR	1.04	EUR	2083333.33
$\frac{d}{60}$	0	200	40	EUR	1.67	EUR	3333333.33
$\frac{d}{60}$	$\frac{d}{60}$	200	34	EUR	1.42	EUR	2833333.33
$\frac{d}{60}$	$\frac{d}{12}$	200	23	EUR	0.96	EUR	1916666.67
varying	$\frac{d}{60}$	200	34	EUR	1.42	EUR	2833333.33
varying	$\frac{d}{12}$	200	23	EUR	0.96	EUR	1916666.67
0	$\frac{d}{60}$	50	10	EUR	0.42	EUR	833333.33
0	$\frac{d}{12}$	50	7	EUR	0.29	EUR	583333.33
$\frac{d}{120}$	0	50	13	EUR	0.54	EUR	1083333.33
$\frac{d}{120}$	$\frac{d}{60}$	50	10	EUR	0.42	EUR	833333.33
$\frac{d}{120}$	$\frac{d}{12}$	50	7	EUR	0.29	EUR	583333.33
$\frac{d}{60}$	0	50	10	EUR	0.42	EUR	833333.33
$\frac{d}{60}$	$\frac{d}{60}$	50	9	EUR	0.38	EUR	750000.00
$\frac{d}{60}$	$\frac{d}{12}$	50	6	EUR	0.25	EUR	500000.00
varying	$\frac{d}{60}$	50	9	EUR	0.38	EUR	750000.00
varying	$\frac{d}{12}$	50	6	EUR	0.25	EUR	500000.00
0	$\frac{d}{60}$	10	2	EUR	0.08	EUR	166666.67
0	$\frac{d}{12}$	10	2	EUR	0.08	EUR	166666.67
$\frac{d}{120}$	0	10	3	EUR	0.12	EUR	250000.00
$\frac{d}{120}$	$\frac{d}{60}$	10	2	EUR	0.08	EUR	166666.67
$\frac{d}{120}$	$\frac{d}{12}$	10	2	EUR	0.08	EUR	166666.67
$\frac{d}{60}$	0	10	2	EUR	0.08	EUR	166666.67
$\frac{d}{60}$	$\frac{d}{60}$	10	2	EUR	0.08	EUR	166666.67
$\frac{d}{60}$	$\frac{d}{12}$	10	2	EUR	0.08	EUR	166666.67
varying	$\frac{d}{60}$	10	2	EUR	0.08	EUR	166666.67
varying	$\frac{d}{12}$	10	2	EUR	0.08	EUR	166666.67

Table 6.19: Monthly bandwidth costs for satellite-based Generalized Greedy Broadcasting transmissions using different transmission parameters

6.4.4 Results

As expected, the analysis shows that Point-to-Point transmissions cannot be used efficiently for media-on-demand via satellite. Nevertheless, due to the high costs of satellite bandwidth, regular monthly costs occur which require a high number of customers to be cost-effective.

6.5 Conclusion

In this chapter, two media-on-demand system approaches have been analyzed from a financial point of view: Firstly, a media-on-demand system which is based on Point-to-Point transmissions which is the commonly used type of system today, and secondly, a system based on the General-

ized Greedy Broadcasting scheme which has been proposed and evaluated in the previous chapters as a highly efficient and flexible solution for pro-active media-on-demand services.

This analysis proved for the assumed preconditions that the pro-active approach of the Generalized Greedy Broadcasting scheme evolved to be equivalent to Point-to-Point based systems in medium-sized environments and superior in large environments, in satellite based environments as well as in setups where the receiver systems are connected via DSL to media-on-demand sender systems. This is especially interesting as DSL-based media-on-demand services using Point-to-Point transmission schemes have spread much in recent years.

For small, Ethernet-based environments, Point-to-Point media-on-demand systems are the less expensive alternative because of higher software and hardware costs of Generalized Greedy transmissions. But even in these environments Generalized Greedy based systems are more cost-effective if the media assortment is very small.

Altogether, this chapter evolved that Generalized Greedy Broadcasting transmissions can substitute Point-to-Point transmissions because of lower costs and better scalability in many application cases.

Chapter 7

Conclusion

IN this thesis, media-on-demand systems have been examined and enhanced, both in theory and from practical points of view. To summarize the attained results, the achievements are reviewed briefly in this final chapter: Section 7.1 covers conceptual achievements which have been reached and sections 7.2 and 7.3 detail improvements of efficiency and utilizability. Finally, in the last section of this chapter, the future trend of media-on-demand is described from the author's point of view.

7.1 Conceptual Achievements

To understand the functioning of media-on-demand systems, chapter 2 started with a dissection of media-on-demand systems into functional components. Besides the knowledge gained from the detailed examination as a result of the decomposition itself, this decomposition revealed different aspects of media-on-demand systems which can be improved:

At a first level of dissection, the overall structure of media-on-demand systems can be enhanced. Therefore, the author of this thesis introduced hierarchical media-on-demand systems: In a hierarchical media-on-demand system, several sender systems are organized in a tree-like or mesh-like structure, forwarding media streams downwards the tree hierarchy or between sender systems in a mesh. This eliminates the need to explicitely distribute media streams to all sender systems, the whole setup works like one huge, distributed media-on-demand system. Additionally, the storage requirements of most of the sender systems are lowered because these systems do not have to store all media stream permanently: When a media stream is requested from a sender system which is not in its storage, it can request a media-on-demand transmission for the media stream from another sender system and forward the data to the receiver system.

A deeper look at the structure of media-on-demand systems disclosed that media-on-demand transmissions are driven by a transmission scheme: The transmission scheme defines how media

streams are transmitted from sender to receiver systems. Therefore the author of this thesis presented a classification and detailed analysis of functioning, advantages and drawbacks of a wide range of transmission schemes in chapter 3.

This examination showed that the schemes can be divided into several groups: Reactive transmission schemes, which schedule transmissions based on requests from receiver systems, pro-active transmission schemes which perform permanent transmissions independent of requests from receiver systems and reactive-pro-active hybrids which combine approaches of both variants.

While reactive transmission schemes perform very well if media streams are only requested by few receiver systems, they do not scale for large environments with lots of receiver systems. In contrast, the functioning of pro-active systems is independent of the number of active receiver systems, but as they perform transmissions permanently, they even consume resources if no receiver system is listening to the transmission. Combining these approaches into hybrid systems allows to benefit from both variants, scalability for large environments and economic use of resources for cases with few active receivers.

Besides this grouping of transmission schemes based on the functioning of their scheduling algorithm, the author rated all examined transmission schemes using several technical and functional classifications, e. g. for the supported playback controls (pause, rewind, fast-forward) and the ability to perform live transmissions.

7.2 Efficiency Improvements

Besides the functioning and capabilities of media-on-demand transmission schemes, the resource requirements for media-on-demand are important, esp. the bandwidth requirements of the sender system, the bandwidth requirements of receiver systems and the amount of needed storage at receiver systems.

For the analysis of the sender system bandwidth requirements, the author of this thesis introduced a measure: the efficiency of the media-on-demand transmission scheme. The efficiency of a transmission scheme quantifies how much bandwidth a transmission scheme uses compared to the amount of bandwidth which is absolutely needed from an information theoretic point of view.

One side-effect of this definition for efficiency is that it is independent of the media bit rate, thus the media-on-demand transmission schemes can be rated independent of the media encoding. Nevertheless, as the media bit rate influences the bandwidth requirements of sender and receiver systems, advanced media compression algorithms have been recommended, allowing to lower the bandwidth requirements without a perceivable loss of media quality.

To compare the bandwidth requirements of the presented transmission schemes, the author of this thesis analyzed the efficiency of the transmission schemes which have been presented and classified in chapter 3. Because of the huge number of pro-active transmissions schemes, the

author divided the group of pro-active media-on-demand transmission schemes (including the pro-active part of hybrid schemes) further into four subgroups by the way how they transmit the media stream:

For the first subgroup, the non-segmenting schemes, the media stream is sent linearly as a whole. Schemes of this subgroup have a very low efficiency and are almost only of historic interest.

For schemes of the second subgroup, the size-based transmission schemes, the media stream is split into segments of increasing size which are transmitted round-robin on separate channels of equal bandwidth. This subgroup showed much better results, but as the analysis of the author revealed, they cannot perform as good as the schemes of the remaining two subgroups.

The third subgroup is of special interest because it contains the transmission schemes with the highest efficiency, even schemes which operate at an efficiency of 100 %: The bandwidth-based transmission schemes of this subgroup perform media-on-demand transmissions by splitting the media stream into segments of equal size and transmitting the segments on channels with decreasing bandwidth. Unfortunately, the schemes of this subgroup have several fundamental drawbacks which render them mostly unusable in a practical system.

Transmission schemes of the last subgroup, the frequency-based schemes, split the media-stream into segments of equal size, too, but transmit these segments on channels of equal bandwidth at decreasing frequency. Determining the period and phase shift for each segment in such a way that they can be sent on the lowest number of channels is one of the major challanges of the frequency-based transmission schemes: As it has been shown that finding the optimal solution is NP-hard, simple static mappings and heuristic algorithms of different type are used for frequency-based schemes. Unfortunately, the author found no transmission scheme of the frequency-based subgroup, which has an efficiency nearly as high as the bandwidth-based schemes.

As a result of the gained insights from the efficiency comparison and because of the lack of a transmision scheme which provides a high efficiency and does not suffer from the problems of the bandwidth-based subgroup, the author of this thesis proposed a new transmission scheme in chapter 4, the Generalized Greedy Broadcasting scheme: Using one of the frequency-based transmission schemes as a starting point, the author generalized it and enhanced its heuristic, thereby reaching an efficiency of more than 99 %.

In addition to reaching a high efficiency, the author of this thesis applied several mechanisms to the Generalized Greedy Broadcasting scheme which allow to lower the sender system bandwidth requirements independent of the efficiency (e. g. preloading the beginning of media streams to receiver systems or utilizing planned breaks in the stream transmission to lower the transmission bandwidth).

Last but not least, the author also extended the Generalized Greedy Broadcasting scheme in such a way that it is able to cope with limitations of receiver systems: For receiver systems with

limited bandwidth or storage, the algorithm has been adjusted to produce a schedule which respects these limitations. Although this increases the bandwidth requirements of the sender system, the Generalized Greedy Broadcasting scheme still operates at the same high efficiency if the receiver system limits are considered in the efficiency calculation.

7.3 Extended Utilizability

Apart from a high efficiency, the utilizability of media-on-demand in general and of the proposed Generalized Greedy Broadcasting scheme in particular has been extended throughout this thesis in several directions.

Therefore, the author of this thesis first identified and analyzed the extensions which have been presented for the transmission schemes in chapter 3. As it emerged, many transmission schemes have been extended in some way to fit a special purpose, but no transmission scheme has been found which supported many of the extensions at once.

As a second step, the author of this thesis extended this list by additional requirements of consumers and providers to improve the utilizability of enhanced media-on-demand systems in many use-cases. Altogether, this complete list included mechanisms to enhance the service for the consumer (e. g. support for immediate playback for pro-active transmission schemes), to extend its applicability (e. g. for live transmissions or variable-bit-rate media streams) and to support efficient use in changing environments (e. g. replacement of media streams or change of media stream transmission parameters in a running system). Due to the preliminary work on efficiency analysis presented in chapter 3, the author was able to perform a detailed study of the impacts to efficiency of each of these extensions.

Next, the author incorporated most of the found extensions into the Generalized Greedy Broadcasting scheme in chapter 4. Thereby, the flexibility and customizability of the Generalized Greedy Broadcasting scheme helped to integrate them in such a way that the efficiency of the Generalized Greedy Broadcasting scheme stayed as high as ever.

Furthermore, many transport level improvements for media-on-demand systems have been proposed in chapter 5 which can be applied to further increase the utilizability. For this purpose, the author examined the availability and usability of different transport enhancement techniques like error correction, layered media streams and security for media-on-demand. Using the standardized, widely-used Real-time Transport Protocol (RTP) together with the Session Description Protocol (SDP) for enhanced media-on-demand transmissions and suggesting a solution for the migration from existing media-on-demand systems to enhanced ones took another important part in this chapter.

Finally, the author provided a comparison of the transmission technique specific costs in chapter 6, comparing the utilizability of the Generalized Greedy Broadcasting scheme with traditional

point-to-point transmissions from a financial point of view. This comparison revealed that the Generalized Greedy Broadcasting scheme is comparable with point-to-point transmission based systems in the examined medium-sized use-case and superior for the large setup.

7.4 Future of Media-on-Demand

Some years ago, when the author started his research on media-on-demand, it was obvious that media-on-demand was not cost-effective yet: The bandwidth requirements were very high, high bandwidth network access was very expensive, the hardware requirements for sender and receiver systems called for huge expenses and the distribution rights for media streams were unclear.

Nevertheless, it was predictable that this would change in near future. And as of today, the state of affairs has changed: The bandwidth requirements have been lowered, many households have a broadband network connection which can be used for media-on-demand services, hardware prices have fallen steadily and content providers started to agree into distribution licenses for on-demand services.

As a result, many major companies have entered the media-on-demand business. But unfortunately, most of the used systems are based on point-to-point transmissions. Besides missing publicity of enhanced media-on-demand transmission techniques, one of the reasons may be that the enhanced techniques bear more risks than the simple systems for providers: As the receiver systems for enhanced media-on-demand systems are more expensive (as they must be equipped with storage), enhanced transmissions are only profitable if the lower sender system costs compensate for this. This gives point-to-point based systems an advantage in the beginning phase where the sender system costs are roughly proportional to the number of customers for point-to-point based systems but proportional to the number of offered media titles for enhanced systems.

But although point-to-point transmission based systems may satisfy the current demands and are less expensive for small systems, it is risky not to investigate into new media-on-demand techniques: Many market analyses project an increase in media-on-demand in the next years and it may even be possible that media-on-demand becomes an established service besides traditional media broadcasting in near future.

The more this state is approached, the more importance has to be attached to advanced, proactive media-on-demand techniques: Highly scalable solutions which can serve any number of recipients gain an advancement to any recipient-based system, flexible and generic solutions will displace any special solution for a single use-case, well-examined and standard-based solutions will supersede experimental and proprietary approaches.

In summary, the author observed the wide range of media-on-demand approaches from the

analytic side in this thesis and evolved a new transmission scheme and further enhancements which can make media-on-demand systems more efficient, more utilizable and thus more cost-effective than the systems of current use for many application cases. Against this background, the achievements of this thesis provide a major advancement for a successful future of media-on-demand services.

Appendix A

Algorithms

THIS appendix provides a selection of the most important algorithms which have been proposed in this thesis. They are presented in a pseudo language, running C versions of these algorithms and many further ones can be retrieved from the author on request.

A.1 Tree-to-Tabular Representation Converter

```
0001: // type for nodes in tree-representation of transmission trees
0002: type Node := record
0003:     segment_no: Integer;        // segment of this node, 0 if none
0004:     children: List of Node;     // child nodes
0005: endrecord;
0006:
0007: // type for a segment of the transmission scheme
0008: type Segment := record
0009:     segment_no: Integer;        // segment number
0010:     period: Integer;            // period of the segment
0011:     phase_shift: Integer;       // phase shift of the segment
0012:     channel_no: Integer;        // channel number of the segment in the transmission scheme
0013: endrecord;
0014:
0015: // extract segments of a tree
0016: // parameters:
0017: //      node                    the node to examine
0018: //      period                  the period of the node
0019: //      phase_shift             the phase shift of the node
0020: //      tree_no                 the number of the tree the node is part of
0021: //      table                   the tabular representation where the extracted segments are added to
0022: procedure extract_segments(
0023:     in node: Node,
0024:     in period: Integer,
0025:     in phase_shift: Integer,
0026:     in tree_no: Integer,
0027:     inout table: List of Segment)
0028: var child_no: Integer;
0029: begin
0030:
0031:     // check if this is a leaf node with a segment attached to it
0032:     if node.segment_no ≠ 0 then
0033:
0034:         // save the found segment to the table
0035:         append new Segment(segment_no ← node.segment_no, period ← period, phase_shift ← phase_shift,
0036:             channel_no ← tree_no) to table;
0037:
0038:     // otherwise it is either an unused node or a non-leaf node
0039:     else
0040:
0041:         // examine all child nodes of the node
0042:         for child_no ← 0 to length(node.children) - 1 do
0043:             extract_segments(node.children[child_no], period · length(node.children),
0044:                 phase_shift + child_no · period, tree_no, table);
0045:         endfor;
0046:     endif;
0047: end;
0048:
0049: // convert a transmission schedule form tree to tabular representation
0050: // parameters:
0051: //      trees                   the created tree representation
0052: //      table                   the tabular representation
0053: procedure trees_to_tabular(
0054:     in trees: List of Node,
0055:     out table: List of Segment)
0056: var tree_no: Integer;
0057: begin
0058:
0059:     // extract the segments of each tree, one by one
0060:     table ← [];
0061:     for tree_no ← 0 to length(trees) - 1 do
0062:         extract_segments(trees[tree_no], 1, 0, tree_no, table);
0063:     endfor;
```

```
0064: end;
```

A.2 Tabular-to-Tree Representation Converter

```
0001: // type for nodes in tree-representation of transmission trees
0002: type Node := record
0003:     period: Integer;            // period of node
0004:     phase_shift: Integer;       // phase shift of node
0005:     tree_no: Integer;           // number of tree this node is part of
0006:     segment_no: Integer;        // segment of this node, 0 if none
0007:     children: List of Node;     // child nodes
0008: endrecord;
0009:
0010: // type for a segment of the transmission scheme
0011: type Segment := record
0012:     segment_no: Integer;        // segment number
0013:     period: Integer;            // period of the segment
0014:     phase_shift: Integer;       // phase shift of the segment
0015:     channel_no: Integer;        // channel number of the segment in the transmission scheme
0016: endrecord;
0017:
0018: // insert a segment into the trees
0019: // parameters:
0020: //      segment                 the segment to insert
0021: //      table                   the tabular representation
0022: //      trees                   the trees where the segment is inserted to
0023: procedure insert_segment(
0024:     in segment: Segment,
0025:     in table: List of Segment,
0026:     inout trees: List of Node)
0027: var periods: List of Integer;
0028:     period: Integer;
0029:     split: Integer;
0030:     child_no: Integer;
0031: begin
0032:     // get the root node of the tree where the segment is to be added to
0033:     node ← trees[segment.tree_no - 1];
0034:
0035:     // step down the tree until the target node has been created
0036:     while node.period ≠ segment.period do
0037:
0038:         // check if the node has to be split
0039:         if node.children = [] then
0040:
0041:             // get the periods of all child nodes of the current node
0042:             periods ← select ⋆.period from table where
0043:                 ⋆.channel_no ← node.tree_no ∧ ⋆.phase_shift % node.period = node.phase_shift;
0044:
0045:             // to ensure that all segments can be inserted to this node, we have to find a split which
0046:             // all child nodes have in common; this can be found using the greatest common divisor
0047:             period ← gcd(periods);
0048:
0049:             // we just perform prime factor splits, so get the first prime factor for the subtree periods
0050:             split ← first_prime_factor(period ÷ node.period);
0051:
0052:             // add the child nodes
0053:             for child_no := 0 to split - 1 do
0054:                 child ← new Node(segment_no ← 0, period ← node.period · split,
0055:                     phase_shift ← node.phase_shift + child_no · node.period, tree_no ← node.tree_no,
0056:                     children ← []);
0057:                 append child to node.children;
0058:             endfor;
0059:         fi;
0060:
0061:         // step down the tree
0062:         node ← node.children[segment.phase_shift ÷ node.period % length(node.children)];
0063:     endwhile;
0064:
0065:     // assign the segment to the node
0066:     node.segment_no ← segment.segment_no;
0067: end;
0068:
0069: // convert a transmission schedule form tabular to tree representation
0070: // parameters:
0071: //      table                   the tabular representation
0072: //      trees                   the created tree representation
0073: procedure tabular_to_trees(
0074:     in table: List of Segment,
0075:     out trees: List of Node)
0076: var no_trees: Integer;
0077:     tree_no: Integer;
0078:     segment: Segment;
0079: begin
0080:
0081:     // get the number of trees
0082:     no_trees ← select max(⋆.tree_no) from table;
0083:
0084:     // add root nodes to the trees
0085:     trees ← [];
0086:     for tree_no := 1 to no_trees do
0087:         node ← new Node(segment_no ← 0, period ← 1, phase_shift ← 0, tree_no ← tree_no, children ← []);
0088:         append node to trees;
0089:     endfor;
0090:
0091:     // insert all segments in the trees
0092:     foreach segment in table do
0093:         insert_segment(segment, table, trees);
0094:     endforeach;
0095: end;
```

A.3 Generalized Greedy Scheduler

```
0001: // type for nodes in tree-representation of transmission trees
0002: type Node := record
0003:     period: Integer;              // period of node
0004:     phase_shift: Integer;         // phase shift of node
0005:     tree_no: Integer;             // number of tree this node is part of
0006:     segment_no: Integer;          // segment of this node, 0 if none
0007:     children: List of Node;       // child nodes
0008: endrecord;
0009:
0010: // type for a segment of the transmission scheme
0011: type Segment := record
0012:     segment_no: Integer;          // segment number
0013:     required_period: Integer;     // required transmission period of the segment
0014: endrecord;
0015:
0016: // find the best split; this function remembers that there is a prime split restriction and decreases the
0017: // split to apply accordingly
0018: // parameters:
0019: //      required_period          the required period of the segment
0020: //      node_period              the period of the found node
0021: //      split_restriction        prime factor split restriction to apply when performing splits
0022: //      best_split               the calculated split to apply
0023: procedure find_best_split(
0024:     in required_period: Integer,
0025:     in node_period: Integer,
0026:     in split_restriction: Integer,
0027:     out best_split: Integer)
0028: var restriction_violated: Boolean;
0029:     current_node_period: Integer;
0030:     remaining_split: Integer;
0031: begin
0032:
0033:     // try as long until we found a split which does not violate the split restriction
0034:     restriction_violated ← true;
0035:     while restriction_violated do
0036:
0037:         // check how much splitting is required
0038:         current_node_period ← node_period;
0039:         remaining_split ← required_period ÷ node_period;
0040:
0041:         // perform one split after another as long as the split restriction is not violated
0042:         restriction_violated ← false;
0043:         while remaining_split > 1 ∧ ¬restriction_violated do
0044:
0045:             // get the next prime split to perform
0046:             next_split ← smallest_prime_factor(remaining_split);
0047:
0048:             // check if this split would violate the prime factor split restriction
0049:             restriction_violated ← (next_split ≥ split_restriction · current_node_period + 1);
0050:
0051:             // calculate the remaining splits
0052:             remaining_split ← remaining_split ÷ next_split;
0053:             current_node_period ← current_node_period · next_split;
0054:         endwhile;
0055:
0056:         // if this split failed, decrease required_period and try again
0057:         if restriction_violated then
0058:             required_period ← required_period - 1;
0059:         endif;
0060:     endwhile;
0061:     best_split ← required_period ÷ node_period;
0062: end;
0063:
0064: // find the best node where a segment should be inserted
0065: // parameters:
0066: //      segment                  the segment to insert
0067: //      available_nodes          available nodes where the segment can be inserted
0068: //      weight                   weight value to use for selecting the node when several nodes qualify
0069: //      split_restriction        prime factor split restriction to apply when performing splits
0070: //      best_node                the best found node if any
0071: //      best_split               the split to apply to this node
0072: procedure find_best_node(
0073:     in segment: Segment,
0074:     in available_nodes: List of Node,
0075:     in weight: Real,
0076:     in split_restriction: Integer,
0077:     out best_node: Node,
0078:     out best_split: Integer)
0079: var node: Node;
0080:     split: Integer;
0081:     quality1, quality2, quality, best_quality: Real;
0082: begin
0083:
0084:     // examine all available nodes
0085:     best_node ← ε;
0086:     best_split ← 0;
0087:     foreach node in available_nodes do
0088:
0089:         // check if the node has a sufficiently low period
0090:         if node.period ≤ segment.required_period then
0091:
0092:             // get the split we have to apply to this node
0093:             find_best_split(segment.required_period, node.period, split_restriction, split);
0094:
0095:             // calculate the quality of this node
0096:             quality1 ← 1 - (segment.required_period - node.period · split)   segment.required_period;
0097:             quality2 ← node.period   segment.required_period;
0098:             quality ← weight · quality1 + (1 - weight) · quality2;
0099:
0100:             // if this node is better than a previously found one, save it
0101:             if best_node = ε ∨ quality > best_quality then
0102:                 best_node ← node;
0103:                 best_quality ← quality;
0104:                 best_split ← split;
0105:             endif;
0106:         endif;
0107:     endforeach;
0108: end;
0109:
0110: // try to insert a segment into the transmission trees
0111: // parameters:
0112: //      segment                  the segment to insert
0113: //      available_nodes          available nodes where the segment can be inserted
0114: //      weight                   weight value to use for selecting the node when several nodes qualify
```

```
0115: //      split_restriction       prime factor split restriction to apply when performing splits
0116: //      node                    the resulting node if successful
0117: procedure insert_node(
0118:     in segment: Segment,
0119:     inout available_nodes: List of Node,
0120:     in weight: Real,
0121:     in split_restriction: Integer,
0122:     out node: Node)
0123: var split: Integer;
0124:     child_no: Integer;
0125: begin
0126:     // find the best node where the segment is to be appended to
0127:     find_best_node(segment, available_nodes, weight, split_restriction, node, split);
0128:
0129:     // successful, then insert the segment
0130:     if node ≠ ε then
0131:
0132:         // check if splits have to be performed
0133:         while split > 1 do
0134:
0135:             // get the next split to perform
0136:             next_split ← smallest_prime_factor(split);
0137:
0138:             // perform the split
0139:             remove node from available_nodes;
0140:             for child_no ← 0 to next_split - 1 do
0141:                 child ← new Node(period ← node.period · split,
0142:                     phase_shift ← node.phase_shift + child_no · node.period, tree_no ← node.tree_no,
0143:                     segment_no ← 0, children ← []);
0144:                 append child to node.children;
0145:                 append child to available_nodes;
0146:             endfor;
0147:
0148:             // remember that we split the node
0149:             split ← split ÷ next_split;
0150:
0151:             // continue with the child node
0152:             node ← node.children[0];
0153:         endwhile;
0154:
0155:         // assign the segment to the node
0156:         remove node from available_nodes;
0157:         node.segment_no ← segment.segment_no;
0158:     endif;
0159: end;
0160:
0161: // create a transmission scheme for the Generalized Greedy Broadcasting scheme for a single set of parameters
0162: // parameters:
0163: //      segments                list of segment parameters for the transmission schedule to create
0164: //      weight                  weight parameter to use
0165: //      split_restriction       prime split restriction to use
0166: //      trees                   the resulting schedule in tree representation
0167: //      used_bandwidth          the average used bandwidth of the schedule
0168: procedure create_schedule_with_parameters(
0169:     in segments: List of Segment,
0170:     in weight: Real,
0171:     in split_restriction: Integer,
0172:     out trees: List of Node,
0173:     out used_bandwidth: Real)
0174: var no_trees: Integer;
0175:     succeeded: Boolean;
0176:     available_nodes: List of Node;
0177:     tree_no: Integer;
0178:     node: Node;
0179: begin
0180:
0181:     // we want to use as few channels as possible, so start with a single tree and increment this if
0182:     // it is not possible to create a schedule
0183:     no_trees ← 1;
0184:     succeeded ← false;
0185:     while ¬succeeded do
0186:
0187:         // initialize for creation
0188:         available_nodes ← [];
0189:         used_bandwidth ← 0;
0190:
0191:         // create the root nodes
0192:         trees ← [];
0193:         for tree_no := 1 to no_trees do
0194:             node ← new Node(period ← 1, phase_shift ← 0, tree_no ← tree_no, segment_no ← 0,
0195:                 children ← []);
0196:             append node to trees;
0197:             append node to available_nodes;
0198:         endfor;
0199:
0200:         // insert one segment after another into the trees
0201:         succeeded ← true;
0202:         foreach segment in segments do
0203:
0204:             // try to insert the segment into the trees
0205:             insert_node(segment, available_nodes, weight, split_restriction, node);
0206:
0207:             // failed, then the schedule is full
0208:             if node = ε then
0209:
0210:                 // we have to use one more channel; abort the loop
0211:                 succeeded ← false;
0212:                 break;
0213:
0214:             // the segment fit into the schedule
0215:             else
0216:
0217:                 // update the used bandwidth
0218:                 used_bandwidth ← used_bandwidth + 1   node.period;
0219:             endif;
0220:         endforeach;
0221:         if ¬succeeded then
0222:             no_trees ← no_trees + 1;
0223:         endif;
0224:     endwhile;
0225: end;
0226:
0227: // create a transmission scheme for the Generalized Greedy Broadcasting scheme
0228: // parameters:
0229: //      segments                list of segment parameters for the transmission schedule to create
0230: //      weights                 list of weight parameters to try; the best resulting schedule is returned
0231: //      split_restrictions      list of prime split restrictions to try; the best resulting schedule is
```

```
0232: //                            returned
0233: //      trees                 the resulting schedule in tree representation
0234: //      used_bandwidth        the average used bandwidth of the schedule
0235: procedure create_schedule(
0236:     in segments: List of Segment,
0237:     in weights: List of Real,
0238:     in split_restrictions: List of Integer,
0239:     out trees: List of Node,
0240:     out used_bandwidth: Real)
0241: var weight: Real;
0242:     split_restriction: Integer;
0243:     current_trees: List of Node;
0244:     current_used_bandwidth: Real;
0245: begin
0246:
0247:     // sort the segments by their required period
0248:     sort segments ascending by required_period;
0249:
0250:     // try all weights and prime split restrictions and return the schedule with the least average bandwidth
0251:     used_bandwidth ← ∞;
0252:     foreach weight in weights do
0253:         foreach split_restriction in split_restrictions do
0254:
0255:             // create a schedule for these parameters
0256:             create_schedule_with_parameters(segments, weight, split_restriction,
0257:                 current_trees, current_used_bandwidth);
0258:
0259:             // if this schedule is better than previous ones, save it
0260:             if current_used_bandwidth < used_bandwidth then
0261:                 trees ← current_trees;
0262:                 used_bandwidth ← current_used_bandwidth;
0263:             endif;
0264:         endforeach;
0265:     endforeach;
0266: end;
```

A.4 Enhanced Startup

```
0001: // type for nodes in tree-representation of transmission trees
0002: type Node := record
0003:     period: Integer;              // period of node
0004:     phase_shift: Integer;         // phase shift of node
0005:     tree_no: Integer;             // number of tree this node is part of
0006:     segment_no: Integer;          // segment of this node, 0 if none
0007:     children: List of Node;       // child nodes
0008: endrecord;
0009:
0010: // type for extracted information from transmission trees
0011: type Info := record
0012:     period: Integer;              // period of a found node
0013:     no_children: Integer;         // number of child nodes of the node
0014: endrecord;
0015:
0016: // type for a segment of the transmission scheme
0017: type Segment := record
0018:     segment_no: Integer;          // segment number
0019:     required_period: Integer;     // required transmission period of the segment
0020: endrecord;
0021:
0022: // type for found positions in transmission trees where a segment can be inserted
0023: type Position := record
0024:     tree_no: Integer;             // tree number of the position
0025:     period: Integer;              // period of the position
0026:     phase_shift: Integer;         // phase shift of the position
0027:     splits: List of Integer;      // splits which have to be performed
0028: endrecord;
0029:
0030: // extract the structure of transmission trees
0031: // parameters:
0032: //      node                  the node to examine
0033: //      infos                 list where to append the extracted information
0034: procedure extract_structure_of_tree(
0035:     in node: Node,
0036:     inout infos: List of Info)
0037: var child_node: Node;
0038: begin
0039:
0040:     // add the node itself
0041:     append new Info(period ← node.period, no_children ← length(node.children)) to infos;
0042:
0043:     // add all child nodes of the node recursively
0044:     foreach child_node in node.children do
0045:         extract_structure_of_tree(child_node, infos);
0046:     endforeach;
0047: end;
0048:
0049: // find the best position where a segment can be reinserted into the transmission trees when rebuilding the
0050: // trees from the extracted information
0051: // parameters:
0052: //      required_period       required (i. e. maximum) period of the segment to insert
0053: //      min_period            minimum period to use; needed to prevent that too many nodes with low period
0054: //                            are consumed and insertion fails in the second round
0055: //      nodes                 list of node to examine for insertion or appending subtrees
0056: //      infos                 list of remaining information from tree structure extraction
0057: //      position              the currently best position, is updated when a better one is found
0058: procedure find_best_position(
0059:     in required_period: Integer,
0060:     in min_period: Integer,
0061:     in nodes: List of Node,
0062:     in infos: List of Info,
0063:     inout position: Position)
0064: var node: Node;
0065: begin
0066:
0067:     // check all available nodes
0068:     foreach node in nodes do
0069:
0070:         // check if the node has a sufficiently low period and is not yet used
```

```
0071:         if node.period ≤ required_period ∧ node.segment_no = 0 then
0072:
0073:             // check if no child nodes have been added yet, i. e. if we can add a subtree
0074:             if node.children = [] then
0075:
0076:                 // search the best position by trying different subtree elements
0077:                 find_best_position_with_reconstruction(required_period, min_period, node.tree_no,
0078:                     node.period, node.phase_shift, infos, [], position);
0079:
0080:             // otherwise child nodes have already been added, so look in them
0081:             else
0082:
0083:                 // look for good positions in the child nodes
0084:                 find_best_position(required_period, min_period, node.children, infos, position);
0085:             endif;
0086:         endif;
0087:     endforeach;
0088: end;
0089:
0090: // find the best position where a segment can be reinserted into the transmission trees when rebuilding the
0091: // trees from the extracted information
0092: // parameters:
0093: //      required_period         required (i. e. maximum) period of the segment to insert
0094: //      min_period              minimum period to use; needed to prevent that too many nodes with low period
0095: //                              are consumed and insertion fails in the second round
0096: //      tree_no                 the tree number of the current node where information pieces can be added
0097: //      period                  the period of the current node where information pieces can be added
0098: //      phase_shift             the phase shift of the current node where information pieces can be added
0099: //      infos                   list of remaining information from tree structure extraction
0100: //      splits                  list of splits which have already been performed to create this node
0101: //      position                the currently best position, is updated when a better one is found
0102: procedure find_best_position_with_reconstruction(
0103:     in required_period: Integer,
0104:     in min_period: Integer,
0105:     in tree_no: Integer,
0106:     in period: Integer,
0107:     in phase_shift: Integer,
0108:     in infos: List of Info,
0109:     in splits: List of Integer,
0110:     inout position: Position)
0111: var info: Info;
0112:     child_no: Integer;
0113: begin
0114:
0115:     // examine all information pieces if one matches the period of the node
0116:     foreach info in infos do
0117:         if info.period = period then
0118:
0119:             // if we have found a leaf node information piece and the period is high enough, we can try to
0120:             // add the segment here
0121:             if info.no_children = 0 ∧ period ≥ min_period then
0122:
0123:                 // check if the gained position is better than the previously found one
0124:                 if position = ε ∨
0125:                     (position.period + position.phase_shift ≥ required_period ∧
0126:                      period + phase_shift < required_period) ∨
0127:                     (period + phase_shift > position.period + position.phase_shift ∧
0128:                      position.period + position.phase_shift ≥ required_period) ∨
0129:                     (position.period + position.phase_shift < period + phase_shift ∧
0130:                      period + phase_shift < required_period) then
0131:
0132:                     // update the currently best position with the found one
0133:                     position ← new Position(tree_no ← tree_no, period ← period, phase_shift ← phase_shift,
0134:                         splits ← splits);
0135:                 endif;
0136:
0137:             // otherwise the information piece describes a split
0138:             else
0139:
0140:                 // check if this split creates nodes with low enough period
0141:                 if period · info.no_children ≤ required_period then
0142:
0143:                     // examine all resulting child nodes
0144:                     for child_no ← 0 to info.no_children - 1 do
0145:                         find_best_position_with_reconstruction(required_period, min_period, tree_no,
0146:                             period · info.no_children, phase_shift + child_no · period, infos,
0147:                             [splits, info.no_children], position);
0148:                     endfor;
0149:                 endif;
0150:             endif;
0151:         endif;
0152:     endforeach;
0153: end;
0154:
0155: // reconstruct a node in the tree at a specified position and attach a segment to it
0156: // parameters:
0157: //      segment_no              the segment to attach to the resulting node
0158: //      phase_shift             the phase shift of the node
0159: //      splits                  splits to perform to find the searched node
0160: //      node                    the (root) node where to reconstruct the node
0161: //      infos                   list of information pieces, used pieces are removed
0162: procedure reconstruct_node_in_tree(
0163:     in segment_no: Segment,
0164:     in phase_shift: Integer,
0165:     in splits: List of Integer,
0166:     in node: Node,
0167:     inout infos: List of Info)
0168: var info: Info;
0169:     child_no: Integer;
0170: begin
0171:
0172:     // skip the parts which already have been reconstructed
0173:     while node.children ≠ [] do
0174:         node ← node.children[phase_shift ÷ node.period % length(node.children)];
0175:     endwhile;
0176:
0177:     // now perform the given splits
0178:     foreach split in splits do
0179:
0180:         // remove the information piece which describes the current split from the list of available pieces
0181:         remove once from infos where infos[*].period = period ∧ infos[*].no_children = split;
0182:
0183:         // create child nodes for the node to split
0184:         for child_no ← 0 to split - 1 do
0185:             child ← new Node(period ← node.period · split,
0186:                 phase_shift ← node.phase_shift + child_no · node.period, tree_no ← node.tree_no,
0187:                 segment_no ← 0, children ← []);
```

```
0188:             append child to node.children;
0189:         endfor;
0190:
0191:         // continue with the right child node
0192:         node ← node.children[phase_shift ÷ node.period % split];
0193:     endforeach;
0194:
0195:     // remove the information piece which describes the current leaf node
0196:     remove once from infos where infos[*].period = period ∧ infos[*].no_children = 0;
0197:
0198:     // attach the segment to the node
0199:     node.segment_no ← segment_no;
0200: end;
0201:
0202: // get the minimum period when inserting a segment; this is necessary to prevent that all nodes with low
0203: // periods are used up in the first round, producing a failure in the second round
0204: // parameters:
0205: //      segments                list of segments which still have to be inserted
0206: //      infos                   list of remaining information pieces from key structure extraction
0207: //      index                   index of the segment to insert in the segments list
0208: //      min_period              the resulting minimum period to apply
0209: procedure get_min_period(
0210:     in segments: List of Segment,
0211:     in infos: List of infos,
0212:     in index: Integer,
0213:     out min_period: Integer)
0214: begin
0215:
0216:     // examine the segments and infos in reverse order
0217:     while index > 0 ∧ segments[index - 1].required_period ≥ infos[index].period do
0218:         index ← index - 1;
0219:     endwhile;
0220:
0221:     // the period of the information piece which cannot be inserted when the first information piece is
0222:     // removed describes the lower bound for the period
0223:     min_period ← infos[index].period;
0224: end;
0225:
0226: // startup enhancement main function; rebuilds transmission trees in such a way that several segment
0227: // transmissions can be omitted
0228: // parameters:
0229: //      segments                list of segments of this transmission schedule
0230: //      trees                   transmission schedule to optimize, will be replaced with the result
0231: procedure enhance_startup(
0232:     in segments: List of Segment,
0233:     inout trees: List of Node)
0234: var infos: List of Info;
0235:     segment: Segment;
0236:     no_trees: Integer;
0237:     tree_no: Integer;
0238:     position: Position;
0239:     int min_period: Integer;
0240:     int index: Integer;
0241: begin
0242:
0243:     // extract the information structure of the transmission schedule
0244:     infos ← [];
0245:     no_trees ← length(trees);
0246:     foreach tree in trees do
0247:         extract_structure_of_tree(tree, infos);
0248:     endforeach;
0249:
0250:     // rebuild the root trees
0251:     trees ← [];
0252:     for tree_no ← 0 to no_trees - 1 do
0253:         append new Node(period ← 1, phase_shift ← 0, tree_no ← tree_no, segment_no ← 0, children ← []) to
0254:             trees;
0255:     endfor;
0256:
0257:     // sort the segments and information pieces so they can simply be traversed
0258:     sort segments ascending by required_period;
0259:     sort infos ascending by no_children and ascending by period;
0260:
0261:     // first round: insert only segments if they can be omitted at the first transmission
0262:     index ← 0;
0263:     while index < length(segments) do
0264:
0265:         // get the segment to insert and its minimum period
0266:         segment ← segments[index];
0267:         get_min_period(segments, infos, index, min_period);
0268:
0269:         // find the best position where to insert it into the transmission trees
0270:         position ← ε;
0271:         find_best_position(segment.required_period, min_period, trees, infos, position);
0272:
0273:         // check if this position allows to omit the segment transmission once
0274:         if position.period + position.phase_shift < segment.required_period then
0275:
0276:             // then reconstruct the corresponding nodes and insert the segment into the trees
0277:             reconstruct_node_in_tree(segment.segment_no, position.phase_shift, position.splits,
0278:                 trees[position.tree_no], infos);
0279:
0280:             // remove the segment from the list of segments to insert
0281:             remove segment from segments;
0282:
0283:         // otherwise advance to the next segment
0284:         else
0285:             index ← index + 1;
0286:         endif;
0287:     endwhile;
0288:
0289:     // round two: insert all remaining segments
0290:     while segments ≠ [] do
0291:
0292:         // get the segment to insert
0293:         segment ← segments[0];
0294:
0295:         // find the best position where to insert it into the transmission trees
0296:         position ← ε;
0297:         find_best_position(segment.required_period, 1, trees, infos, position);
0298:
0299:         // reconstruct the corresponding nodes and insert the segment into the trees
0300:         reconstruct_node_in_tree(segment.segment_no, position.phase_shift, position.splits,
0301:             trees[position.tree_no], infos);
0302:
0303:         // remove the segment from the list of segments to insert
0304:         remove segment from segments;
```

```
0305:     endwhile;
0306: end;
```

A.5 Brute-Force Storage Requirement Calculation

```
0001: // type for a segment of the transmission scheme
0002: type Segment := record
0003:     segment_no: Integer;         // segment number
0004:     required_period: Integer;    // required transmission period of the segment
0005:     period: Integer;             // utilized period in the transmission scheme
0006:     phase_shift: Integer;        // phase shift of the segment in the transmission scheme
0007:     channel_no: Integer;         // channel number of the segment in the transmission scheme
0008: endrecord;
0009:
0010: // type for specification of segment reception times
0011: type Spec := record
0012:     segment_no: Integer;         // segment number which is described
0013:     first: Integer;              // the first recipient (in slot intervals after media startup) which
0014:                                  // received this segment
0015:     number: Integer;             // the number of slot intervals this segment is received by receivers
0016:     period: Integer;             // period of the segment
0017: endrecord;
0018:
0019: // calculate storage requirements of a transmission scheme using the brute force approach
0020: // parameters:
0021: //      segments                 the segments to examine
0022: //      t                        the playback time (in slot intervals) to calculate the storage requirements
0023: //                               for
0024: //      storage_requirement      the calculated storage requirement (in segments)
0025: procedure calculate_storage_requirement(
0026:     in segments: List of Segment,
0027:     in t: Integer,
0028:     out storage_requirement: Integer)
0029: var specs: List of Spec;
0030:     spec: Spec;
0031:     count: Integer;
0032:     s: Integer;
0033: begin
0034:
0035:     // calculate which segments are received under which circumstances (i. e. which join time); therefore
0036:     // examine all segments
0037:     specs ← [];
0038:     foreach segment in segments do
0039:
0040:         // check if the segment has not yet been played
0041:         if t ≤ segment.needed_period - 1 then
0042:
0043:             // calculate the join times of the receiver which received the segment first and the number
0044:             // of slot intervals of receivers which received the segment, too
0045:             first ← (segment.phase_shift - t) % segment.period;
0046:             number ← min(t + 1, segment.period, segment.period + t - segment.needed_period + 1);
0047:
0048:             // if the segment is received (and will not be received again) by any receiver, save this
0049:             if number > 0 then
0050:                 append new Spec(segment_no ← segment.segment_no, first ← first, number ← number,
0051:                     period ← segment.period) to specs;
0052:             endif;
0053:         endif;
0054:     endforeach;
0055:
0056:     // calculate the period of all specs we found
0057:     count ← 1;
0058:     foreach spec in specs do
0059:         count ← lcm(count, spec.period);
0060:     endforeach;
0061:
0062:     // examine the storage requirements for all possible receivers
0063:     storage_requirement ← 0;
0064:     for s ← 0 to count - 1 do
0065:
0066:         // calculate the storage requirements of a receiver which joined s slot intervals after media
0067:         // startup
0068:         new_storage_requirement ← 0;
0069:         foreach spec in specs do
0070:             if spec.number ≥ spec.period ∨ (s - spec.first) % spec.period < spec.number then
0071:                 new_storage_requirement ← new_storage_requirement + 1;
0072:             endif;
0073:         endforeach;
0074:
0075:         // if this result is higher than anything before, save it
0076:         if new_storage_requirement > storage_requirement then
0077:             storage_requirement ← new_storage_requirement;
0078:         endif;
0079:     endfor;
0080: end;
```

A.6 Exact Storage Requirement Calculation

```
0001: // type for a segment of the transmission scheme
0002: type Segment := record
0003:     segment_no: Integer;         // segment number
0004:     required_period: Integer;    // required transmission period of the segment
0005:     period: Integer;             // utilized period in the transmission scheme
0006:     phase_shift: Integer;        // phase shift of the segment in the transmission scheme
0007:     channel_no: Integer;         // channel number of the segment in the transmission scheme
0008: endrecord;
0009:
0010: // type for specification of segment reception times
0011: type Spec := record
```

```
0012:     segment_no: Integer;        // segment number which is described
0013:     first: Integer;             // the first recipient (in slot intervals after media startup) which
0014:                                 // received this segment
0015:     number: Integer;            // the number of slot intervals this segment is received by receivers
0016:     period: Integer;            // period of the segment
0017: endrecord;
0018:
0019: // a factor for prime factor splitting of periods
0020: type Factor := record
0021:     base: Integer;              // the base of the factor
0022:     exponent: Integer;          // the exponent of the factor
0023: endrecord;
0024:
0025: // iteration value when examining several possible values for a prime factor
0026: type Defactorization := record
0027:     base: Integer;              // the base of the factor
0028:     exponent: Integer;          // the exponent of the factor
0029:     value: Integer;             // the currently examined value for this defactorization
0030: endrecord;
0031:
0032: // examine storage requirements by splitting the segment specifications
0033: // parameters:
0034: //      specs                   the segment specifications to examine
0035: //      t                       the playback time (in slot intervals) to calculate the storage requirements
0036: //                              for
0037: //      defactorizations        the currently selected set of defactorizations
0038: //      storage_requirement     the calculated storage requirement
0039: procedure examine_by_splitting(
0040:     in specs: List of Spec,
0041:     in t: Integer,
0042:     in defactorizations: List of Defactorization,
0043:     out storage_requirement: Integer)
0044: var spec: Spec;
0045:     prime_factors: List of Factor;
0046:     prime_factor: Factor;
0047:     defactorization: Defactorization;
0048:     factors: List of Integer;
0049:     new_factors: List of Integer;
0050:     new_specs: List of Spec;
0051:     further_specs_found: Boolean;
0052: begin
0053:
0054:     // no specs, then no storage requirements
0055:     if length(specs) = 0
0056:         storage_requirement ← 0;
0057:
0058:     // a single specs, then the segment has to be stored in worst case
0059:     elsif length(specs) = 1
0060:         storage_requirement ← 1;
0061:
0062:     // otherwise its more complicated
0063:     else
0064:
0065:         // extract specs until all specs have been examined
0066:         storage_requirement ← 0;
0067:         while specs ≠ [] do
0068:
0069:             // get the next spec and the prime factors of its period
0070:             spec ← specs[0];
0071:             remove spec from specs;
0072:             get_prime_factors(spec.period, prime_factors);
0073:
0074:             // add the prime factors of this spec which are not defactorized
0075:             foreach prime_factor in prime_factors do
0076:                 defactorization ← select * from defactorizations where
0077:                     defactorizations[*].base = prime_factor.base;
0078:                 if defactorization = ε ∨ prime_factor.exponent < defactorization.exponent then
0079:                     append prime_factor.base to factors;
0080:                 endif;
0081:             endforeach;
0082:
0083:             // start building a subset of specs
0084:             new_specs ← [spec];
0085:             do
0086:
0087:                 // continue until no specs are found any more
0088:                 further_specs_found ← false;
0089:
0090:                 // examine all specs for their prime factors
0091:                 foreach spec in specs do
0092:
0093:                     // check which factors this spec has, disregarding the defactorized ones
0094:                     get_prime_factors(spec.period, prime_factors);
0095:                     new_factors ← [];
0096:                     foreach prime_factor in prime_factors do
0097:                         defactorization ← select * from defactorizations where
0098:                             defactorizations[*].base = prime_factor.base;
0099:                         if defactorization = ε ∨ prime_factor.exponent < defactorization.exponent then
0100:                             append prime_factor.base to new_factors;
0101:                         endif;
0102:                     endforeach;
0103:
0104:                     // if there are common factors, this spec has to be examined together with the previous
0105:                     // ones
0106:                     if intersection(factors, new_factors) ≠ [] then
0107:                         remove spec from specs;
0108:                         append spec to new_specs;
0109:                         factors ← union(factors, new_factors);
0110:                         further_specs_found ← true;
0111:                     endif
0112:                 endforeach;
0113:             while further_specs_found;
0114:
0115:             // examine this subset of specs and add the resulting storage requirements of all subsets
0116:             examine_by_defactorization(new_specs, t, defactorizations, new_storage_requirement);
0117:             storage_requirement ← storage_requirement + new_storage_requirement;
0118:         endwhile;
0119:     endif;
0120: end;
0121:
0122: // examine storage requirements by defactorizing the segment specifications
0123: // parameters:
0124: //      specs                   the segment specifications to examine
0125: //      t                       the playback time (in slot intervals) to calculate the storage requirements
0126: //                              for
0127: //      defactorizations        the currently selected set of defactorizations
0128: //      storage_requirement     the calculated storage requirement
```

```
0129: procedure examine_by_defactorization(
0130:     in specs: List of Spec,
0131:     in t: Integer,
0132:     in defactorizations: List of Defactorization,
0133:     out storage_requirement: Integer)
0134: var factors: List of Factor;
0135:     factor: Factor;
0136:     spec: Spec;
0137:     prime_factors: List of Factor;
0138:     prime_factor: Factor;
0139:     defactorization: Defactorization;
0140:     exponent: Integer;
0141:     i: Integer;
0142:     new_specs: List of Spec;
0143:     add_storage_requirement: Integer;
0144:     period: Integer;
0145:     phase_shift: Integer;
0146:     fperiod: Integer;
0147:     fphase: Integer;
0148:     new_storage_requirement: Integer;
0149: begin
0150:
0151:     // no specs, then no storage requirements
0152:     if length(specs) = 0
0153:         storage_requirement ← 0;
0154:
0155:     // a single specs, then the segment has to be stored in worst case
0156:     elsif length(specs) = 1
0157:         storage_requirement ← 1;
0158:
0159:     // otherwise its more complicated
0160:     else
0161:
0162:         // get the prime factors of all segment specs, disregarding factors which already have been
0163:         // defactorized
0164:         factors ← [];
0165:         foreach spec in specs do
0166:             get_prime_factors(spec.period, prime_factors);
0167:             foreach prime_factor in prime_factors do
0168:                 defactorization ← select * from defactorizations where
0169:                     defactorizations[*].base = prime_factor.base;
0170:                 if defactorization = ε then
0171:                     exponent ← prime_factor.exponent;
0172:                 else
0173:                     exponent ← prime_factor.exponent - defactorization.exponent;
0174:                 endif;
0175:                 factor ← select * from factors where factors[*].base = prime_factor.base;
0176:                 if factor = ε then
0177:                     append new Factor(base ← factor.base, exponent ← exponent) to factors;
0178:                 else
0179:                     factor.exponent ← exponent;
0180:                 endif;
0181:             endforeach;
0182:         endforeach;
0183:
0184:         // select a prime factor we want to defactorize; this is just a heuristic
0185:         factor ← select * from factors where
0186:             firstly factors[*].exponent is greatest and secondly factors[*].base is smallest;
0187:
0188:         // update the currently used defactorization to reflect the new factor
0189:         defactorization ← select * from defactorizations where defactorizations[*].base = factor.base;
0190:         if defactorization = ε then
0191:             append new Defactorization(base ← factor.base, exponent ← 0, value ← 0) to
0192:                 defactorizations;
0193:         endif;
0194:         defactorization.exponent ← defactorization.exponent + 1;
0195:
0196:         // examine the storage requirements for each value of the factor
0197:         storage_requirement ← 0;
0198:         for i ← 0 to defactorization.base - 1 do
0199:
0200:             // calculate the segment specifications for the newly selected defactorization
0201:             new_specs ← [];
0202:             add_storage_requirement ← 0;
0203:             foreach spec in specs do
0204:
0205:                 // if the segment is not influenced by this defactorization, add it to the new list
0206:                 if spec.period % (defactorization.base ^ defactorization.exponent) ≠ 0 then
0207:                     append spec to new_specs;
0208:
0209:                 // otherwise check how it is influenced
0210:                 else
0211:
0212:                     // calculate the time when the segment is received when the current defactorization is
0213:                     // in use
0214:                     period ← 1;
0215:                     phase_shift ← 0;
0216:                     get_prime_factors(spec.period, prime_factors);
0217:                     foreach prime_factor in prime_factors do
0218:                         defactorization ← select * from defactorizations where
0219:                             defactorizations[*].base = prime_factor.base;
0220:                         if defactorization.base ≠ ε then
0221:                             fperiod ← prime_factor.base ^
0222:                                 min(prime_factor.exponent, defactorization.exponent);
0223:                             fphase ← defactorization.value % fperiod;
0224:                             egcd(period, fperiod, u, v);
0225:                             phase_shift ← (u · period · fphase + v · fperiod · phase_shift) %
0226:                                 (period · fperiod);
0227:                             period ← period · fperiod;
0228:                         endif;
0229:                     endforeach;
0230:
0231:                     // check if the segment is received at all under the current defactorization
0232:                     if spec.number ≥ period ∨ (phase_shift - spec.first) % period < spec.number then
0233:
0234:                         // check if it is received exactly under the current defactorization
0235:                         if spec.period = period then
0236:
0237:                             // then we do not need to examine it further, just increment the storage
0238:                             // requirements
0239:                             add_storage_requirement ← add_storage_requirement + 1;
0240:
0241:                         // otherwise we have to examine this segment specification further
0242:                         else
0243:
0244:                             // so add it to the list for further examination
0245:                             append spec to new_specs;
```

```
0246:                         endif;
0247:                     endif;
0248:                 endif;
0249:             endforeach;
0250:
0251:             // examine the reduced set of segment specifications
0252:             examine_by_splitting(new_specs, t, defactorizations, new_storage_requirement);
0253:
0254:             // if the resulting storage requirements are higher than the previous ones, remember the new
0255:             // result
0256:             if new_storage_requirement + add_storage_requirement > storage_requirement then
0257:                 storage_requirement ← new_storage_requirement + add_storage_requirement;
0258:             endif;
0259:
0260:             // update the defactorization value
0261:             defactorization.value ← defactorization.value +
0262:                 defactorization.base ^ (defactorization.exponent - 1);
0263:         endfor;
0264:     endif;
0265: end;
0266:
0267: // calculate exact storage requirements
0268: // parameters:
0269: //      segments                the segments to examine
0270: //      t                       the playback time (in slot intervals) to calculate the storage requirements
0271: //                              for
0272: //      storage_requirement     the calculated storage requirement
0273: procedure calculate_storage_requirement(
0274:     in segments: List of Segment,
0275:     in t: Integer,
0276:     out storage_requirement: Integer)
0277: var specs: List of Spec;
0278: begin
0279:
0280:     // calculate which segments are received under which circumstances (i. e. which join time); therefore
0281:     // examine all segments
0282:     specs ← [];
0283:     foreach segment in segments do
0284:
0285:         // check if the segment has not yet been played
0286:         if t ≤ segment.needed_period - 1 then
0287:
0288:             // calculate the join times of the receiver which received the segment first and the number
0289:             // of slot intervals of receivers which received the segment, too
0290:             first ← (segment.phase_shift - t) % segment.period;
0291:             number ← min(t + 1, segment.period, segment.period + t - segment.needed_period + 1);
0292:
0293:             // if the segment is received (and will not be received again) by any receiver, save this
0294:             if number > 0 then
0295:                 append new Spec(segment_no ← segment.segment_no, first ← first, number ← number,
0296:                     period ← segment.period) to specs;
0297:             endif;
0298:         endif;
0299:     endforeach;
0300:
0301:     // start calculation
0302:     examine_by_splitting(specs, t, [], storage_requirement);
0303: end
```

A.7 Fast Storage Requirement Estimation

```
0001: // type for nodes in tree-representation of transmission trees
0002: type Node := record
0003:     period: Integer;            // period of node
0004:     phase_shift: Integer;       // phase shift of node
0005:     tree_no: Integer;           // number of tree this node is part of
0006:     segment_no: Integer;        // segment of this node, 0 if none
0007:     children: List of Node;     // child nodes
0008: endrecord;
0009:
0010: // type for a segment of the transmission scheme
0011: type Segment := record
0012:     segment_no: Integer;        // segment number
0013:     required_period: Integer;   // required transmission period of the segment
0014:     period: Integer;            // utilized period in the transmission scheme
0015:     phase_shift: Integer;       // phase shift of the segment in the transmission scheme
0016:     channel_no: Integer;        // channel number of the segment in the transmission scheme
0017: endrecord;
0018:
0019: // calculate the storage requirements for a single node
0020: //      t                       the playback time (in slot intervals) to calculate the storage requirements
0021: //                              for
0022: //      number                  the number of transmissions of this node since joining the transmission
0023: //      node                    the node to examine
0024: //      storage_requirement     the calculated storage requirement
0025: procedure examine_node(
0026:     in t: Integer,
0027:     in number: Integer,
0028:     in node: Node,
0029:     out storage_requirement: Integer)
0030: var child: Node;
0031:     i: Integer;
0032:     j: Integer;
0033:     low: List of Integer;
0034:     hi: List of Integer;
0035:     add_storage_requirement: Integer;
0036: begin
0037:
0038:     // check if the node has been transmitted at all
0039:     storage_requirement ← 0;
0040:     if number > 0 then
0041:
0042:         // check if the node is a leaf node
0043:         if length(node.children) = 0 then
0044:
0045:             // check if the node has a segment attached to it
0046:             if node.segment_no ≠ 0 then
0047:
```

```
0048:                 // check if the segment of the node has not yet been played
0049:                 segment ← select * from segments where segments.segment_no = node.segment_no;
0050:                 if t ≤ segment.needed_period - 1 ∧
0051:                     segment.period + t - segment.needed_period + 1 > 0 then
0052:                     storage_requirement ← 1;
0053:                 endif;
0054:             endif;
0055:
0056:         // otherwise it is a non-leaf node
0057:         else
0058:
0059:             // if the number of transmissions of this node is dividable by the number of its children, each
0060:             // child has been transmitted equally often
0061:             if number % length(node.children) = 0 then
0062:
0063:                 // in this case we examine the storage requirements of the child nodes for the according
0064:                 // fraction of transmissions and add these values
0065:                 foreach child in node.children do
0066:                     examine_node(t, number ÷ length(node.children), child, add_storage_requirement);
0067:                     storage_requirement ← storage_requirement + add_storage_requirement;
0068:                 endforeach;
0069:
0070:             // otherwise it is not clear how often each of the child nodes have been transmitted
0071:             else
0072:
0073:                 // but the number of transmissions of the child node only differ by one and are consecutive
0074:                 // in a cyclic manner; so calculate the storage requirements of the childs for both cases
0075:                 low ← [];
0076:                 hi ← [];
0077:                 foreach child in node.children do
0078:                     examine_node(t, number ÷ length(node.children), child, add_storage_requirement);
0079:                     append add_storage_requirement to low;
0080:                     examine_node(t, number ÷ length(node.children) + 1, child, add_storage_requirement);
0081:                     append add_storage_requirement to hi;
0082:                 endforeach;
0083:
0084:                 // now iterate all possibilities which childs have been transmitted less often and which more
0085:                 // often
0086:                 for i ← 0 to length(node.children) do
0087:
0088:                     // add storage requirements of all childs
0089:                     add_storage_requirement ← 0;
0090:                     for j ← 0 to length(node.children) - 1 do
0091:
0092:                         // add some storage requirements of the more often transmitted nodes and some of the
0093:                         // less often transmitted ones
0094:                         if j < number % length(node.children) then
0095:                             add_storage_requirement ← add_storage_requirement +
0096:                                 hi[(i + j) % length(nod.children)];
0097:                         else
0098:                             add_storage_requirement ← add_storage_requirement +
0099:                                 low[(i + j) % length(nod.children)];
0100:                         endif;
0101:                     endfor;
0102:
0103:                     // if the result is higher than the previous ones, remember it
0104:                     if add_storage_requirement > storage_requirement then
0105:                         storage_requirement ← add_storage_requirement;
0106:                     endif;
0107:                 endfor;
0108:             endif;
0109:         endif;
0110:     endif;
0111: end;
0112:
0113: // calculate approximate storage requirements
0114: // parameters:
0115: //      segments                the segments to examine
0116: //      trees                   tree representation of the transmission schedule
0117: //      t                       the playback time (in slot intervals) to calculate the storage requirements
0118: //                              for
0119: //      storage_requirement     the calculated storage requirement
0120: procedure calculate_storage_requirement(
0121:     in segments: List of Segment,
0122:     in trees: List of Node,
0123:     in t: Integer,
0124:     out storage_requirement: Integer)
0125: var tree: Node;
0126:     add_storage_requirement: Integer;
0127: begin
0128:
0129:     // examine the storage requirements of each tree and add them
0130:     storage_requirement ← 0;
0131:     foreach tree in trees do
0132:         examine_node(t, t + 1, tree, add_storage_requirement);
0133:         storage_requirement ← storage_requirement + add_storage_requirement;
0134:     endforeach;
0135: end;
```

Appendix B

Nomenclature

IN this thesis, symbol names in equations have been used uniformly wherever possible. The following table provides a reference of the used symbol, its meaning and unit. For scalar values, the unit is displayed as 1, for functions, domain and codomain are separated by an arrow (→).

Symbol	Description	Unit
b	bandwidth of media stream (only applicable for constant-bit-rate media streams)	$\frac{bytes}{s}$
d	total duration of media stream	s
$f(t)$	cumulative bandwidth function, $f(t)$ is the position of the first byte in the media stream which is played at time t after beginning of playback	$s \rightarrow bytes$
$f^{-1}(p)$	inverse cumulative bandwidth function, $f^{-1}(p)$ is the time stamp in the media stream when the position p is reached	$bytes \rightarrow s$
n	number of segments of the media stream	1
s	size of media stream	$bytes$
B	total sender bandwidth of the transmission	$\frac{bytes}{s}$
B^-	theoretical lower bound for B	$\frac{bytes}{s}$
B°	average total sender bandwidth of the transmission	$\frac{bytes}{s}$
B_i	bandwidth of channel #i	$\frac{bytes}{s}$
C	fraction for lowering channel bandwidth	1
$F(t)$	playback function of the receiver system, based on the utilized playback delay; $F(t)$ is the position in the media stream which is played at time t	$s \rightarrow bytes$
$F^-(t)$	playback function of the receiver system, based on the minimum playback delay	$s \rightarrow bytes$

Symbol	Description	Unit
$F^{+}(t)$	playback function of the receiver system, based on the maximum playback delay	$s \rightarrow bytes$
$F^{-1}(p)$	inverse playback function, based on utilized playback delay	$bytes \rightarrow s$
$F^{--1}(p)$	inverse playback function, based on minimum playback delay	$bytes \rightarrow s$
$F^{+-1}(p)$	inverse playback function, based on maximum playback delay	$bytes \rightarrow s$
$\mathcal{F}(t)$	antiderivative of the playback function	s
J	protocol specific latency for leaving and joining a channel	s
J_i	joining latency of channel #i relative to beginning of reception	s
L	Prime split limit; if $\left\lfloor \frac{\pi_i^+}{\Pi_N} \right\rfloor \geq L \cdot \Pi_N + 1$, a prime split of a node is prohibited	1
N	number of channels of the transmission	1
P	size of preloaded amount of media stream	$bytes$
$M^{+}(t)$	worst case storage requirement function, $M^{+}(t)$ gives the worst case storage requirement of a playback at time t	$s \rightarrow bytes$
P_i	total period of channel #i	1
P_i^+	imposed limit on P_i	1
$Q_{i,N}$	quality value for scheduling segment #i at node N, used to find the best node in the scheduling algorithm	1
$Q_{i,N}^{I}$	first of the quality values which are combined to $Q_{i,N}$	1
$Q_{i,N}^{II}$	second of the quality values which are combined to $Q_{i,N}$	1
$Q_{i,N}^{III}$	third of the quality values which are combined to $Q_{i,N}$	1
R	number of channels the receiver system can receive simultaneously	1
S	broadcasting series of size-based transmission schemes	$1 \rightarrow 1$
$T_{\text{playbackdelay}}(t)$	transmission function for additional playback delay	$s \rightarrow s$
$T_{\text{partialpreloading}}(p)$	transmission function for partial preloading	$bytes \rightarrow bytes$
$T_{\text{breakinsertion}}(t)$	transmission function for break insertion	$s \rightarrow s$
W^{-}	minimum possible playback delay for the transmission	s
W^{+}	maximum possible playback delay for the transmission	s
W°	average playback delay for the transmission	s
W^{II}	weight for weighting of $Q_{i,N}^{II}$ in $Q_{i,N}$	1
W^{III}	weight for weighting of $Q_{i,N}^{III}$ in $Q_{i,N}$	1
β	bandwidth of each segment transmission/bandwidth of each channel, $\beta = \frac{\sigma}{\delta}$	$\frac{bytes}{s}$
β_i	bandwidth of segment #i in the schedule	$\frac{bytes}{s}$

Symbol	Description	Unit
δ	slot interval = transmission duration of one segment	s
δ_i	duration of a segment #i	s
κ_i	channel number which is used for segment #i in the schedule; symbol taken from the Greek word κανάλι	1
π_i	utilized period of segment #i in the schedule; symbol taken from the Greek word περίοδος	1
π_i^+	maximum period of segment #i which is needed in the schedule for proper playback	1
σ	size of segments	$bytes$
σ_i	size of segment #i	$bytes$
τ_i	playback time of segment #i	s
ϕ_i	phase shift of segment #i, $0 \leq \phi_i < \pi_i$; symbol taken from the Greek word φάση	1
K_N	channel number which is used for node #i in the schedule	1
Π_N	utilized period of node #N in the schedule	1
Φ_N	phase shift of node #N, $0 \leq \Phi_i < \Pi_i$	1
η	efficiency of a transmission schedule, $0 < \eta \leq 1$	1
e	Euler's number, the base of the natural logarithm ($e \approx 2.718\,281\,828$)	1
γ	Euler-Mascheroni constant ($\gamma \approx 0.577\,215\,665$)	1

The following mathematical symbols, functions and operators have been used throughout this thesis:

Operator	Description
∞	infinity
$\lfloor\rfloor$	truncation to next integer (towards $-\infty$)
$\lceil\rceil$	rounding up to next integer (towards $+\infty$)
$\circ$	function composition, $(f \circ g)(x) = f(g(x))$
$\wedge$	logical and
$\vee$	logical or
div	integer division, a div $b = \left\lfloor \frac{a}{b} \right\rfloor$
gcd	greatest common divisor
lcm	least common multiple
mod	integer modulo, a mod $b = a - b \cdot \left\lfloor \frac{a}{b} \right\rfloor$

Appendix C

Abbreviations

THIS appendix lists all abbreviations which are used in this thesis for convenience reasons.

Abbreviation	Meaning
ACK	**Ack**nowledge; used to signal successful reception of data
ADSL	**A**symmetric **d**igital **s**ubscriber **l**ine; DSL variant where the download link has a higher bandwidth than the upload link
ARQ	**A**utomatic **r**epeat re**q**uest; method for reliable data transmission using ACK and/or NAK requests
b/w	**b**and**w**idth; used in some table headings for compactness
CATV	**Ca**ble **t**ele**v**ision or **c**ommunity **a**ntenna **t**ele**v**ision; a television (and radio) network based on optical fibers or coaxial cables
CBR	**C**onstant-**b**it-**r**ate; used in the context of media streams
CPE	**C**ustomer **p**remise **e**quipment; in context of media-on-demand: equipment which is connected to a carrier's network and which is responsible for interacting with the media-on-demand sender system and receiving and displaying media streams to the consumer
CPS	**C**ontent **p**rovider **s**ystem; system which offers and sells media streams to service providers; part of a media-on-demand system according to the DAVIC reference model
DAVIC	**D**igital **A**udio **Vi**sual **C**ouncil; a non-profit organization which has been created to provide end-to-end interoperability of broadcast and interactive digital audio-visual information

Abbreviation	Meaning
DSL	**D**igital **s**ubscriber **l**ine; a common type of home Internet access using telephone lines; compared to traditional home Internet access via acoustic modems or ISDN, DSL provides a much higher data rate (currently up to 15 Mbit/s)
DVD	**D**igital **v**ersatile **d**isk
EUR	Currency of the **Eur**opean Currency Union
FCFS	**F**irst **c**ome **f**irst **s**erved; scheduling strategy where the requests are served one after another in the order of requests
FEC	**F**orward **e**rror **c**orrection; increasing reliability of a transmission by adding redundancy to support recovery of lost data
ISDN	**I**ntegrated **s**ervices **d**igital **n**etwork; a type of digital, circuit switched telephone network which is capable to transmit both voice and digital data; sometimes used for home Internet access (providing a data rate of up to 128 Kbit/s on copper telephone lines)
MFQL	**M**aximum **f**actored **q**ueue **l**ength; strategy where the requests are grouped in queued depending on the requested media stream and served sorted by queue length times a queue specific factor, starting with the longest product
MQL	**M**aximum **q**ueue **l**ength; batching strategy where the requests are grouped in queued depending on the requested media stream and served sorted by queue length, starting with the longest queue
NAK	**N**egative **ac**k**k**nowledge; used to signal data losses
RTCP	**R**eal-time **t**ransport **c**ontrol **p**rotocol; protocol for transmission of auxiliary data (e. g. for stream synchronization and reception quality feedback) for an RTP transmission
RTP	**R**eal-time **t**ransport **p**rotocol; protocol for transmission of data streams with timed delivery
SCS	**S**ervice **c**onsumer **s**ystem; system which allows a consumer to receive a media stream from a service provider system and which is implemented in a customer premise equipment; part of a media-on-demand system according to the DAVIC reference model

Abbreviation	Meaning
SDP	**S**ession **d**escription **p**rotocol; protocol for describing and announcing multimedia session parameters
SIP	**S**ession **i**nitiation **p**rotocol; protocol for initiating and controlling multimedia sessions
SPS	**S**ervice **p**rovider **s**ystem; system which keeps media streams from a content provider system in its media database and which accepts and processes schedules (either fixed ones or based on requests) for delivery of media streams to service consumer systems; part of a media-on-demand system according to the DAVIC reference model
STB	**S**et-**t**op **b**ox; a device which receives data from a network (cable, satellite, radio broadcast) and prepares it for displaying on a TV system
TV	**T**ele**v**ision
US, USA	**U**nited States (of **A**merica)
USD	**U**nited **S**tates **D**ollar; see below for exchange rate which has been used throughout this thesis
VBR	**V**ariable-**b**it-**r**ate; used in the context of media streams
VCR	**V**ideo **c**assette **r**ecorder
VDSL	**V**ery high data rate **d**igital **s**ubscriber **l**ine; ADSL successor which provides a download bit rate of up to 51.85 Mbit/s

For the cost analysis in chapter 6 in this thesis, results from various marketing studies have been used. Where values must be compared or are included in calculations, they have been converted to euro (EUR), the currency unit of the European currency union. The following exchange rate for United States dollar (USD) has been taken on April 23, 2008 and has been used within this thesis:

$$1.00 \text{ EUR} = 1.59 \text{ USD}$$

The results of currency conversions have been rounded to a precision of three digits.

Appendix D

Typesetting Conventions

FOR improved clearness, the following typesetting rules have have been used throughout this thesis:

- Definitions of terms which are used unambiguously in this thesis are printed in bold letters:

  Media-on-demand is a technique for transmitting **media streams** from **media-on-demand sender systems** to **media-on-demand receiver systems** in **real-time**.

- To emphasize text, italics is used:

  With media-on-demand, the viewers can select themselves *what to see* and the time *when to see* it.

- Quotations are typed in italics and indented left and right:

  Video on Demand (VOD) is a technology to view and order video content as and when required in the home [. . .].

- Program code is printed in a small fixed width font, prefixing each line with a line number and using bold letters for keywords:

```
0001: // calculate the greatest common divisor of two integers
0002: // parameters:
0003: //      f               the first integer
0004: //      g               the second integer
0005: //      result          the greatest common divisor of f and g
0006: procedure gcd(
0007:     in f: Integer,
0008:     in g: Integer,
0009:     out result: Integer)
0010:     var remainder: Integer;
0011: begin
0012:
0013:     // use euclidean algorithm for calculation
0014:     while g ≠ 0 do
0015:         remainder ← f % g;
0016:         f ← g;
0017:         g ← remainder;
0018:     endwhile;
0019:
0020:     // return the result
0021:     result ← f;
0022: end;
```

- Equations are written indented and are labeled at the right margin. Symbols in equations are always set in italic type and follow additionally these conventions:
  - Values related to segments are represented using lowercase Greek letters, suffixed by an index which denotes the segment number where needed:

    Segment #i has period π_i.
  - For values related to nodes of the tree representation of a transmission scheme, uppercase Greek letters, suffixed by a node where needed:

    Node #N has period Π_N.
  - For values related to the media stream, lowercase Roman letters are used:

    The total duration of the media stream is d.
  - To represent values specific to a media stream transmission, uppercase Roman letters are utilized:

    The total sender bandwidth is B.

  Regardless of these rules, η is used for the efficiency of transmission schemes due to common conventions.
- Numbers are shown in groups of three digits, using a period as decimal point:

  $\pi^{10} \approx 93\,648.047\,477$

  Opposed to this rule, years are given in a single group:

  In year 2000, (opposed to 2 000 segments) ...
- Citations are shown in brackets, using the first three characters of the last name of the author in case of a single author, the initials of the last names of all authors in case of two to four authors or the initials of the last names of the first four authors suffixed by a plus sign if the citation has more than four authors, always followed by the year of publication:

  In [BNL02], the authors proposed ...

Appendix E

Available Media-on-Demand Systems

SINCE 1990 when the first commercial video-on-demand service was offered in Hong Kong [Wik08e, Oft03], many companies entered the video-on-demand market and developed their own systems. But nearly all media-on-demand systems are still based on simple media-on-demand transmission schemes. Table E.1 and E.2 show a list of currently available video-on-demand solutions, the supported transmission scheme[1] and web links to web locations where all data about the systems has been retrieved.

Most of these systems are based on one of the following three transmission schemes:

- Point-to-Point Transmissions: Most of the media-on-demand systems use Point-to-Point connections for streaming media. The video-on-demand architecture recommendation "Demand Adaptive and Locality Aware (DALA) streaming media server cluster architecture" [GJS02] and the video-on-demand system architecture of the "Digital Audio Visual Council" [Dav99] are based on Point-to-Point transmissions.
- Single Broadcast: Some of the systems support media broadcast (often including live broadcast) using multicasting, unfortunately only at predefined time and frequency.
- Complete Prefetching: Although none of the companies in the list sell Media-on-Demand systems using Complete Prefetching, many of them have a (non-real-time) content distribution product which may be used for this purpose.

The only exception the author of this thesis has knowledge of which uses an enhanced media-on-demand transmission scheme (the original Greedy Broadcasting scheme) is the TelliVision system from Tellitec Communications bvba, a system which has been developed partially by the author.

Many other companies are involved in media-on-demand techniques, too, e. g. by developing specialized video streaming hardware. But as this thesis focuses to media-on-demand transmission

[1] The list of supported transmission schemes may be incomplete in some cases because of missing information in product descriptions.

Company	System	Scheme	References
Apple Computer, Inc.	Quicktime Broadcaster, Quicktime Streaming Server, Darwin Streaming Server	SB, PtP	http://www.apple.com/de/quicktime/broadcaster/
Cisco Systems, Inc.	Cisco IPTV video server	PtP, SB	http://www.cisco.com/en/US/netsol/ns340/ns394/ns158/ns88/networking_solutions_package.html
Compagnie Financière Alcatel	Alcatel Open Media Content Manager	PtP, SB	http://www.alcatel.de/products/productsummary.jhtml?repositoryID=/com/en/appxml/opgproduct/alcatel5959openmediacontentmanagertcm228121551635.jhtml
Concurrent Computer Corp.	MediaHawk	PtP	http://www.adelphia.com/pdf/Adelphia_LA_VOD.pdf
Envivio, Inc.	Envivio 4Caster/IPTV	PtP, SB	http://www.envivio.com/solutions/iptv.php
Hewlett-Packard Development Company, L. P.	HP IPTV	PtP, SB	http://h71028.www7.hp.com/enterprise/cache/312831-0-0-225-121.html
IBM Corp.	IPTV	PtP	http://www-03.ibm.com/servers/eserver/xseries/solutions/ind/iptv_overview.html
InfoValue Computing, Inc.	QuickVideo	PtP, SB	http://www.infovalue.com/products/product.htm
Kasenna	Kasenna MediaBase video server software	PtP, SB	http://www.kasenna.com/solutions/products/mediabase_xmp.php
Minerva Networks, Inc.	iTVManager	PtP, SB	http://www.minervanetworks.com/images/itvManager2.pdf

PtP=Point-to-Point, SB=Single Broadcasting, GB=Greedy Broadcasting

Table E.1: Available media-on-demand systems (selection)

Company	System	Scheme	References
Netscape Communications, Corp./Insoft Network Television, Corp.	INTV!	PtP	http://wp.netscape.com/newsref/pr/newsrelease81.html
Oracle Corp.	Oracle Media Server	PtP	http://www.vccs.edu/its/projects/oracle/ovs-gl1.htm
RealNetworks, Inc.	Helix Server	PtP, SB	http://www.realnetworks.com/products/media_delivery.html
SeaChange	SeaChange IP Video Solutions	PtP, SB	http://www.schange.com/Products/IP_Video/Main.asp
Starlight Networks	StarWorks	PtP	http://www.highbeam.com/library/docfree.asp?DOCID=1G1:18742882\&ctrlInfo=Round19%3AMode19b%3ADocG%3AResult\&ao=
Stony Brook University	Stony Brook Video Server	PtP	http://www.ecsl.cs.sunysb.edu/~andrew/VideoServer/videoserver/index/book1.html
Sun ShowMe	Sun Microsystems	SB	http://www.sun.com/products-n-solutions/hardware/docs/pdf/802-6020-13.pdf
Newtec Cy N. V.	TelliVision	PtP, GB	http://www.tellitec.be/tellivision/index.html
Thirdspace Living Ltd.	Thirdspace Open Video Server	PtP	http://ccur.com/corp_news_pressrelease.asp?pressreleaseid=269
University of Berkeley	Berkeley Distributed Video-on-Demand System	PtP	http://bmrc.berkeley.edu/frame/research/storage/

Table E.2: Available Media-on-Demand Systems (PtP=Point-to-Point, SB=Single Broadcasting, GB=Greedy Broadcasting)

techniques, a comparison of different hardware systems for media-on-demand is beyond the scope of this thesis.

Bibliography

[ABEL94] A. Albanese, J. Blömer, J. Edmonds, and M. Luby, "Priority encoding transmission," in *Proc. Annual IEEE Symposium on Foundations of Computer Science (FOCS)*, Santa Fe, New Mexico, Nov. 1994, pp. 604–612.

[Arc] "Arcor video on demand." Arcor AG & Co. KG. [Online]. Available: http://www.arcor.de/vod/vod_1_0.jsp

[ASS01] S. V. Anastasiadis, K. C. Sevcik, and M. Stumm, "Server-based smoothing of variable bit-rate streams," in *Proc. Annual ACM Int. Conf. on Multimedia (ACM MM)*, Ottawa, ON, USA, Oct. 2001, pp. 147–158.

[AWY96a] C. C. Aggarwal, J. L. Wolf, and P. S. Yu, "On optimal piggyback merging policies for video-on-demand systems," in *Proc. ACM Int. Conf. on Measurement and Modeling of Computer Systems (SIGMETRICS)*, Philadelphia, PA, USA, May 1996, pp. 200–209.

[AWY96b] C. C. Aggarwal, J. L. Wolf, and P. S. Yu, "A permutation-based pyramid broadcasting scheme for video-on-demand systems," in *Proc. IEEE Int. Conf. on Multimedia Computing and Systems (ICMCS)*, Tokyo, Japan, June 1996, pp. 118–126.

[AWY01] C. C. Aggarwal, J. L. Wolf, and P. S. Yu, "The maximum factor queue length batching scheme for video-on-demand systems," *IEEE Trans. on Computers*, vol. 50, no. 2, pp. 97–110, 2001.

[BBDF+01] C. Bormann, C. Burmeister, M. Degermark, H. Fukushima, H. Hannu, L.-E. Jonsson, R. Hakenberg, T. Koren, K. Le, Z. Lie, A. Martensson, A. Miyanzaki, K. Svanbro, T. Wiebke, T. Yoshimura, and H. Zheng, "RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP and uncompressed," IETF Proposed Standard RFC 3095, July 2001.

[BCMM00] G. Boggia, P. Camarda, P. L. Mazzeo, and M. Mongiello, "Multicast distribution for multimedia on demand services," in *Proc. Int. Conf. on Communication Technology (ICCT)*, Beijing, China, Aug. 2000, pp. 1236–1243.

[Ber05a] D. S. Bernstein, "Matrix mathematics." Princeton University Press, 2005, p. 119.

[Ber05b] D. S. Bernstein, "Matrix mathematics." Princeton University Press, 2005, pp. 211–212.

[BKKK+95] J. Blömer, M. Kalfane, R. Kamp, M. Karpinski, M. Luby, and D. Zuckerman, "An XOR-based erasure-resilient coding scheme," International Computer Science Institute, University of Berkeley, Berkeley, CA, USA, Tech. Rep. TR–95–048, 1995.

[BLM98] J. W. Byers, M. Luby, and M. Mitzenmacher, "Accessing multiple mirror sites in parallel: Using Tornado codes to speed up downloads," International Computer Science Institute, University of Berkeley, Berkeley, CA, USA, Tech. Rep. TR-98-021, 1998.

[BLM99] J. W. Byers, M. Luby, and M. Mitzenmacher, "Accessing multiple mirror sites in parallel: Using Tornado codes to speed up downloads," in *Proc. Annual Joint Conf. of the IEEE Computer and Communcations Societies (INFOCOM)*, New York, NY, USA, Mar. 1999, pp. 275–283.

[BLMR98a] J. W. Byers, M. Luby, M. Mitzenmacher, and A. Rege, "A digital fountain approach to reliable distribution of bulk data," in *Proc. ACM Conf. on Applications, Technologies, Architectures and Protocols for Computer Communications (SIGCOMM)*, Vancouver, BC, Canada, Sept. 1998, pp. 56–67.

[BLMR98b] J. W. Byers, M. Luby, M. Mitzenmacher, and A. Rege, "A digital fountain approach to reliable distribution of bulk data," International Computer Science Institute, University of Berkeley, Berkeley, CA, USA, Tech. Rep. TR-98-013, 1998.

[BM99] Y. Birk and R. Mondri, "Tailored transmissions for efficient near-video-on-demand service," in *Proc. IEEE Int. Conf. on Multimedia Computing and Systems (ICMCS)*, Florence, Italy, June 1999, pp. 226–231.

[BMNC+04] M. Baugher, D. McGrew, M. Naslund, E. Carrara, and K. Norrman, "The secure real-time transport protocol (SRTP)," IETF Proposed Standard RFC 3711, Mar. 2004.

[BNBNS98] A. Bar-Noy, R. Bhatia, J. Naor, and B. Schieber, "Minimizing service and operation costs of periodic scheduling," in *Proc. ACM/SIAM Symposium on Discrete Algorithms (SODA)*, San Francisco, CA, USA, Jan. 1998, pp. 11–20.

[BNGL03] A. Bar-Noy, J. Goshi, and R. E. Ladner, "Off-line and on-line guaranteed start-up delay for media-on-demand with stream merging," in *Proc. ACM Symposium on Parallel Algorithms and Architectures (SPAA)*, San Diego, CA, USA, June 2003, pp. 164–173.

[BNGLT02] A. Bar-Noy, J. Goshi, R. E. Ladner, and K. Tam, "Comparison of stream merging algorithms for media-on-demand," in *Proc. Conf. on Multimedia Computing and Networking (MMCN)*, San Jose, CA, USA, Jan. 2002, pp. 115–129.

[BNL02] A. Bar-Noy and R. E. Ladner, "Windows scheduling problems for broadcast systems," in *Proc. ACM/SIAM Symposium on Discrete Algorithms (SODA)*, San Francisco, CA, USA, Jan. 2002, pp. 433–442.

[BNL03] A. Bar-Noy and R. E. Ladner, "Windows scheduling problems for broadcast systems," *SIAM Journal on Computing*, vol. 32, no. 4, pp. 1091–1113, 2003.

[BNL04] A. Bar-Noy and R. E. Ladner, "Efficient algorithms for optimal stream merging for media-on-demand," *SIAM Journal on Computing*, vol. 33, no. 5, pp. 1011–1034, 2004.

[BS91] I. N. Bronštejn and K. A. Semendjaev, "Taschenbuch der Mathematik," 25th ed. Stuttgart/Leipzig, Germany: B. G. Verlagsgesellschaft, 1991, p. 115.

[CC01] S.-H. G. Chan and E. Chang, "Providing scalable on-demand interactive video service by means of multicasting and client buffering," in *Proc. Int. Conf. on Communications (ICC)*, Helsinki, Finland, June 2001, pp. 1607–1611.

[CDJC84] R. A. Comroe and J. Daniel J. Costello, "ARQ schemes for data transmission in mobile radio systems," *IEEE Journal of Selected Areas in Communications*, vol. 2, no. 4, pp. 472–481, 1984.

[CDKF+02] B. Cain, S. Deering, I. Kouvelas, B. Fenner, and A. Thyagarajan, "Internet group management protocol, version 3," IETF Proposed Standard RFC 3376, Oct. 2002.

[CGIM+99] R. Canetti, J. Garay, G. Itkis, D. Micciancio, M. Naor, and B. Pinkas, "Multicast security: A taxonomy and some efficient constructions," in *Proc. Annual Joint Conf. of the IEEE Computer and Communcations Societies (INFOCOM)*, New York, NY, USA, Mar. 1999, pp. 708–716.

[CHIS05] H. S. Cruickshank, M. P. Howarth, S. Iyengar, and Z. Sun, "A comparison between DVB conditional access and secure IP multicast," in *Proc. IST Mobile & Wireless Communications Summit*, Dresden, Germany, June 2005.

[CHV99] Y. Cai, K. A. Hua, and K. Vu, "Optimizing patching performance," in *Proc. Conf. on Multimedia Computing and Networking (MMCN)*, San Jose, CA, USA, Jan. 1999, pp. 204–215.

[Cis04] "How Starwood Hotels reduced costs and increased revenues with Cisco LRE solution." Cisco Systems, Inc. 1992–2004. [Online]. Available: http://www.cisco.com/warp/public/cc/pd/si/casi/ps5213/prodlit/swood_cs.pdf

[Cis05] "Cisco glossary v." Cisco Systems, Inc. 1992–2005. [Online]. Available: http://www.cisco.com/univercd/cc/td/doc/cisintwk/ita/v12.pdf

[CJ99] S. Casner and V. Jacobson, "Compressing IP/UDP/RTP headers for low-speed serial links," IETF Proposed Standard RFC 2508, Feb. 1999.

[CJM02] E. G. Coffman, jr, P. Jelenković, and P. Momčilović, "The dyadic stream merging algorithm," *ACM Journal on Algorithms*, vol. 43, no. 1, pp. 120–137, 2002.

[CL97] S. W. Carter and D. D. E. Long, "Improving video-on-demand server efficiency through stream tapping," in *Proc. IEEE Int. Conf. on Computer Communications and Networks (ICCCN)*, Las Vegas, NV, USA, Sept. 1997, pp. 200–207.

[CL99] S. W. Carter and D. D. E. Long, "Improving bandwidth efficiency on video-on-demand servers," *Int. Journal of Computer and Telecommunications Networking*, vol. 31, no. 1–2, pp. 111–123, 1999.

[CLP00] S. W. Carter, D. D. E. Long, and J.-F. Pâris, "An efficient implementation of interactive video-on-demand," in *Proc. Int. Symposium on Modeling, Analysis, and Simulation of Computer Systems (MASCOTS)*, San Francisco, CA, USA, Aug. 2000, pp. 172–179.

[CMS93] E. Cruselles, J. L. Melus, and M. Soriano, "An overview of security in Eurocrypt conditional access system," in *Proc. IEEE Global Telecommunications Conf. (GLOBECOM)*, Houston, TX, USA, Nov. 1993, pp. 188–193.

[CNE01] "Microsoft plays for keeps — A $500 million gamble." CNET News. Nov. 2001. [Online]. Available: http://news.com.com/2009-1040-275793.html

[CNE07] "Xbox 306 get video on demand in Europe." CNET News. Dec. 2007. [Online]. Available: http://news.cnet.co.uk/gamesgear/0,39029682,49294611,00.htm

[Com07] "Video-on-demand." The Computer Language Company Inc. 1995–2007. [Online]. Available: http://www.pcmag.com/encyclopedia_term/0%2C2542%2Ct%3Dvideo-on-demand\&i%3D56186%2C00.asp

[CST00] C.-H. Chang, J.-P. Sheu, and Y.-C. Tseng, "A recursive frequency-splitting scheme for broadcasting hot video in VOD service," unpublished slides, Department of CSIE, National Central University, Taiwan, 2000.

[CT99] S.-H. G. Chan and F. A. Tobagi, "On achieving profit in providing near video-on-demand services," in *Proc. Int. Conf. on Communications (ICC)*, Vancouver, Canada, June 1999, pp. 988–994.

[Dav99] "DAVIC reference model." Digital Audio-Visual Council. 1999. [Online]. Available: http://www.davic.org/down1.htm

[Dee89] S. Deering, "Host extensions for IP multicasting," IETF Standard RFC 1112, Aug. 1989.

[Deu] "T-Online Vision-Paket." Deutsche Telecom AG T-Com. [Online]. Available: http://t-online-vision.de/c/48/23/25/4823250.html

[DPK94] A. L. Drapeau, D. A. Patterson, and R. H. Katz, "Toward workload characterization of video server and digital library applications," *ACM SIGMETRICS Performance Evaluation Review*, vol. 22, no. 1, pp. 274–275, 1994.

[DSS94] A. Dan, D. Sitaram, and P. Shahabuddin, "Scheduling policies for an on-demand video server with batching," in *Proc. Annual ACM Int. Conf. on Multimedia (ACM MM)*, San Francisco, CA, USA, Oct. 1994, pp. 15–23.

[(Ed07] A. L. (Ed.), "RTP payload format for generic forward error correction," IETF Proposed Standard RFC 5109, Dec. 2007.

[EFV99] D. L. Eager, M. C. Ferris, and M. K. Vernon, "Optimized regional caching for on-demand data delivery," in *Proc. Conf. on Multimedia Computing and Networking (MMCN)*, San Jose, CA, USA, Jan. 1999, pp. 301–316.

[ES01] L. Engebretsen and M. Sudan. "Harmonic broadcasting is optimal assuming constant bit rate." July 2001. [Online]. Available: http://www.nada.kth.se/~enge/papers/Harmonic.pdf

[ES02] L. Engebretsen and M. Sudan, "Harmonic broadcasting is optimal," in *Proc. ACM/SIAM Symposium on Discrete Algorithms (SODA)*, San Francisco, CA, USA, Jan. 2002, pp. 431–432.

[EV98] D. L. Eager and M. K. Vernon, "Dynamic skyscraper broadcasts for video-on-demand," in *Proc. Workshop on Multimedia Information Systems (MIS)*, Istanbul, Turkey, Sept. 1998, pp. 18–32.

[Eve06] R. Even, "RTP payload format for H.261 video streams," IETF Proposed Standard RFC 4587, Aug. 2006.

[EVZ99a] D. Eager, M. Vernon, and J. Zahorjan, "Minimizing bandwidth requirements for on-demand data delivery," in *Proc. Workshop on Multimedia Information Systems (MIS)*, Indian Wells, CA, USA, Oct. 1999, pp. 80–87.

[EVZ99b] D. Eager, M. Vernon, and J. Zahorjan, "Optimal and efficient merging schedules for video-on-demand servers," in *Proc. Annual ACM Int. Conf. on Multimedia (ACM MM)*, Orlando, FL, USA, Nov. 1999, pp. 199–202.

[EVZ99c] D. Eager, M. Vernon, and J. Zahorjan, "Optimal and efficient merging schedules for video-on-demand servers," University of Washington, Tech. Rep. CSE TR #99-08-01, 1999, unpublished report, thanks to Mr. Eager for sending me a draft version.

[EVZ01] D. Eager, M. Vernon, and J. Zahorjan, "Minimizing bandwidth requirements for on-demand data delivery," *IEEE Trans. on Knowledge and Data Engineering*, vol. 13, no. 5, pp. 742–757, 2001.

[Fen97] W. Fenner, "Internet group management protocol, version 2," IETF Proposed Standard RFC 2236, Nov. 1997.

[Fip02] "Secure hash standard," National Institute of Standards and Technology Standard FIPS 180-2, Aug. 2002.

[FR01] F. H. P. Fitzek and M. Reisslein, "MPEG-4 and H.263 video traces for network performance evaluation," *IEEE Network*, vol. 15, no. 6, pp. 40–54, 2001.

[GC01] M. Gallant and G. Cote, "High rate, high resolution video using h26l," Santa Barbara, CA, USA, Sept. 2001, ITU-T Q.15/16, document VCEG-N84.

[GH90] A. D. Gelman and S. Halfin, "Analysis of resource sharing in information providing services," in *Proc. IEEE Global Telecommunications Conf. (GLOBECOM)*, San Diego, CA, USA, Dec. 1990, pp. 312–316.

[Gin96a] D. Ginsburg, "ATM solutions for enterprise internetworking." Essex, England: Addison-Wesley Longman Ltd., 1996, pp. 203–208.

[Gin96b] D. Ginsburg, "ATM solutions for enterprise internetworking." Essex, England: Addison-Wesley Longman Ltd., 1996, pp. 338–339.

[GJS02] Z. Ge, P. Ji, and P. Shenoy, "A Demand Adaptive and Locality Aware (DALA) streaming media server cluster architecture," in *Proc. ACM Int. Workshop on Network and Operation System Support for Digital Audio and Video (NOSSDAV)*, Miami, FL, USA, May 2002, pp. 139–146.

[GKSW91] A. D. Gelman, H. Kobrinski, L. S. Smoot, and S. B. Weinstein, "A store-and-forward architecture for video-on-demand service," in *Proc. Int. Conf. on Communications (ICC)*, Denver, CO, USA, June 1991, pp. 842–846.

[GKT02] L. Gao, J. Kurose, and D. Towsley, "Efficient schemes for broadcasting popular videos," *ACM Multimedia Systems Journal*, vol. 8, no. 4, pp. 284–294, 2002.

[GLM95] L. Golubchik, J. C. S. Lui, and R. R. Muntz, "Reducing I/O demand in video-on-demand storage servers," in *Proc. ACM Int. Conf. on Measurement and Modeling of Computer Systems (SIGMETRICS)*, Ottawa, Ontario, Canada, May 1995, pp. 25–36.

[GLM96] L. Golubchik, J. C. S. Lui, and R. R. Muntz, "Adaptive piggybacking: A novel technique for data sharing in video-on-demand storage servers," *ACM Multimedia Systems Journal*, vol. 4, no. 3, pp. 140–155, 1996.

[Goo05] "Rights for distribution." Google Answers. 2005. [Online]. Available: http://answers.google.com/answers/threadview?id=520295

[GT99] L. Gao and D. Towsley, "Supplying instantaneous video-on-demand services using controlled multicast," in *Proc. IEEE Int. Conf. on Multimedia Computing and Systems (ICMCS)*, Florence, Italy, June 1999, pp. 117–121.

[GT01] L. Gao and D. Towsley, "Threshold-based multicast for continuous media delivery," *IEEE Trans. on Multimedia*, vol. 3, no. 4, pp. 405–414, 2001.

[Han07] "Alice homeTV bringt Oscar prämierte Filme ins Wohnzimmer," Hansenet, 2007. [Online]. Available: http://www.alice-dsl.de/kundencenter/export/de/unternehmen/presse/news/2007/pdf/2007_03_12_alice_und_warner_bros._starten_kooperation.pdf

[HCS98a] K. A. Hua, Y. Cai, and S. Sheu, "Exploiting client bandwidth for more efficient video broadcast," in *Proc. IEEE Int. Conf. on Computer Communications and Networks (ICCCN)*, Lafayette, LA, USA, Oct. 1998, pp. 848–856.

[HCS98b] K. A. Hua, Y. Cai, and S. Sheu, "Patching: A multicast technique for true video-on-demand services," in *Proc. Annual ACM Int. Conf. on Multimedia (ACM MM)*, Bristol, UK, Sept. 1998, pp. 191–200.

[Hel02] M. E. Hellman, "An overview of public key cryptography," *IEEE Communications Magazine*, vol. 40, no. 5, Part Anniversary, pp. 42–49, 2002.

[HFGC98] D. Hoffman, G. Fernando, V. Goyal, and M. Civanlar, "RTP payload format for MPEG1/MPEG2 video," IETF Proposed Standard RFC 2250, Jan. 1998.

[HJP06] M. Handley, V. Jacobson, and C. Perkins, "SDP: Session description protocol," IETF Proposed Standard RFC 4566, July 2006.

[HNvB99] A. Hu, I. Nikolaidis, and P. van Beek, "On the design of efficient video-on-demand broadcast schedules," in *Proc. Int. Symposium on Modeling, Analysis, and Simulation of Computer Systems (MASCOTS)*, Maryland, MD, USA, Oct. 1999, pp. 262–269.

[HP99] M. Handley and C. Perkins, "Guidelines for writers of RTP payload format specifications," IETF Best Current Practice RFC 2736/BCP 0036, Dec. 1999.

[HPW00] M. Handley, C. Perkins, and E. Whelan, "Session Announcement Protocol," IETF Experimental RFC 2974, Oct. 2000.

[HS97] K. A. Hua and S. Sheu, "Skyscraper broadcasting: A new broadcasting scheme for metropolitan video-on-demand systems," in *Proc. ACM Conf. on Applications, Technologies, Architectures and Protocols for Computer Communications (SIGCOMM)*, Cannes, France, Oct. 1997, pp. 89–100.

[HS00] K. A. Hua and S. Sheu, "An efficient periodic broadcast technique for digital video libraries," *ACM Journal of Multimedia Tools and Applications*, vol. 10, no. 2–3, pp. 157–177, 2000.

[HSR98a] E. Horowitz, S. Sahni, and S. Rajasekaran, "Computer algorithms." New York: Computer Science Press, 1998, pp. 569–571.

[HSR98b] E. Horowitz, S. Sahni, and S. Rajasekaran, "Computer algorithms." New York: Computer Science Press, 1998, pp. 444–445.

[HSW00] Y.-L. Huang, S.-P. W. Shieh, and J.-C. Wang, "Practical key distribution schemes for channel protection," in *Proc. IEEE Annual Int. Computer Software and Applications Conf. (COMPSAC)*, Taipei, Taiwan, Oct. 2000, pp. 569–574.

[Hu01] A. Hu, "Video-on-demand broadcasting protocols: A comprehensive study," in *Proc. Annual Joint Conf. of the IEEE Computer and Communcations Societies (INFOCOM)*, Anchorage, AK, USA, Apr. 2001, pp. 508–517.

[Iec05] "Voice quality (vq) in converging telephony and ip networks." International Engineering Consortium. 2005. [Online]. Available: http://www.iec.org/online/tutorials/voice_qual/topic05.html

[Inq05] "Microsoft Xbox costs $525 to make — and sells for $399." The Inquirer. Nov. 2005. [Online]. Available: http://www.theinquirer.net/?article=27907

[Itu92] "Conditional-access broadcasting systems," International Telecommunication Union Recommendation ITU-R BT.810, Mar. 1992.

[Itu93a] "Video codec for audiovisual services at p x 64 kbit/s," International Telecommunication Union Recommendation ITU-T H.261, Mar. 1993.

[Itu93b] "Coding of moving pictures and associated audio for digital storage media at up to about 1.5 mbit/s — part 2: Video," International Telecommunication Union/International Organization for Standardization/International Electrotechnical Commision Published Standard ISO/IEC 11 172-2, 1993.

[Itu98] "Packet-based multimedia communications systems," International Telecommunication Union Recommendation ITU-T H.323, Feb. 1998.

[Itu00] "Generic coding of moving pictures and associated audio information — part 2: Video," International Telecommunication Union/International Organization for Standardization/International Electrotechnical Commision Published Standard ISO/IEC 13 818-2, 2000.

[Itu04] "Coding of audio-visual objects — part 2: Visual," International Telecommunication Union/International Organization for Standardization/International Electrotechnical Commision Published Standard ISO/IEC 14 496-2, 2004.

[Itu05a] "Video coding for low bit rate communication," International Telecommunication Union Recommendation ITU-T H.263, Jan. 2005.

[Itu05b] "Coding of audio-visual objects — part 10: Advanced video coding," International Telecommunication Union/International Organization for Standardization/International Electrotechnical Commision Recommendation/Published Standard ITU-T H.264/ISO/IEC 14 496-10 AVC, 2005.

[JL02] M. Johanson and A. Lie. “Layered encoding and transmission of video in heterogeneous environments.” 2002. [Online]. Available: http://www.item.ntnu.no/~arnelie/papers/ITC18-paper_041202.pdf

[JT97a] L.-S. Juhn and L.-M. Tseng, “Harmonic broadcasting for video-on-demand service,” *IEEE Trans. on Broadcasting*, vol. 43, no. 3, pp. 268–271, 1997.

[JT97b] L.-S. Juhn and L.-M. Tseng, “Staircase data broadcasting and receiving scheme for hot video service,” *IEEE Trans. on Consumer Electronics*, vol. 43, no. 4, pp. 1110–1117, 1997.

[JT98] L.-S. Juhn and L.-M. Tseng, “Fast data broadcasting and receiving scheme for popular video service,” *IEEE Trans. on Broadcasting*, vol. 44, no. 1, pp. 100–105, 1998.

[JWX02] R. Janakiraman, M. Waldvogel, and L. Xu, “Fuzzycast: Efficient video-on-demand over multicast,” in *Proc. Annual Joint Conf. of the IEEE Computer and Communcations Societies (INFOCOM)*, New York, NY, USA, June 2002, pp. 920–929.

[KCGT+03] T. Koren, S. Casner, J. Geevarghese, B. Thompson, and P. Ruddy, “Enhanced compressed RTP (CRTP) for links with high delay, packet loss and reordering,” IETF Proposed Standard RFC 3545, July 2003.

[Kle06] A. Klemets, “RTP payload format for video codec 1 (VC-1),” IETF Proposed Standard RFC 4425, Feb. 2006.

[KNFM+00] Y. Kikuchi, T. Nomura, S. Fukunaga, Y. Matsui, and H. Kimata, “RTP payload format for mpeg-4 audio/visual streams,” IETF Proposed Standard RFC 3016, Nov. 2000.

[Knu97] D. E. Knuth, “The art of computer programming,” 3rd ed. Bosten, MA, USA: Addison-Wesley Professional, 1997, vol. 1, pp. 13–18.

[Koo02] P. Koopman, “32-bit cyclic redundancy codes for internet applications,” in *Proc. IEEE Dependable Systems and Networks (DSN)*, Washington, DC, USA, June 2002, pp. 459–472.

[KRB06a] S. Kotz, C. B. Read, and D. L. Banks, Eds., “Encyclopedia of statistical sciences,” 2nd ed. Wiley-Interscience, 2006, vol. 15, pp. 9223–9224.

[KRB06b] S. Kotz, C. B. Read, and D. L. Banks, Eds., “Encyclopedia of statistical sciences,” 2nd ed. Wiley-Interscience, 2006, vol. 9, pp. 6190–6197.

[Kuh99] C. Kuhl, "Serving up a new business — server, set-top price cuts make VOD viable," *CED Magazine*, no. 5, 1999. [Online]. Available: http://www.cedmagazine.com/ced/9905/9905bb1.htm

[Kuh02] C. Kuhl, "VOD's bottom-line improving — new business models require more complicated math," *CED Magazine*, no. 2, 2002. [Online]. Available: http://www.cedmagazine.com/ced/2002/0202/02d.htm

[Laz06] J. Lazzaro, "Framing real-time transport protocol (RTP) and RTP control protocol (RTCP) packets over connection-oriented transport," IETF Proposed Standard RFC 4571, July 2006.

[Lee96] W. Lee, "Key distribution and management for conditional access system on dbs," in *Proc. Int. Conf. on the Theory and Applications of Cryptology and Information Security (ASIACRYPT)*, Kyongju, Korea, Nov. 1996, pp. 82–86.

[Lee99] J. Y. B. Lee, "UVoD: A unified architecture for video-on-demand services," *IEEE Communication Letters*, vol. 3, pp. 277–279, 1999.

[LHF05] B. Link, T. Hager, and J. Flaks, "RTP payload format for AC-3 audio," IETF Proposed Standard RFC 4184, Oct. 2005.

[LKCC02] S.-W. Lee, K.-W. Kim, J.-M. Choi, and K.-D. Chung, "An alternative multicast transmission scheme for VCR operations in a large VOD system," in *Proc. IEEE Int. Conf. on Information Networking (ICOIN)*, Cheju Island, Korea, Jan. 2002, pp. 479–490.

[LL00] V. C. H. Lee and J. Y. B. Lee, "Improving UVoD system efficiency with batching," in *Proc. Int. Conf. on Software, Telecommunications and Computer Networks (SoftCOM)*, Venice, Oct. 2000.

[LLG97] M. Y. Y. Leung, J. C. S. Lui, and L. Golubchik, "Buffer and I/O resource pre-allocation for implementing batching and buffering techniques for video-on-demand systems," in *Proc. IEEE Int. Conf. on Data Engineering (ICDE)*, Birmingham, UK, Apr. 1997, pp. 344–353.

[LLG98] S.-W. Lau, J. C. S. Lui, and L. Golubchik, "Merging video streams in a multimedia storage server: Complexity and heuristics," *ACM Multimedia Systems Journal*, vol. 6, no. 1, pp. 29–42, 1998.

[LMS98a] M. Luby, M. Mitzenmacher, and A. Shokrollahi, "Analysis of random processes via and-or tree evaluation," in *Proc. ACM/SIAM Symposium on Discrete Algorithms (SODA)*, San Francisco, CA, USA, Jan. 1998, pp. 364–373.

[LMS98b] M. Luby, M. Mitzenmacher, and A. Shokrollahi, "Analysis of random processes via and-or tree evaluation," International Computer Science Institute, University of Berkeley, Berkeley, CA, USA, Tech. Rep. TR-97-042, 1998.

[LMSS+97] M. G. Luby, M. Mitzenmacher, M. A. Shokrollahi, D. A. Spielman, and V. Stemann, "Practical loss-resilient codes," in *Proc. Annual ACM Symposium on Theory of Computing (STOC)*, El Paso, TX, USA, May 1997, pp. 150–159.

[LMSS01] M. G. Luby, M. Mitzenmacher, M. A. Shokrollahi, and D. A. Spielman, "Efficient erasure correcting codes," *IEEE Trans. on Information Theory*, vol. 47, no. 2, pp. 569–584, 2001.

[LN99] F. Li and I. Nikolaidis, "Trace-adaptive fragmentation for periodic broadcast of VBR video," in *Proc. ACM Int. Workshop on Network and Operation System Support for Digital Audio and Video (NOSSDAV)*, Basking Ridge, NJ, USA, June 1999, pp. 253–264.

[LN00] F. Li and I. Nikolaidis, "SCB: StairCase Broadcast for media-on-demand systems," in *Proc. on Int. Workshop on Mobile Multimedia Communications (MoMuC)*, Tokyo, Japan, June 2000, pp. 116–117.

[loo01] "Video-on-demand financing snapshot," *Cable World*, Aug. 2001. [Online]. Available: http://www.findarticles.com/p/articles/mi_m0DIZ/is_33_13/ai_77434123

[LT95] J. Lauderdale and D. H. K. Tsang, "Bandwidth scheduling of prerecorded VBR video sources for ATM networks," in *Proc. IEEE ATM Workshop*, Washington, DC, USA, Oct. 1995.

[LV93] T. D. C. Little and D. Venkatesh, "Prospects for interactive video-on-demand," *IEEE Multimedia*, vol. 1, no. 3, pp. 14–24, 1993.

[Man05] J. Mandese, "Nielsen to TV channels: If we don't monitor you, you don't exist," *MediaPost Communications*, May 2005. [Online]. Available: http://www.cofc.edu/%7Eferguson/bcp/updates/chap2/If%20We%20Don't%20Monitor%20You,%20You%20Don't%20Exist.htm

[MEVSS01] A. Mahanti, D. L. Eager, M. K. Vernon, and D. J. Sundaram-Stukel, "Scalable on-demand media streaming with packet loss recovery," in *Proc. ACM Conf. on Applications, Technologies, Architectures and Protocols for Computer Communications (SIGCOMM)*, San Diego, CA, USA, Aug. 2001, pp. 97–108.

[Mil92] D. Mills, "Network time protocol (version 3) specification, implementation and analysis," IETF Draft Standard RFC 1305, Mar. 1992.

[MJV96] S. McCanne, V. Jacobson, and M. Vetterli, "Receiver-driven layered multicast," in *Proc. ACM Conf. on Applications, Technologies, Architectures and Protocols for Computer Communications (SIGCOMM)*, Stanford, CA, USA, Aug. 1996, pp. 117–130.

[ML05] A. Mayer and H. Linder, "A survey of adaptive layered video multicast using MPEG-2 streams," in *Proc. IST Mobile & Wireless Communications Summit*, Dresden, Germany, Dec. 2005.

[Moo05] S. M. Moore, "Pushing the internet video-on-demand envelope," *Entertainment Law Reporter*, Sept. 2005. [Online]. Available: http://www.stroock.com/SiteFiles/Pub373.pdf

[MP04] D. Micciancio and S. Panjwani, "Optimal communication complexity of generic multicast key distribution," in *Proc. Eurocrypt Conf.*, Interlaken, Switzerland, May 2004, pp. 153–170.

[MQ95] B. M. Macq and J.-J. Quisquater, "Cryptology for digital TV broadcasting," *Proc. of the IEEE*, vol. 83, no. 6, pp. 944–957, 1995.

[MS99] A. Malis and W. Simpson, "PPP over SONET/SDH," IETF Proposed Standard RFC 3545, June 1999.

[MS02] H. Ma and K. G. Shin, "Multicast video-on-demand services," *ACM SIGCOMM Computer Communication Review*, vol. 32, no. 1, pp. 31–43, 2002.

[NBT98] J. Nonnenmacher, E. W. Biersack, and D. Towsley, "Parity-based loss recovery for reliable multicast transmission," *IEEE/ACM Trans. on Networking*, vol. 6, no. 4, pp. 349–361, 1998.

[NOBB05] B. Nikolaus, J. Ott, C. Bormann, and U. Bormann, "Generalized greedy broadcasting for efficient media-on-demand transmissions," *IEEE Trans. on Broadcasting*, vol. 51, no. 3, pp. 354–359, 2005.

[NOBB06] B. Nikolaus, J. Ott, C. Bormann, and U. Bormann, "Reducing bandwidth consumption at startup of media transmissions," *IEEE Trans. on Broadcasting*, vol. 52, no. 3, pp. 368–373, 2006.

[OBSW+07] J. Ott, C. Bormann, G. Sullivan, S. Wenger, and R. E. (Ed.), "RTP payload format ITU-T Rec. H.263 video," IETF Proposed Standard RFC 2429, Jan. 2007.

[OC08] J. Ott and E. Carrara, "Extended secure RTP profile for real-time transport control protocol (RTCP)-based feedback (RTP/SAVPF)," IETF Standards Track RFC 5124, Feb. 2008.

[Oft03] "Trade fund report." Office of the Telecommunications Authority, Hong Kong. 2003. [Online]. Available: http://www.ofta.gov.hk/en/trade-fund-report/0203/html/eng/02.htm

[OWSB+06] J. Ott, S. Wenger, N. Sato, C. Burmeister, and J. Rey, "Extended RTP profile for real-time transport control protocol (RTCP)-based feedback (RTP/AVPF)," IETF Proposed Standard RFC 4585, July 2006.

[PB61] W. W. Peterson and D. T. Brown, "Cyclic codes for error detection," *Proc. IEEE Institute of Radio Engineers (IRE)*, vol. 49, no. 1, pp. 228–235, 1961.

[PCL98a] J.-F. Pâris, S. R. Carter, and D. D. E. Long, "Bandwidth allocation issues in near video on demand services," in *Proc. on Multimedia Technology and Applications Conf. (MTAC)*, Anaheim, CA, USA, Sept. 1998, pp. 196–200.

[PCL98b] J.-F. Pâris, S. W. Carter, and D. D. E. Long, "Efficient broadcasting protocols for video on demand," in *Proc. Int. Symposium on Modeling, Analysis, and Simulation of Computer Systems (MASCOTS)*, Montreal, Canada, July 1998, pp. 127–132.

[PCL98c] J.-F. Pâris, S. W. Carter, and D. D. E. Long, "A low bandwidth broadcasting protocol for video on demand," in *Proc. IEEE Int. Conf. on Computer Communications and Networks (ICCCN)*, Lafayette, LA, USA, Oct. 1998, pp. 690–697.

[PCL99a] J.-F. Pâris, S. W. Carter, and D. D. E. Long, "Combining pay-per-view and video-on-demand services," in *Proc. Int. Symposium on Modeling, Analysis, and Simulation of Computer Systems (MASCOTS)*, College Park, MD, USA, Oct. 1999, pp. 270–276.

[PCL99b] J.-F. Pâris, S. W. Carter, and D. D. E. Long, "A hybrid broadcasting protocol for video on demand," in *Proc. Conf. on Multimedia Computing and Networking (MMCN)*, San Jose, CA, USA, Jan. 1999, pp. 317–326.

[PCL00a] J.-F. Pâris, S. W. Carter, and D. D. E. Long, "A reactive broadcasting protocol for video on demand," in *Proc. Conf. on Multimedia Computing and Networking (MMCN)*, San Jose, CA, USA, Jan. 2000, pp. 216–223.

[PCL00b] J.-F. Pâris, S. W. Carter, and D. D. E. Long, "A universal distribution protocol for video-on-demand," in *Proc. IEEE Int. Conf. on Multimedia and Expo (ICME)*, New York City, NY, USA, July 2000, pp. 49–52.

[PD96] J.-L. G. P. Deutsch, "ZLIB compressed data format specification 3.3," IETF Informational RFC 1950, May 1996.

[PL00] J.-F. Pâris and D. D. E. Long, "Limiting the client bandwidth of broadcasting protocols for videos on demand," in *Proc. Euromedia Conf.*, Antwerp, Belgium, May 2000, pp. 107–111.

[PL01] J.-F. Pâris and D. D. E. Long, "The case for aggressive partial preloading in broadcasting protocols for video-on-demand," in *Proc. IEEE Int. Conf. on Multimedia and Expo (ICME)*, Tokyo, Japan, Aug. 2001, pp. 113–116.

[PL03] J.-F. Pâris and D. D. E. Long, "A variable bandwidth broadcasting protocol for video-on-demand," in *Proc. Conf. on Multimedia Computing and Networking (MMCN)*, Santa Clara, CA, USA, Jan. 2003, pp. 209–219.

[PLE03] A. Papagiannis, D. Lioupis, and S. Egglezos, "A scalable, low-cost VoD server with multicast support," in *Proc. Int. Conf. on. Information Technology, Research and Education (ITRE)*, Newark, NJ, USA, Aug. 2003, pp. 464–468.

[PLM99] J.-F. Pâris, D. D. E. Long, and P. E. Mantey, "Zero-delay broadcasting protocols for video-on-demand," in *Proc. Annual ACM Int. Conf. on Multimedia (ACM MM)*, Orlando, FL, USA, Oct./Nov. 1999, pp. 189–197.

[Poi04] "Video on demand," *Broadband Money Makers*, no. 5, 2004. [Online]. Available: http://www.point-topic.com/content/bmm/profiles/video+on+demand.htm

[Pos80] J. Postel, "User datagram protocol," IETF Standard RFC 0768/STD 0006, Aug. 1980.

[Pos81] J. Postel, "Transmission control protocol," IETF Standard RFC 0793/STD 0007, Sept. 1981.

[PQ02] G. Poirier and C.-G. Quimper, "Harmonic broadcasting: Improving optimal bandwidth requirements by introducing advertisements," University of Waterloo, Waterloo, Ontario, Canada, Tech. Rep. CS 860, 2002.

[Pâr99a] J.-F. Pâris, "A broadcasting protocol for compressed video," in *Proc. Euromedia Conf.*, Munich, Germany, Apr. 1999, pp. 78–84.

[Pâr99b] J.-F. Pâris, "A simple low-bandwidth broadcasting protocol for video on demand," in *Proc. IEEE Int. Conf. on Computer Communications and Networks (ICCCN)*, Boston-Natick, MA, USA, Oct. 1999, pp. 118–123.

[Pâr01a] J.-F. Pâris, "A fixed-delay broadcasting protocol for video-on-demand," in *Proc. IEEE Int. Conf. on Computer Communications and Networks (ICCCN)*, Scottsdale, AZ, USA, Oct. 2001, pp. 418–423.

[Pâr01b] J.-F. Pâris, "An interactive broadcasting protocol for video-on-demand," in *Proc. IEEE Int. Performance, Computing and Communication Conf. (IPCCC)*, Phoenix, AZ, USA, Apr. 2001, pp. 347–353.

[Pâr01c] J.-F. Pâris, "A stream tapping protocol with partial preloading," in *Proc. Int. Symposium on Modeling, Analysis, and Simulation of Computer Systems (MASCOTS)*, Cincinnati, OH, USA, Aug. 2001, pp. 423–430.

[Pâr02] J.-F. Pâris, "A broadcasting protocol for video-on-demand using optional partial preloading," in *Proc. Int. Conf. on Communications (ICC)*, Mexico City, Mexico, Nov. 2002, pp. 319–329.

[Rig98] W. Riggert, "ATM — Technik und Einführung." Kaarst, Germany: bhv Verlags GmbH, 1998, pp. 130–137.

[Riv92] R. Rivest, "The MD5 message-digest algorithm," IETF Informational RFC 1321, Apr. 1992.

[Riv98] R. L. Rivest, "Chaffing and winnowing: Confidentiality without encryption," *RSA Laboratories CryptoBytes Journal*, vol. 4, no. 1, pp. 12–17, 1998.

[Riz97] L. Rizzo, "Effective erasure codes for reliable computer communication protocols," *ACM SIGCOMM Computer Communication Review*, vol. 27, no. 2, pp. 24–36, 1997.

[RR99] S. G. Rao and S. V. Raghavan, "Fast techniques for the optimal smoothing of stored video," *ACM Multimedia Systems Journal*, vol. 7, no. 3, pp. 222–233, 1999.

[RSCJ+02] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, "SIP: Session Initiation Protocol," IETF Proposed Standard RFC 3261, June 2002.

[RSU01] T. J. Richardson, M. A. Shokrollahi, and R. L. Urbanke, "Design of capacity-approaching irregular low-density parity-check codes," *IEEE Trans. on Information Theory*, vol. 47, no. 2, pp. 619–637, 2001.

[San03] R. Santitoro. "Metro Ethernet services — A technical overview." Metro Ethernet Forum. 2003. [Online]. Available: http://metroethernetforum.org/PDF_Documents/metro-ethernet-services.pdf

[SC03] H. Schulzrinne and S. Casner, "RTP profile for audio and video conferences with minimal control," IETF Standard RFC 3551/STD 0065, July 2003.

[SCFJ03] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: A transport protocol for real-time applications," IETF Standard RFC 3550/STD 0064, July 2003.

[Sch96] B. Schneier, "Applied cryptography," 2nd ed. New York/Chichester/Brisbane/Toronto/Singapore: John Wiley & Sons, Inc., 1996, ch. 22.1, pp. 513–516.

[SGRT99] S. Sen, L. Gao, J. Rexford, and D. Towslew, "Optimal patching schemes for efficient multimedia streaming," in *Proc. ACM Int. Workshop on Network and Operation System Support for Digital Audio and Video (NOSSDAV)*, Basking Ridge, NJ, USA, June 1999, pp. 455–463.

[SHH97] S. Sheu, K. A. Hua, and T.-H. Hu, "Virtual batching: A new scheduling technique for video-on-demand servers," in *Proc. IEEE Int. Conf. on Database Systems for Advanced Applications (DASFAA)*, Melbourne, Australia, Apr. 1997, pp. 481–490.

[SHT97] S. Sheu, K. A. Hua, and W. Tavanapong, "Chaining: A generalized batching technique for video-on-demand systems," in *Proc. IEEE Int. Conf. on Multimedia Computing and Systems (ICMCS)*, Ottawa, Canada, June 1997, pp. 110–117.

[Sig06] "EM8620L series — digital media processors with MPEG-4.10 (H.264) and VC-1." Sigma Designs. 2006. [Online]. Available: http://www.sigmadesigns.com/products/em8620Lseries.htm

[Sin90] W. D. Sincoskie, "Video on demand: Is it feasible?" in *Proc. IEEE Global Telecommunications Conf. (GLOBECOM)*, San. Diego, CA, USA, Dec. 1990, pp. 201–205.

[SmNs04] R. Steinmetz and K. Nahrstedt, "Multimedia systems." Springer Berlin/Heidelberg, 2004, pp. 283–285.

[Soc06] "Television — VC-1 compressed video bitstream format and decoding process," Society of Motion Picture and Television Engineers Recommendation/Published Standard SMPTE 421M, 2006.

[SRL98] H. Schulzrinne, A. Rao, and R. Lanphier, "Real time streaming protocol (RTSP)," IETF Proposed Standard RFC 2326, Apr. 1998.

[SRR99] D. Saparilla, K. W. Ross, and M. Reisslein, "Periodic broadcasting with VBR-encoded video," in *Proc. Annual Joint Conf. of the IEEE Computer and Communcations Societies (INFOCOM)*, New York, NY, USA, Mar. 1999, pp. 464–471.

[SRR04] D. Saparilla, K. W. Ross, and M. Reisslein, "Periodic broadcasting with VBR-encoded video," *ACM Multimedia Systems Journal*, vol. 9, no. 6, pp. 503–516, 2004.

[SZKT96] J. D. Salehi, Z.-L. Zhang, J. F. Kurose, and D. F. Towsley, "Supporting stored video: Reducing rate variability and end-to-end resource requirements through optimal smoothing," in *Proc. ACM Int. Conf. on Measurement and Modeling of Computer Systems (SIGMETRICS)*, Philadelphia, PA, USA, May 1996, pp. 222–231.

[Tan81] A. S. Tanenbaum, "Computer networks." Prentice Hall, 1981, pp. 406–407.

[TCS04] Y.-C. Tseng, Y.-C. Chueh, and J.-P. Sheu, "Seamless channel transition for the staircase video broadcasting scheme," *IEEE/ACM Trans. on Networking*, vol. 12, no. 3, pp. 559–571, 2004.

[THS02] M. A. Tantaoui, K. A. Hua, and S. Sheu, "A scalable technique for VCR-like interactions in video-on-demand applications," in *Proc. IEEE Int. Conf. on Distributed Computing Systems (ICDCS)*, Vienna, Austria, July 2002, pp. 246–251.

[THYL+01] Y.-C. Tseng, C.-M. Hsieh, M.-H. Yang, W.-H. Liao, and J.-P. Sheu, "Data broadcasting and seamless channel transition for highly-demanded videos," *IEEE Trans. on Communications*, vol. 49, no. 5, pp. 863–874, 2001.

[Tvh05] "Television facts and statistics — 1939 to 2000." TVhistory.TV. 2005. [Online]. Available: http://www.tvhistory.tv/facts-stats.htm

[TYC02] Y.-C. Tseng, M.-H. Yang, and C.-H. Chang, "A recursive frequency-splitting scheme for broadcasting hot videos in VoD service," *IEEE Trans. on Communications*, vol. 50, no. 8, pp. 1348–1355, 2002.

[Usc05a] "Statistical abstract of the United States." U.S. Census Bureau. 2005. Section 13: Information and Communications 2004–2005, Table 1120. [Online]. Available: http://www.census.gov/prod/www/statistical-abstract-04.html

[Usc05b] "Stistical abstract of the United States." U.S. Census Bureau. 2005. Section 13: Information and Communications 2004–2005, Table 1119. [Online]. Available: http://www.census.gov/prod/www/statistical-abstract-04.html

[vdMMSS+03] J. van der Meer, D. Mackie, V. Swaminathan, D. Singer, and P. Gentric, "RTP payload format for transport MPEG-4 elementary streams," IETF Proposed Standard RFC 3640, Nov. 2003.

[vdW91] B. L. van der Waerden, "Algebra I," 9th ed. Berlin/Heidelberg/New York: Springer-Verlag Berlin Heidelberg New York, 1991, pp. 131–134.

[VI96] S. Viswanathan and T. Imielinski, "Metropolitan area video-on-demand service using pyramid broadcasting," *ACM Multimedia Systems Journal*, vol. 4, no. 4, pp. 197–208, 1996.

[Wal01a] G. Walz, Ed., "Lexikon der Mathematik." Heidelberg/Berlin, Germany: Spektrum Akademischer Verlag, 2001, vol. 2, p. 96.

[Wal01b] G. Walz, Ed., "Lexikon der Mathematik." Heidelberg/Berlin, Germany: Spektrum Akademischer Verlag, 2001, vol. 2, p. 99.

[Wei06] E. W. Weisstein. "Digamma function." MathWorld — A Wolfram Web Resource. 1999–2006. [Online]. Available: http://mathworld.wolfram.com/DigammaFunction.html

[WGL00] C. K. Wong, M. Gouda, and S. S. Lam, "Secure group communications using key graphs," *IEEE/ACM Trans. on Networking*, vol. 8, no. 1, pp. 16–30, 2000.

[WHA99a] D. Wallner, E. Harder, and R. Agee, "Key management for multicast: Issues and architectures," IETF Informational RFC 2627, June 1999.

[WHA99b] D. M. Wallner, E. G. Harder, and R. C. Agee, "Key management for multicast: issues and architecture," IETF Informational RFC 2627, June 1999.

[WHSW+05] S. Wenger, M. M. Hannuksela, T. Stockhammer, W. Westerlund, and D. Singer, "RTP payload format for H.264 video," IETF Proposed Standard RFC 3984, Feb. 2005.

[Wik08a] "Radio." Wikipedia. 2001–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Radio\&oldid=207856377

[Wik08b] "Streaming media." Wikipedia. 2001–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Streaming_media\&oldid=206816320

[Wik08c] "Television." Wikipedia. 2001–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Television\&oldid=207861161

[Wik08d] "Poisson distribution." Wikipedia. 2002–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Poisson_distribution\&oldid=207960883

[Wik08e] "Video on demand." Wikipedia. 2002–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Video_on_demand\&oldid=203722452

[Wik08f] "Zipf's law." Wikipedia. 2002–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Zipf%27s_law\&oldid=206292280

[Wik08g] "Bin packing problem." Wikipedia. 2003–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Bin_packing_problem&oldid=192304336

[Wik08h] "Interactive television." Wikipedia. 2003–2008. [Online]. Available: http://en.wikipedia.org/w/index.php?title=Interactive_television\&oldid=207310957

[WMS06] M.-Y. Wu, S. Ma, and W. Shu, "Scheduled video delivery — a scalable on-demand video delivery scheme," *IEEE Trans. on Multimedia*, vol. 8, no. 1, pp. 179–187, 2006.

[Wol04] L. Wolf, "Multimedia media-on-demand/interactive TV," Slides for Course on Multimedia, University of Technology in Braunschweig, 2004.

[Xbo07] "Microsoft integrates IPTV software platform with Xbox 360." Xbox. Jan. 2007. [Online]. Available: http://www.xbox.com/en-us/community/events/ces2007/microsoftintegratesiptvsoftwareplatform.htm

[Xu03] L. Xu, "Resource-efficient delivery of on-demand streaming data using UEP codes," *IEEE Trans. on Communications*, vol. 51, no. 1, pp. 63–71, 2003.

[YCTY+01] Z.-Y. Yang, Y.-M. Chen, L.-M. Tseng, H.-F. Yu, and C.-D. Won, "Data stream broadcasting with live program support," in *Proc. Taiwan Academic Network Conf. (TANet)*, Chiayi, Taiwan, Oct. 2001.

[YK03] E. M. Yan and T. Kameda, "An efficient VOD broadcasting scheme with user bandwidth limit," in *Proc. Conf. on Multimedia Computing and Networking (MMCN)*, Santa Clara, CA, USA, Jan. 2003, pp. 200–208.

[YP01] P.-F. You and J.-F. Pâris, "A better dynamic broadcasting protocol for video-on-demand," in *Proc. IEEE Int. Performance, Computing and Communication Conf. (IPCCC)*, Phoenix, AZ, USA, Apr. 2001, pp. 84–89.

[YYT03] H.-C. Yang, H.-F. Yu, and L.-M. Tseng, "Adaptive live broadcasting for highly-demand videos," *IIS Journal of Information Science and Engineering*, vol. 19, no. 3, pp. 531–549, 2003.

[ZP02] Q. Zhang and J.-F. Pâris, "A channel-based heuristic distribution protocol for video-on-demand," in *Proc. IEEE Int. Conf. on Multimedia and Expo (ICME)*, Lausanne, Switzerland, Aug. 2002, pp. 245–248.