MŰEGYETEM 1782

Budapest University of Technology and Economics

Enhanced Methods for IPTV Delivery over Wireless Networks

Thesis Booklet

by

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May 2014

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Version 1.0

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This paper meets the requirements of
ANSI/NISO z39.48-1992 (Permanence of Paper).

Abstract

The rapid development of the wireless access technologies challenged the IPTV platform operators to exploit this new medium, although the IPTV systems were designed for fixed transport, and they cannot easily accommodate to the mobile environment. Two aspect of this challenge are analyzed in my dissertation: the bandwidth allocation and packet retransmission; and the charging solutions.

First, a Markovian channel model is created, which is able to capture the specific requirements of the multimedia transmission in wireless networks, then new methods are developed for bandwidth allocation and packet retransmission for ensuring optimal user experience. The second part introduces a novel quality based charging architecture for IPTV solutions, and provides the necessary considerations to implement this method for mobile networks. All the results are evaluated on theoretical basis, which was supported by measurements conducted in two testbeds.

The results increase the service quality and mitigate the negative effect of wireless transport by providing new functions and design processes for IPTV solutions. The quality based charging ensures a fair and simple billing method with the aim of gain customer satisfaction.

1 Introduction

Since Swinton (1908) described its theory of “Distant electric vision” in Nature, television and—as a consequence—ourselves changed dramatically. Today, television plays an important role in the day-by-day life, the broadcast industry shapes our knowledge, desire, expectation, and vision of future.

The Internet era, as a major driving force in the last decade, triggered an evolution in the telecommunication industry. In spite of the increasing share of fixed broadband connections, the revenue of the fixed line retail services continues to decline. To preserve business profitability, the legacy telecommunication industry extended its portfolio with various value adding services, like rich communication, mobile payment, and triple-play services to attract customers. This successful strategy resulted a higher revenue, but the growing market of the new generation over-the-top (OTT) services challenged the competitiveness of these triple-play services.

Google recently introduced its interactive TV platform, which combines the television and web user experiences. Microsoft positioned the XBox One console as an equivalent gaming and home entertainment device. Apple announced that HBO Go and WatchESPN come to AppleTV and extended the features of its successful iTunes service. Amazon released an iPad application to access its on-demand video portal, and entered into contracts with several content owners to enlarge the content base of its video services. It can be clearly seen that there is a big race for the customers and the telekom sector has to assess these threats and opportunities from OTT players with enhanced television and video services.

The Global IPTV Forecasts report estimates that the number of subscribers, paying for IPTV service in 97 countries, will double in the next 5 years. This means a 15% yearly growth rate between 2013 and 2018, which will be resulted by the new IPTV deployments mainly in Asia and by the increasing penetration of IPTV again the traditional broadcast technologies. From financial point of view, the revenue will shoot up to \$21 billion from \$12.0 billion in 2018.

Recognizing this importance and business potential, I decided to choose the *enhanced methods of IPTV delivery* for the topic of my research. I investigate and develop new solutions for service quality improvements and competitive pricing methods for IPTV systems.

2 Research Objectives

The IPTV service providers are interested by the maximization of their customer reach, but in many cases, long digital subscriber liness (DSLs) offer inadequate bandwidth for high definition services (Open IPTV Forum 2011). The network providers need to find a solution that enables them to utilize their current infrastructure. The answers may include the implementation of a more advanced encoding algorithm (H.264, H.265), which results in having the same quality on smaller throughput (Abramowski 2011); the introduction of a hybrid service, which replaces the most bandwidth consuming scheduled content service transport with digital video broadcasting (DVB-X) technology (Benoit 2008); the usage of progressive download; or—as I point out in my work—the implementation of a more effective bandwidth allocation in the access network, which will ensure a more efficient transport.

Besides optimizing delivery, IPTV service providers must face the new problem of mobile entertainment devices. Customers access content not only on STBs, but also from various hand-held devices. Today, the prime wireless technology within the consumer domain is the wireless local area network (WLAN), therefore operators have to adapt their IPTV solutions to support the specific requirements of this communication channel. In addition to that, the rapid evolution of the WLAN technology enables set-top-box (STB) manufacturers to build a wireless interface into the devices (Sanchez et al. 2010), which creates value for customers by saving the effort of in-house cabling or the purchase of expensive powerline adapters.

Today, the core transport technology of the IPTV service is not sufficient for wireless delivery, it is not able to effectively handle the special requirements and attributes of the mixed, DSL and WLAN environment (Jursonovics and Imre 2013). As conclusion, I formulate my research question on the following way:

> How can the quality of the traditional xDSL delivery of IPTV services, extended over WLAN networks, be enhanced to ensure excellent service delivery?

If IPTV service providers could improve the key factors and processes, which influences the IPTV delivery in a environment of mixed access technologies (xDSL and WLAN), they could increase the service quality, and—therefore—the customer satisfaction.

First, I claim that the traditional IPTV solutions does not address both the limited resource environment of the xDSL technology and the specific requirements of the wireless channels at the same time. Second, quality is determined by the bandwidth allocation, which is limited by the three aspects of packet retransmission: *general*, *intra-burst*, and *inter-burst*. Third, the packet loss and packet retransmission can be described by stochastic processes, which I unite in one, single model of *three state channem model (3SCM)*. Fourth, I define the optimal bandwidth allocation, which will ensure the maximal quality of service. Finally, I describe a new retransmission algorithm, which improves the quality of the IPTV delivery in wireless networks.

In relation to the development of wireless home access technologies, evolution of the mobile telecommunication technology has accelerated in the last decade. The legacy UMTS and general packet radio service (GPRS) solutions allowed only a limited range of services, for instance e-mail or WEB based Internet surfing, but with the new LTE technology, mobile customers can access the Internet faster than ever. The 4th generation mobile system ensures the option to offer various value added services, like IP telephony, cloud computing, peer-to-peer sharing, or even high definition IPTV delivery over wireless networks (Olariu et al. 2005; Bataa et al. 2012; Wolfinger 2012).

Though the radio transmission and solution core become state-of-the-art, but the latest research of the mobile charging solutions have not addressed the special requirements of the multimedia delivery over mobile access; most of them still implement a time- or volume-based accounting method. I am going to show that the charging solutions shall consider the quality of the user experience in IPTV services offered by mobile networks, because the visual quality of a streaming media depends on complex network conditions, which has to be incorporated to the charging process. In other words, a small, but specific degradation could jeopardize the user experience of the media delivery, which would led to unfair customer service charge.

Moreover, if customers access streaming services from third party platform providers then the charging of the service is handled externally to the network operator. The inter-company charging of the mobile service has to reflect the quality of the offered multimedia service, where the implementation of a single volume based method could not provide an accurate basis.

I concentrated these ideas in my second research question:

> How can the charging of the IPTV services reflect the user experience to increase customer satisfaction?

The answer will directly result a fair pricing and billing scheme for IPTV delivery

The second part of my dissertation challenges the charging methods in wireless networks, and discusses the introduction of a novel quality based charging solution. First, I claim that the traditional charging methods are not sufficient for providing a quality of experience based charging. Second, I select the network operator centric business model, and I create a charging architecture according to the 3GPP's recommendation. Third, I objectively define the quality of the streaming. Fourth, I point out the dependence of the loss attributes and the quality. Fifth, I define a sufficient charging policy based on the quality and loss pattern. Finally, I conclude all the methods in one, overall quality based charging method.

3 Research Methodology

The evaluation of the quality of a video codec or a quality degradation, caused by a frame loss, requires mathematical methods, which are able to provide an objective basis for comparison. To achieve this goal, several reference based image quality metrics have been developed, the peak signal noise ratio (PSNR) is one of the well known ones (Huynh-Thu and Ghanbari 2008):

$$PSNR = 10 \log \left(\frac{MAX^2}{MSE} \right), \tag{1}$$

where MSE is the mean square error of the frame pixels.

I have considered other metrics as well, but I concluded that I adapt the PSNR, because it is widely used and several research results are documented with this metric. This ensures the compatibility and easy validation of my own results.

Furthermore, I applied a statistical approach in several theses to evaluate and conclude my new findings, which approach required a closed form expression of the probability of several specific loss characteristics. To obtain these parameters, I used a Monte Carol analysis to confirm my conjecture on a convergence, which I extended the description of the analytical proof as well.

The empirical evaluation of my work requires sample video sequences, encoding and decoding functionalities. I decided, that I will consecutively use the video sources introduced by Seeling and Reisslein (2012), because they are well known and widely used in several research article. This approach enables the easy validation of my own findings

To evaluate the video quality deterioration, caused by frame losses, I use the JM Reference Software (Tourapis et al. n.d.). The 18.4 decoder version has correctly implemented the error-concealment feature, which is crucial for mitigating the effect of non error free transmission, and my findings rely on this feature.

I carefully considered the realization of an NS2 simulation (Mccanne, Floyd, and Fall n.d.), but I found that the required efforts to implement such a complex element like a retransmission function would be to great compared to the easy setup of a testbed.

I did not conclude any disadvantages of implementing a testbed environment, therefore I followed the OIPF System Architecture (Open IPTV Forum 2011), and implemented the following OIPF functions shoved by figure 1, with the aim of evaluating my model and methods in the first part of my dissertation.

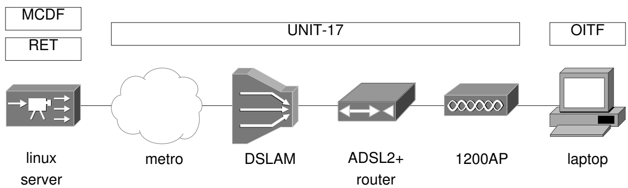

Figure 1. Testbed for IPTV retransmission simulation

Multicast content delivery function (MCDF). ser.cpp, my c++ application, a simple H.264 packetizer, generates UDP/RTP multicast traffic based on the RFC 6184 (Wang et al. 2011), and implements a simple control protocol. Hosted by an x86 Linux server connected to the core network of Deutsche Telekom (DT).

Retransmission server (RET). My own c++ application (ret.cpp), captures the multicast traffic in a circular buffer, and implements a simple retransmission request protocol. Hosted by the same x86 Linux server.

Unit–17 interface, part a. Core metro network of DT.

Unit–17 interface, part b. DSLAM, digital subscriber line access multiplexer.

Unit–17 interface, part c. Realized by the ADSL2+ access network of DT, provided by a Speedport 720w ADSL2+ modem.

Unit–17 interface, part d. Realized by a 802.11b wireless local area network (WLAN) network, provided by a Cisco 1200 series wireless access point, connected to the asymmetric DSL modem.

Open IPTV terminal function (OITF). cli.cpp, my own c++ application, implements a simple multicast receiver and control functions of the multicast content delivery function and RET Server. Hosted on a x86 Linux laptop, connected to the Cisco access point.

I built a second testbed based on the 3GPP charging architecture principles to conduct the measurements required by the second part of my work.

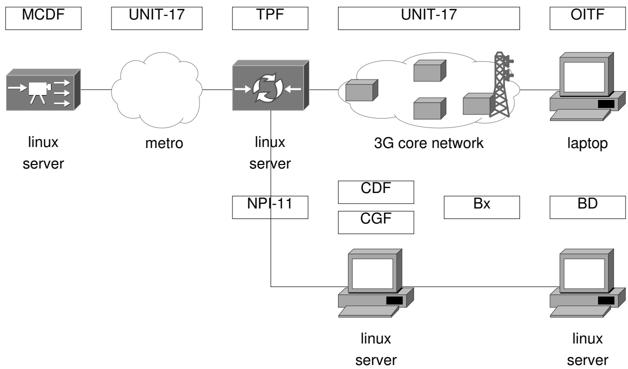

Figure 2. Testbed for quality based charging simulation

Multicast content delivery function (MCDF). application, developed by myself (ser2.cpp), a simple H.264 packetizer, generates UDP/RTP multicast traffic based on the RFC 6184 (Wang et al. 2011), and implements a simple control protocol. Hosted by an x86 Linux server, connected to the core network of DT.

Unit–17 interface, part a. Core metro network of DT.

Transport processing function (TPF). My c++ application, a simple transparent streaming proxy based on the PCAP library. This functional entity includes the functions needed to support real-time multicast and unicast streams, optimizing network usage in the physical network, and enforcing related traffic policies coming from resource and admission control.

Unit–17 interface, part b. 3G mobile network of DT.

OITF. My c++ application (cli2.cpp), a simple UDP/RTP receiver, which implements the basic client functionality, streaming server control protocol, and an internal jitter buffer. The application issues commands for the streaming server, receives the RTP/UDP stream. It continuously analyzes the streaming flow and periodically reports the loss pattern according to the proposed RTSP-RR format to the steaming server. Hosted on a x86 Linux laptop with 4G data card.

NPI–11. Reference point for sending events and charging information. This is the Rf reference point defined by 3GPP (3rd Generation Partnership Project 2010).

Charging data function (CDF). The CDF receives charging events from the CDF via the Rf reference point. It uses the information contained in the charging events to construct a charging data record (CDR).

Charging gateway function (CGF). My own script collection (ana.cpp, mse.cpp hist.cpp, createtrace.sh, decompress.sh). The CDRs produced by the CDF are transferred immediately to the charging gateway function (CGF) via the Ga reference point. The CGF acts as a gateway between the 3GPP network and the billing domain. It uses the Bx reference point for the transfer of CDR files to the BD. It implements a basic charging gateway functionality. It correlates the receiver report files, and based on the metadata description, it estimates the quality of the streaming session (total distortion: $D[\kappa]$), and creates the appropriate charging data records, incorporating the quality estimate.

Bx. The Bx reference point supports interaction between a charging gateway function and the billing domain. The information crossing this reference point is comprised of CDR files. Implemented by a common, standard file transfer protocol, e.g. FTAM (file transfer, access and management) or FTP.

Billing domain (BD). My own c++ application (chg.cpp), implements the quality based charging functions inclusive the quality based policy decision, and creates the bill and discount for the streaming session.

4 New Results

According to my research objectives, the following sections present my theses and theorems in two groups.

THESISGROUP I (presented in 4.1, 4.2, 4.3, 4.4, and 4.5). *Retransmission in IPTV solutions over burst-error channels: I created a new bandwidth allocation method and a new retransmission algorithm, which do provide a better performance for IPTV solutions extended with wireless access, than the traditional methods.*

I also state and prove three theorems in 4.2 to provide a solid mathematical foundation of my claims.

THESISGROUP II (presented in 4.6 and 4.7). *Quality based charging solutions: based on my analysis of charging requirements, I created a novel quality based charging solution, which fulfills the required business models, and provides an user experience based charging and billing functionality.*

Due to the limited size of the thesis booklet, I state only the most important equations and figures in this short document. For a more comprehensive description, please refer to my dissertation (Jursonovics 2014).

4.1 Bandwidth Allocation in IPTV Solutions

The balance, between the assigned bandwidth for AV data and RET service in a typical triple-play service, is crucial for achieving the maximal quality. On one hand, the more throughput is assigned for the AV data, the better stream quality can be achieved by the increase of the encoding bitrate, but the less opportunity is given for error correction. A smooth, sharp stream may be disturbed by blocking or full frame outages due to the insufficient RET throughput. On the other hand, reserving high bandwidth for error correction degrades the overall stream quality due to the low encoding bitrate. A suboptimal ratio of RET bandwidth to AV data may significantly reduce the throughput and—consequently—the quality of the IPTV service.

THESIS I.1 (Jursonovics and Imre 2013). *I claim that the optimal bandwidth allocation prevents frame losses due late packet arrival compared to its playout time. Considering the general, intra- and inter-burst limitations, there is an inverse relationship between the optimal bandwidth allocation and the playout buffer, and a direct relation with the round trip time and average packet size according to (6). I determined this optimal allocation by a minimization problem described in (5).*

The following three equations states these general, intra- and inter-burst limitations in an IPTV solutions respectively.

$$k_{R,general,max}(n) = n\frac{B_{RET}}{B_{AV}}, \tag{2}$$

$$k_{RET,intra,max}\left(\frac{B_{RET}}{B_{AV}}\right) = \frac{D_{playo} - 2D_{pkg} - RTT}{D_{pkg}} \cdot \frac{B_{RET}}{B_{AV}}, \tag{3}$$

$$n_{RET,inter,min}\left(k, \frac{B_{RET}}{B_{AV}}\right) = (k+1)\frac{1}{\frac{B_{RET}}{B_{AV}}} + \frac{RTT - D_{playo}}{D_{pkg}} + k + 2. \tag{4}$$

I claim that to ensure optimal service quality, the allocated retransmission bandwidth has to be ready (non-blocking) for packet retransmission to be able to correct the errors caused by packet losses. To achieve this goal, I present the combined effect of the intra- and inter-burst limitation in figure 3. This graph tells us that at relatively small retransmission bandwidth ($\frac{B_{AV}}{B_{RET}} \approx .05$) only short burst of losses (≈ 10) can be retransmitted, and the allocated retransmission bandwidth will be blocked for a long number of packets (≈ 30). This negative effect annihilates on a slowing rate with the increase of the AV and RET ratio, but above 20% it becomes relative static compared to the intra-burst limitation; the two curves are going to fit each other. According to this effect, there is no extra gain from increasing $\frac{B_{AV}}{B_{RET}}$ above approximately 20-25%, which corresponds to the chosen value of AV and RET bandwidth in commercial IPTV implementations.

This leads to the following minimization problem:

$$\min_{\frac{B_{RET}}{B_{AV}} \in \mathbb{R}^+} \left(n_{RET,inter,min}\left(k_{RET,intra,max}, \frac{B_{RET}}{B_{AV}} \right) \right). \tag{5}$$

In other words, the lowest points on each n_{RET} curves (showed by figure 3) have to be identified, which will represent the optimal bandwidth values. I solve this problem with the first derivative test

$$B_{RET}\big|_{optimum} = B_{AV} \cdot \sqrt{\frac{1}{\frac{D_{playo} - RTT}{D_{pkg}} - 2}}. \tag{6}$$

This equation shows, that there is an inverse relationship between the optimal choice for retransmission bandwidth and the size of the playout

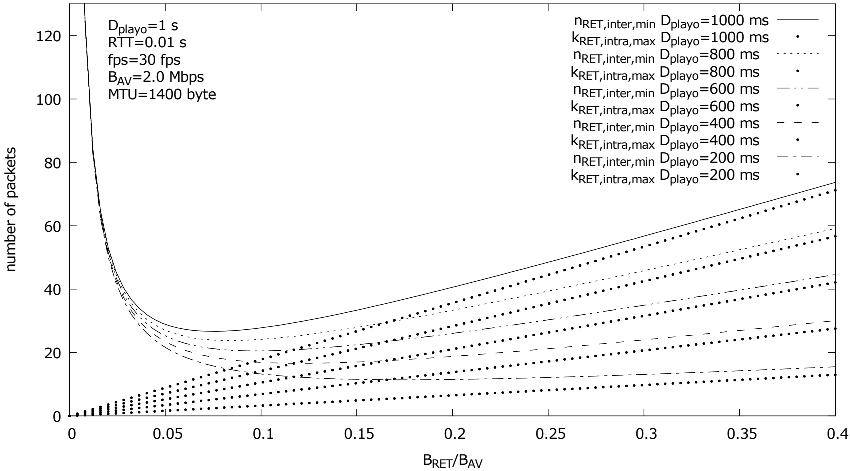

Figure 3. The common effect of the intra- and inter-burst limitation. The $n_{RET,inter,min}$ curves show the number of packets, during which the retransmission bandwidth is blocked (occupied) with retransmission traffic, therefore a new packet loss cannot be corrected. To minimize this negative effect, the optimal values of B_{RET}/B_{AV} should be chosen according to the minimum of the curves.

buffer. Due to its delay compensation role, a longer playout buffer can tolerate the same number of packet losses if smaller bandwidth is available for retransmission services. An increase in the round trip time will also require a higher retransmission bandwidth, because an additional delay in the retransmission handshake will shorten the time window, in which packets can be retransmitted. Convincingly, higher average packet size demands faster retransmission. This result can be used for retransmission bandwidth sizing in IPTV solutions:

4.2 The Three State Channel Model

Let me look at my research question from a different viewpoint. To find the answer, I propose a new model for packet transmission and retransmission for IPTV solutions (Jursonovics and Imre 2011, 2013). This model does address the special features of the wireless channel, and does provide the necessary mathematical instruments for analysis, evaluation, and forecast of

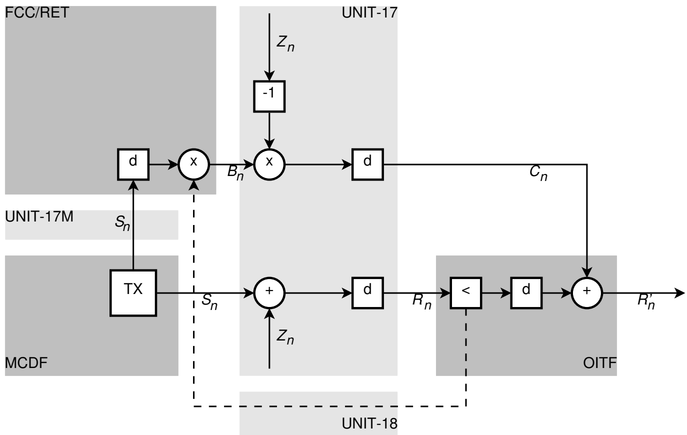

Figure 4. Channel model for IPTV analysis

traffic parameters.

> Thesis I.2 (Jursonovics and Imre 2011, 2013). *I constructed a new mathematical model, the three state channem model (3SCM) for the description of the retransmission effect on bursty packet losses in wireless network for IPTV solution. My model unites the description of packet loss and retransmission, and I proposed the method of forecasting loss and retransmission probabilities.*

Let $S_n = \{0, \forall n\}$ represent the number of sent packets at the time n in figure 4, MCDF the Multicast Content Delivery Function, $Z_n = \{0; 1, \forall n\}$ an additive noise in the Unit-17 interface, d the transmission delay, and OITF the open IPTV terminal function.

I assume that the receiver (represented by $<$) detects a packet loss by checking the sequence numbers of packets, and requests the retransmission through the Unit-18 interface: $B_n = \{1,$ for packet retransmission; 0 otherwise$\}$ of every lost packets only once from the FCC/RET Server. I also assume that this communication is protected by an error-free protocol, like TCP, and the transmission delay on Unit-18 is negligible less than on Unit-17 due to the small size of the retransmission requests.

I showed that a Markov chain can be constructed to describe the following events: let $X_n = 0$, if the n-th packet is received correctly; $X_n = 1$, if the n-th packet is lost and has not been retransmitted; and $X_n = 2$, if the n-th packet is successfully retransmitted after loss.

The key attributes for the characterization of the IPTV transmission are the steady state probabilities ($P_{L,steady}$, $P_{R,steady}$ and $P_{G,steady}$), which determine the probability of loss, successful retransmission, and good transmission states; the run-length probabilities ($P_{L,burst}(l)$ and $P_{R,burst}(l)$), which determine the probabilities of an l long consecutive loss and a consecutively retransmission events respectively. These values can be calculated from the state functions and the state transition probabilities:

$$\begin{aligned} P_{L,steady} &= \lim_{k\to\infty} y_1[k] = \lim_{k\to\infty} (B_r\delta[k] + u[k]A_1 d_1^k \\ &\quad + u[k]A_2 d_2^k + u[k]A_3 d_3^k), && (7) \\ P_{G,steady} &= \lim_{k\to\infty} y_0[k] = \lim_{k\to\infty} (B_{0,r}\delta[k] + u[k]A_{0,1} d_{0,1}^k \\ &\quad + u[k]A_{0,2} d_{0,2}^k + u[k]A_{0,3} d_{0,3}^k), && (8) \\ P_{R,steady} &= \lim_{k\to\infty} y_2[k] = \lim_{k\to\infty} (B_{2,r}\delta[k] + u[k]A_{2,1} d_{2,1}^k \\ &\quad + u[k]A_{2,2} d_{2,2}^k + u[k]A_{2,3} d_{2,3}^k), && (9) \end{aligned}$$

$$\begin{aligned} P_{L,burst}(l) &= (1 - (1 - p_{10} - p_{12}))(1 - p_{10} - p_{12})^{l-1} \\ &= (p_{10} + p_{12})(1 - p_{10} - p_{12})^{l-1}, && (10) \\ P_{R,burst}(l) &= (p_{20} + p_{21})(1 - p_{20} - p_{21})^{l-1}, && (11) \\ P_{G,burst}(l) &= (p_{01} + p_{02})(1 - p_{01} - p_{02})^{l-1}. && (12) \end{aligned}$$

The steady-state probabilities of the 3SCM are important parameters for statistical analysis and long term forecasts. I am going to use them to estimate the packet loss rate of an IPTV connection, and based on this information, make a decision on allowing or denying a packet retransmission. Unfortunately, these probabilities are not deduced in a closed form expression in (7), (8), and (9) and their current form does not allow fast and accurate prediction. I prove, that all of them can be expressed in a closed form.

First, let have a look on the steady-state packet loss rate of $y_1[k]$:

$$P_{L,steady} = \lim_{k\to\infty} y_1[k] = \lim_{k\to\infty} (B_r\delta[k] + u[k]A_1 d_1^k + u[k]A_2 d_2^k + u[k]A_3 d_3^k). \tag{13}$$

To be able to describe this probability in a closed form, the d_i poles have to converge to number less or equal than 1. This is self-evident for $d_1 = 1$, and I state and prove the following theorems for the other two poles:

THEOREM 4.1 (Jursonovics and Imre 2011). *The convergence of the d_2 pole exists: $|d_2| \leq 1$.*

THEOREM 4.2 (Jursonovics and Imre 2011). *The convergence of the d_3 pole exists: $|d_3| \leq 1$.*

Due to my model's simplicity, I conjectured the existence of the convergence above, therefore I examined the values of the d_i poles with a Monte Carlo analysis performed on one million random samples.

I found, that the complex values of d_2 and d_3 are within the unit circle (figure 4.5) in all my test cases. (I also enclosed the algebraic proof the theorems in my dissertation.)

Therefore I determine the steady-state packet loss probabilities:

THEOREM 4.3 (Jursonovics and Imre 2011). *The 3SCM is stable, the steady-state packet loss probability can be expressed in the closed form of*

$$P_{L,steady} = \frac{p_{02}p_{21} + p_{01}p_{20} + p_{01}p_{21}}{p_{10}p_{02} + p_{12}p_{01} + p_{12}p_{02} + p_{21}p_{10} + p_{20}p_{10} + p_{20}p_{12} + p_{01}p_{20} + p_{01}p_{21} + p_{02}p_{21}}$$

4.3 Forecasting Performance of the 3SCM

In relation to the performance evaluation, I compared my model's prediction ability of different transport parameters to the measured values.

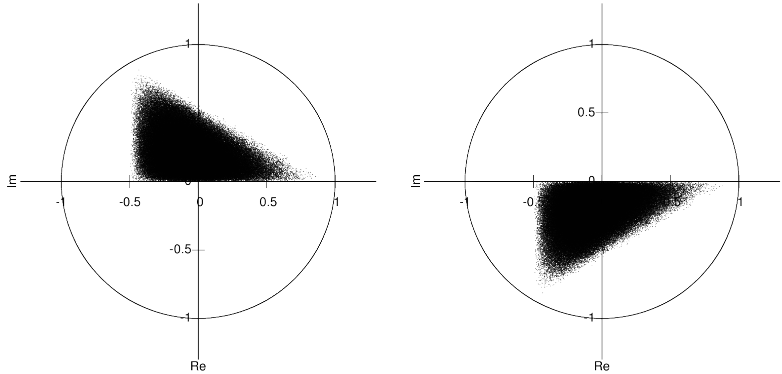

Figure 4.5. Monte Carlo analysis of the poles. These two graphs show that every simulation point of the d_2, d_3 poles, represented by black dots, lie within the unit circle, therefore the model is concluded to be stable.

> THESIS I.3 (Jursonovics and Imre 2011, 2013; Jursonovics and Butyka 2004). *I proved that the 3SCM effectively describes and predicts the loss characteristics of both the packet transmission and retransmission by conducting and analyzing several sample video sequence transmission on the testbed. The relative averaged prediction error remained below 20% for steady state retransmission probabilities and below one magnitude for short burst losses.*

I conducted several measurements in my testbed described. At $w = 10sec$ sliding windows size, the predicted values are close to the measured values. The maximal single relative prediction error is 3.0; in average less than 0.4. It can be also observed, that the prediction accuracy of retransmission probability is better than for packet losses, because packet loss events occur much less, than retransmission (in my model, a packet loss equals to a lost retransmission packet,) and a Markov chain has a well known estimation error for low probabilities (Bartholomew 1975). This effect also results higher relative prediction errors values (20%-40%), but I would like to emphasize, that they represent really small absolute errors: a 40% relative error for a 10^{-6} probability means, that the difference between the estimated value and the real value is less than $0.4 \cdot 10^{-6}$.

Using a larger prediction window, the estimated values are almost exactly the same as the measured values (single relative prediction error is less than 1.2; in average 0.3). The peak error at 90 s is caused by a burst loss on the wireless channel; the Markov chain over-predicted the loss rate based on this concentrated loss of packets.

Next, I analyzed the prediction probabilities of the loss bursts and retransmission bursts at different estimation windows sizes. It can be observed, that the model slightly overestimated the probability of loss bursts, the peak single relative estimation error for a three long loss of bursts is 10, which value remained unchanged for a larger estimation windows. The prediction accuracy of a Markov chain is decreasing for small probability values (the probability of a 3 long loss of burst events is approximately 10^{-4}). However, the probability of the retransmission bursts is almost accurately predicted, the single relative estimation error is 0.1.

4.4 The Optimal Bandwidth Allocation

The smart choice of the allocated retransmission bandwidth is a crucial point in any commercial IPTV design process. As I state in 4.1, an inappropriate value may jeopardize the quality of the customer experience. In real-life solutions, the perfect quality cannot be met; the allowed perception of quality deterioration is a design parameter, which has to be considered in the bandwidth allocation method. To achieve this goal in this section, I unite the two main aspects of bandwidth allocation (see 4.1) and channel model (see 4.2) to provide a bandwidth sizing method for IPTV solutions over wireless networks in this section.

I describe the wireless communication with the state transition characteristics, then with the help of the probability forecast features of 3SCM, I estimate the relevant probabilities for the three limitation aspect of retransmission, which will yields the probability of packet loss in this complex scenario. This will allow me to select (allocate) the retransmission bandwidth to a given packet loss probability.

> Thesis I.4. *I created a method for retransmission bandwidth sizing in IPTV solutions over wireless networks. This method ensures a minimal packet loss probability, and determines the bandwidth allocation accordingly.*

Considering the above introduced two aspects of the retransmission limitations and the steady state probabilities, I use a statistical approach to find the bandwidth allocation with the minimal packet loss probabilities:

$$\frac{B_{RET}}{B_{AV}} = \frac{D_{pkg}}{D_{playo} - 2D_{pkg} - RTT} \cdot \frac{\ln(P_{skip,intra}) - \ln(C_1) + \ln(C_2) - \ln(-\ln(C_1))}{\ln(C_2)} + 1. \quad (14)$$

where C_i are constants.

This equation provides the required method for determining the retransmission bandwidth for a given channel packet skip probability.

4.5 The Optimal Retransmission Algorithm

The traditional retransmission algorithms request all lost packets, therefore they have to implement a network layer traffic shaping to fit the actual retransmission throughput into the allocated bandwidth. This is usually carried out by packet queuing, which increases the overall packet retransmission time, therefore the probability—that a retransmission packet arrives late after its playout time—is higher.

The main advantages of my method are the low resource needs, the consideration of the wireless channel, and a minimal additional delay. My method asses the RET mechanism on the network layer, skips (forbids) the retransmission requests of a lost packet according to the above described intra- and inter-burst channel blocking, and takes the special properties of the wireless channel into consideration.

> THESIS I.5 (Jursonovics and Imre 2013). *I created an algorithm for packet retransmission in IPTV solutions over wireless networks based on my 3SCM. I proved, that the algorithm is effectively kept the retransmission throughput within its allocated bandwidth, and I showed that my algorithm produces fewer overall packet losses, than traditional retransmission, therefore ensures a better service quality. The theoretical formulas of the loss and retransmission probabilities are also calculated.*

Figure 6 shows the means and the standard deviations of the overall packet loss rates for 20, individual streaming sessions in the testbed. It can be seen that in most of the cases (17 of 20), my method provided better results by keeping the mean of the overall packet loss rate below the mean of the traditional retransmission algorithm. In these cases, the average packet loss rate achievement was 0.63%. For the remaining 3 cases, the traditional retransmission algorithm provided in average 0.37% better packet loss values.

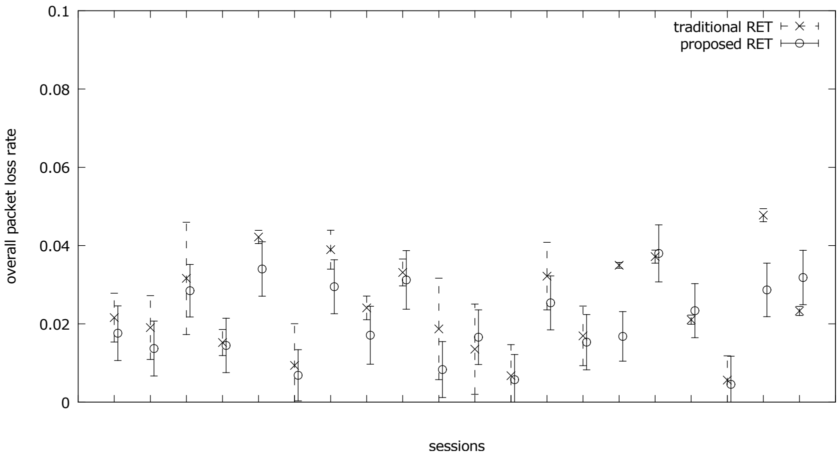

Figure 6. Mean and standard deviation for overall packet loss rate

4.6 The Proposed Business Model and Architecture

Now, I continue the introduction with the second thesis group. The implementation of the 4th generation billing functionality is based on the ascendant system; thus, in the early age of the main concept of billing scheme correspond with the present concept, which is a very mobile service provider centered.

Thesis II.1 (Jursonovics et al. 2004; Jursonovics and Imre 2005; Jursonovics, Butyka, and Imre 2005, 2008; Jursonovics and Imre 2014). *I selected the network operator centric business model for multimedia charging in mobile network based on the charging requirements, and I extended the standard charging architecture with my proxy solution to be able to realize quality based multimedia charging functions in mobile networks.*

To select the appropriate model, the typical product portfolio of IPTV has to be considered. The proposal should enable network operators to collect all service fees in the IPTV bundle, and provide a single bill for IPTV customers. Besides that, the model should allow IPTV providers to extend their subscription packages with partner offerings (for example: HBO Go) to be able to retain customers with competitive product portfolio. Considering these guidelines, I have chosen the network operator centric model as the basis of my own model, which will ensure the easy integration into the mobile landscape.

According to figure 7, over a classic mobile data session, the customers' communication is initiated by a mobile equipment (ME), which establishes a packet data protocol (PDP) context through the radio network subsystem (RNS), Node B, radio network controller (RNC), serving GPRS support node (SGSN), and gateway GPRS support node (GGSN). The SGSN is able to separate the users' flows from one another with their PDP context, but it cannot look inside the IP payload (as it does not decode the encapsulated protocol). Hence the SGSNs are unsuitable for my charging method that would based on the properties of these protocols.

LTE offers an optimized method for creating a data connection: the evolved packet system (EPS) bearer setup reduces the number of signaling messages that need to be sent over the air, however it uses the same tunneling protocols through the evolved Node B (eNB), serving gateway (SGW), PDN gateway (PDN-GW). According to the same argumentation, the SGW is also unsuitable for charging purposes of multimedia streaming.

To overcome these limitations, I introduce a new network element: the streaming proxy, which fits into and completes the existing charging infrastructure, and realizes the required quality based charging functions of the IPTV delivery.

The streaming proxy monopolizes the management of the multimedia delivery, and enables the users to connect through itself to the streaming providers (value added service providers – VASPs). To ensure that the charging proxy cannot be bypassed, I also implement an access restriction function (firewall – FW) in the architecture, which will block all non-authorized media access attempts.

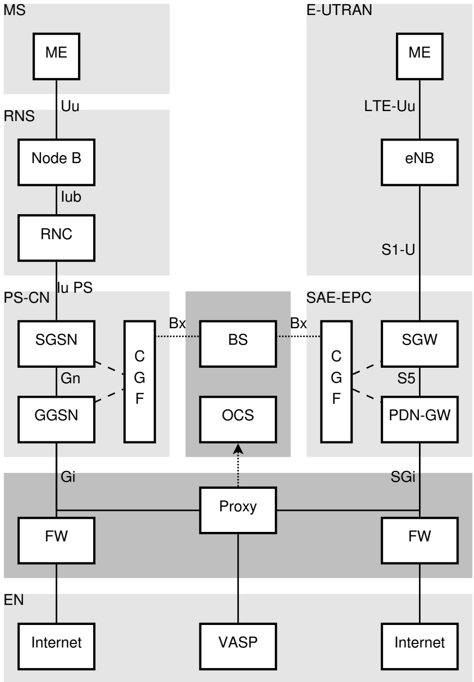

Figure 7. Proxy based charging architecture

4.7 The Quality Based Charging Solution

At last but not least, this section creates a complex quality based charging solution. The $F()$ charging process determines the price ($price$) of a service according to the $\mathbf{c}$ charging parameters including but not limited to data volume, event, resolution, content, watching time:

$$price = F(\mathbf{c}). \tag{15}$$

According to the above described charging architecture, quality assessment, and loss pattern prediction, the quality based charging solution requires the modification of this price regarding the $\mathbf{q}$ perceived quality parameters, which I define with the $P()$ quality based charging policy function:

$$price = F(\mathbf{c}) \cdot P(\mathbf{q}), \text{ where} \qquad (16)$$
$$0 \leq P() \leq 1$$

Thesis II.2 (Jursonovics and Imre 2005; Jursonovics, Butyka, and Imre 2005, 2008; Jursonovics and Imre 2014; appendix D, E). *I created a quality based charging method for IPTV delivery over wireless networks which method effectively considers the quality attributes of the IPTV service and provides a quality based charging feature. I provided the formula of the expected value and distribution of the price determined by my method. I pointed out the advantage of my solution, which reflects the quality differences in the price than traditional volume based charging methods.*

I introduce three similar approximations of the charging policy. The price of an asset is steadily reduced by the linear approximation. Second, the cosine function smooths the sharp edges of the linear policy function, but still leaves a sharp and sudden fall at charging termination. The elliptic approximation gives a smooth transition at both ends.

$$P_{PSNR,linear}(x) = \begin{cases} \text{undefined} & \text{if } x < Q_m, \\ \frac{1-r}{Qp-Qm}x + 1 - \frac{Qp(1-r)}{Qp-Qm} & \text{if } Q_m \leq x \leq Q_p, \\ 1 & \text{if } x > Q_p. \end{cases} \qquad (17)$$

$$P_{PSNR,cos}(x) = \begin{cases} \text{undefined} & \text{if } x < Q_m, \\ \frac{r-1}{2}\cos\left(\frac{x-Q_m}{Q_p-Q_m}\pi\right) + r + \frac{1-r}{2} & \text{if } Q_m \leq x \leq Q_p, \\ 1 & \text{if } x > Q_p. \end{cases} \qquad (18)$$

$$P_{PSNR,elliptic}(x) = \begin{cases} \text{undefined} & \text{if } x < Q_m, \\ \sqrt{1 - \left(\frac{x-Q_p}{Q_p-Q_m}\right)^2} & \text{if } Q_m \leq x \leq Q_p, \\ 1 & \text{if } x > Q_p, \end{cases} \qquad (19)$$

Q_p is the smallest value of quality deterioration that does not involve price reduction, and Q_m is the minimal acceptable quality, on which the service can be still offered, and $Q_p > Q_m > 0$.

The relevant steps for the quality based charging solution follows:

0a. Asset publication: during the publication of an asset, every frame of the asset will be analyzed and the MSE values of the frame differences ($d[k]$) are calculated, then stored, and on request, sent to the the streaming proxy as metadata.

0b. Live broadcast: during live broadcast, the live stream is continuously analyzed and the MSE values of frame differences ($d[k]$) are calculated and sent to the streaming proxy as a real-time metadata feed.

1. Loss reporting: the mobile client is continuously monitoring the packet arrival process, and periodically reports the packet loss pattern (κ) in the above defined RTCP receiver reports to the streaming proxy for scheduled content services and content on demand. Concerning other properties of this RTCP report process (for example: transmission interval, send and receive rules), I adapt of the recommendations of Schulzrinne et al. 2003.

2. Quality estimation: the proxy collects the receiver reports and estimates the total distortion ($D[\kappa]$) with the help of the loss pattern (κ) and the pre-defined frame error sequence ($d[k]$). The charging parameters ($\mathbf{c}$) and the quality parameters ($D[\kappa] \in \mathbf{q}$) are sent to the billing system in form of a CDR.

3. Billing: the billing system implements the charging functions ($F(\mathbf{c})$, $P(\mathbf{q})$) and determines the price and a bill, which reflects the streaming quality.

5 Conclusion

I carried out my research into IPTV delivery over the combination of xDSL and wireless access technologies to explore the special requirements of this mixed environment, and discover the possibilities of service quality improvement. The general research on this field examines the medium access layer transport in the first place, and discusses enhancement proposals concerning the radio transmission, however this is not sufficient for IPTV delivery, because in most cases the wireless device is not managed by the IPTV service provider. In my theses, I addressed the application layer and presented an end to end optimization according to my research objectives:

1. Explore and define the optimal bandwidth allocation scheme of packet retransmission in IPTV Solutions to optimize service quality.

2. Find and evaluate the limitations of packet retransmission in IPTV delivery.
3. Create and analyze a mathematical model of packet retransmission in this mixed environment.
4. Synthesize the efficient bandwidth allocation and optimal retransmission algorithm.

In the second part, I also explored the concept of charging solutions for IPTV delivery over mobile networks, the role and importance of quality of service in the charging process and the establishment of a fair pricing model and charging solution. The general research on this field developed several individual aspect of this context but no common and complete solution was offered for my research objectives:

1. Create a charging method in wireless networks which does consider the quality of the offered multimedia service and offers a fair charging and billing process.
2. Find and evaluate the correlation between service charging and the various attributes of wireless transport.
3. Provide an answer for the problem of a priori quality based pricing.

The main empirical research was conducted on a testbed built for this specific reason, integrating my own realization of the newly created entities, methods and functions:

Loss prediction of the 3SCM. I stated that the prediction performance of loss patterns is accurate, although the low probabilities are slightly overestimated by the model.

Optimal retransmission algorithm. I showed that implementing my algorithm, the IPTV delivery has better (smaller) overall packet loss rates—therefore higher quality—than traditional retransmission algorithms.

Accuracy of quality assessment. I showed that the chosen quality estimation technique is able to accurately predict the quality deterioration caused by packet losses. According to the sample sequence the error remained always between -10% and 15% per frame.

Distribution of the total distortion. Based on several test sequences I deduced that the density of the total distortion follows a heavy side distribution which can be effectively described with an exponential or weibull distribution. I proved my findings with Q-Q plots and Chi-Square tests.

Comparison of different charging policies. Based on a test sequence I compared the three recommended charging policy function and the attention was drawn to the expected advantage of the elliptic based function which provided a smoother transition for charging although it may allowed free of charge service.

In further research, I am going to extend my study with the deeper examination of the throughput management in adaptive bitrate streaming technologies. The emerging standard of the Scalable Video Coding (Fraunhofer Heinrich Hertz Institute 2013) enables operators to offer rate adaptive scheduled content services over multicast transport, which would result a better efficiency of bandwidth utilization (a higher B/B_{RET} ratio) therefore better service quality. To achieve this goal, the retransmission limitation model should to be extended with the inter-dependencies of the scalable video encoding and the packet loss prediction model should reflect the loss probabilities in this multilevel architecture.

As a second objective, I am going to shift the scope of optimization from the end-to-end stream delivery to the radio bearer; I am interested in the optimization possibilities and performance of the 3SCM model in the radio interface. This low lever retransmission could be also enhanced by the a priori knowledge of the loss probabilities for retry mechanism (Miguel et al. 2011).

I am going to extend my study with the deeper examination of the two interesting areas of RTCP-RR falsification and adaptive policy function. The sender reports can be easily falsified (and therefore the service can be accessed on a smaller charge) due to the lack of authentication support of RTCP. I would like to further investigate the possibility of trusted clients and RTCP message signature.

The importance of good quality based charging varies according to the content of the multimedia asset. A poorer quality may be accepted for a quiet static scene than for a scene with actions and key moments. The current proposal for the policy based charging function does not consider this aspect which I will more deeply investigate in further research.

My study offered an evaluative perspective on IPTV delivery, however it encountered a number of limitations, which need to be considered:

MAC layer. The WLAN standard already defines a radio bearer level retry mechanism (IEEE 802.11 Working Group 1999), which is not feasible for IPTV delivery but improves the overall loss probabilities. In my study, this effect was not explicitly considered, therefore my results slightly biased by this feature.

IP fragmentation. Due to the simplicity of the testbed components, the streaming server (ser) did not realized the fragmentation unit feature of the H.264 payload format therefore every frame was encoded into one big packet and the fragmentation was done by the IP stack.

Frames per packet. The resolution (CIF) of the test sequences resulted to encode every frame in a single packet but in commercial scenarios this may be different and several slices of a frame could be encoded into different packets.

I am confident that these new results contribute to the service quality optimization of IPTV Services, and I hope that IPTV platform operators can leverage my findings to ensure an excellent service delivery. I do believe that the quality based charging methods does offer a value both for customers and service providers through the customer's satisfaction and fair pricing, although the existing business cases have to be reconsidered, and business managers have to adapt this new approach, which may take some time till the solution reaches wide commercial implementation and market.

6 Application of the New Results

My dissertation introduced several achievements for service quality improvement of IPTV services over wireless networks, and for fair, quality based charging implementation. Following the detailed introduction and discussion of my methods, I showed, how an IPTV service operator benefits from the realization of these features, and how can the service quality and customer's satisfaction be improved. In my opinion, these factors will be shortly in the foreground because of to the highly competing television market and increasing importance of user experience.

By implementing my new results, the following important benefits will be achieved:

- Cost saving: telecommunication enterprises are challenged to achieve better efficiency and optimize their internal processes to operate with lower costs. Today, by a major US Internet provider, the installation time of a triple-play service takes about four hours (Multimedia Research Group 2008), but the introduction of a wireless STB could reduces the time of on site installation to three hours, and provides a real differentiator from cable providers by making the cabling unnecessary and improving the customer's satisfaction.

- Service portability: WLAN technology allows users to access the entertainment services more conveniently within the household on a tablet, or on the go on mobile devices.
- Operational complexity: maintaining one single, multipurpose IPTV platform instead of many, dedicated to legacy and OTT services, significantly reduces the operational costs and efforts on long term.
- Service quality: the quality based charging will establish relationship between the revenue stream and the service quality, which will motivate IPTV service providers to offer their products with the highest possible quality. This simple financial incentive will ensure continuous technological development, and therefore customer satisfaction.
- Customer satisfaction: a fair pricing will increase customer's satisfaction, and lead to smaller number of complain.
- Quality feedback: as addition, the quality of the IPTV service will be continuously monitored; the quality report can be directly used for quality assurance purposes.

I am confident that these new results contribute to the service quality optimization of IPTV Services, and I hope that IPTV platform operators can leverage my findings to ensure an excellent service delivery.

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