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Research Article

Bandwidth Management in Wireless Home Networks for IPTV Solutions

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The optimal allocation of the retransmission bandwidth is essential for IPTV service providers to ensure maximal service quality. This paper highlights the relevance of the wireless transport in today's IPTV solution and discusses how this new media affects the existing broadcast technologies. A new Markovian channel model is developed to address the optimization issues of the retransmission throughput, and a new method is presented which is evaluated by empirical measurements followed by mathematical analysis.

1. Preface

The terminology of Open IPTV Forum (OIPF) specification release 2 [1] has been chosen to describe the internet protocol television IPTV features in this paper because we experience a wide diversity of terms in several journals which may confuse the reader. We believe that the OIPF terms are straightforward, and they can be easily interpreted on any IPTV solutions although our work is independent from the standard itself.

2. Motivation

The telecommunication industry is tardily changing. The emerging market of the new generation over-the-top (OTT) services from Google, Microsoft, Apple, or Amazon had put a big pressure on operators to move away from the traditional telecom model and assess threats and opportunities from OTT players. There is a big race for the customers today, and legacy industry has to extend its portfolio with various value adding services like triple-play, rich communication, or mobile payment. This paper evaluates one specific topic of this competition, the video broadcasting services.

We have observed the rapid evolution of the IPTV services in the last decade. The high-definition broadcast got popular since its introduction in 2004, and the accessibility of 3D content is growing year by year. The consumer electronic devices become integrated part of our life, customers access digital content from set-top boxes (STBs) to tablets and mobile devices. From the IPTV service provider point of view, the demand of high-quality services emerged; however, the infrastructure of the access network remained the same. The main technology of telecommunication operators providing internet remained the 20–30 years old twisted copper pairs.

On one hand, the IPTV Service Providers are motivated by the maximization of their customer reach, but in many cases digital subscriber line xDSL offers inadequate bandwidth for high quality services [1]. access network providers need to find a solution which 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 [2] or the introduction of a hybrid service which replaces the most bandwidth consuming scheduled content transport with digital video broadcasting DVB-X technology [3], the usage of progressive download, or, as we point out in our work, the implementation of a more effective bandwidth allocation in the access network which would ensure a more efficient transport.

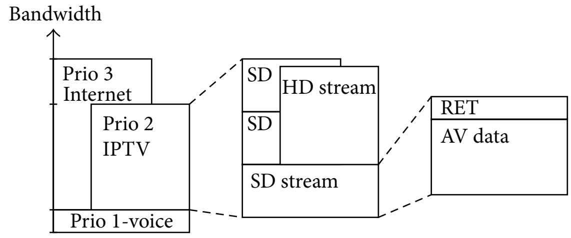

Figure 1: Bandwidth allocation in the access network.

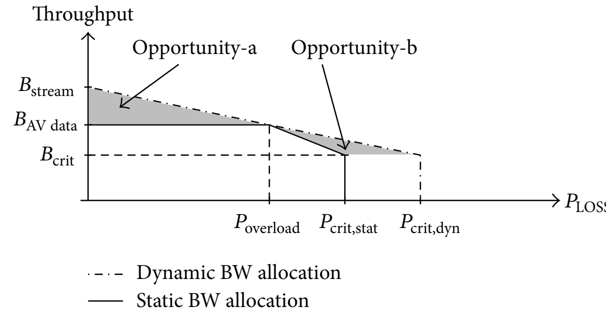

Figure 2: Static versus dynamic bandwidth allocation.

On the other hand, the service portability allows consumers to access the content not only on STBs, but also from various hand-held devices. Today the prime wireless access technology within the consumer domain is the wireless local area network WLAN connection, therefore service platform providers have to adapt their IPTV solution to support the specific requirements of this wireless communication channel.

Most of the current research papers [4–6] take only one of the above mentioned two aspects into consideration. The effect of the wireless channel is usually addressed on the media access control layer, and the research of the IPTV distribution focuses only on the quality of the fixed network infrastructure. Our research does address the above described overlap of the IPTV delivery over WLAN in a limited resource environment of xDSL technology. We present the concept and main problems of bandwidth allocation in Section 3. Our new theoretical model is introduced by Section 4. We develop a more effective bandwidth allocation algorithm in Section 5, which we evaluate and validate. Finally, Section 6 concludes our work and shows some potential further application areas.

3. Bandwidth Allocation

Let us begin the discussion of bandwidth management by introducing a typical triple-play bandwidth allocation scheme on Figure 1. Access network providers usually dedicate a reserved bandwidth in the access network for voice communication and share the remaining throughput between IPTV and internet services with a priority for the former one.

The actual throughput of IPTV service depends on the user's configuration. In most of the cases, a token-based stream management allows the customers to simultaneously receive multiple streams for live viewing or recording purposes (1SD+1HD or 3SD+0HD).

One token allocates bandwidth for the AV (audio-video) data transport and reserves a dedicated bandwidth for the retransmission RET service. We discuss the problem and tradeoff of this bandwidth allocation, therefore, in the following paragraphs, we are going to describe it in details.

The balance between the assigned bandwidth for AV data and the reserved bandwidth for RET service is crucial for achieving the maximal quality in IPTV solutions. On one hand the stream bandwidth as constant in time (because of the widely applied is considered constant bitrate CBR video encoding) the more throughput is assigned for the AV data and 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 hands reserving high bandwidth for error correction degrades the overall stream quality due to the low encoding bitrate. Based on different network installations, the ratio of RET bandwidth to AV data is usually tuned between 10 and 25%, but a suboptimal value may significantly reduce the throughput and, quality of an IPTV Solution.

4. Proposed Model for Bandwidth Allocation IPTV Solutions in WLAN Home Networks

The main concept and benefits of our research are showed by Figure 2. The solid line represents the theoretical throughput of the AV data stream at various packet loss probabilities in case of static bandwidth allocation. The function is constant till P_{overload} loss probability, where the loss rate is so high that the retransmission traffic fully occupies its reserved bandwidth. Above this, rate retransmission does not have enough bandwidth to recover all the packet losses; therefore, the actual throughput of AV data stream is decreasing (not transmitted packets), and customers experience quality deterioration. We also declare a critical bandwidth value (B_{crit}) for AV data transmission. [Below this value, it does not make sense to provide IPTV service due to the massive losses.

A dynamic bandwidth allocation (dash-dot line on Figure 2) increases the throughput of the AV stream at low packet loss rates to achieve a better quality for the IPTV service (opportunity-a). Secondly, at higher loss rates, the throughput of AV stream decreased to avoid double packet delivery (opportunity-b) which enables the operator to expand the value of the critical packet loss rate ($P_{\text{crit,dyn}}$) and provides a lower quality but error-free IPTV service in a worst environment.

In this paper we present only one part of our overall research, the optimal selection of the retransmission throughput. Our method—introduced by the upcoming sections—predicts the loss attributes of the wireless transmission and defines the optimal value of the RET throughput considering the overall loss parameters with the aim of

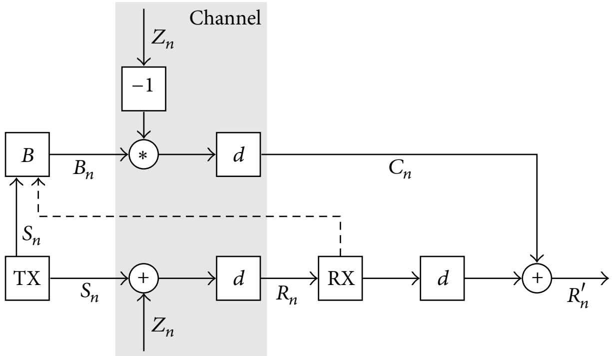

FIGURE 3: Channel model.

minimizing packet losses. First, we describe the channel and the bandwidth models.

4.1. The Channel Model. Considering the requirements above and the attributes of a WLAN transport, we decided to introduce a discrete-time channel and a Markov model for the mathematical description of the UNIT-17 interface (AV data streams in the Access and residential Networks). We consider not only the packet arrival, but also the retransmission traffic as a discrete-time stochastic process, and we prove that it is a homogeneous Markov chain.

Let $S_n = \{0, \text{for all } n\}$ represent the number of sent packets at the time n on Figure 3, TX the multicast content delivery function, $Z_n = \{0; 1, \text{for all } n\}$ an additive white noise in the wireless communication channel, d the transmission delay, and RX the open IPTV terminal function (OITF). The received packets are expressed with $R_n = \{0$, for successful packet transmission; 1, for packet loss$\}$. We assume that the receiver detects a packet loss (by checking the sequence numbers of packets) and requests the retransmission $B_n = \{1$, for a packet retransmission; 0 otherwise$\}$ of every lost packets only once from the B Fast Channel Change/Retransmission server. We also assume that the retransmission request communication is protected by an error-free protocol, like TCP. By this definition, we obtain

$$B_n = R_n = Z_{n-d}. \tag{1}$$

The B_n signal travels through the same wireless channel; therefore, it is also effected by the same Z_n noise, and it may be also lost. We model this effect by expressing the received correction signal $C_n = \{0, 1\}$ with three operators, an inverter (−1), a multiplier (*), and a channel transmission delay (d). These functions enable us to assign the value of 1 for C_n only in the case when the retransmission signal is not effected by the channel noise (not lost), and the value of 0 otherwise.

The final received and corrected signal $R'_n = \{0$, for successful packet transmission, 1 for packet loss (unsuccessful retransmission), and 2 for successful packet retransmission$\}$ is

$$\begin{aligned} R'_n &= R_{n-d} + C_n = S_{n-2d} + Z_{n-2d} + B_{n-d} * Z^{-1}_{n-d} \\ &= S_{n-2d} + Z_{n-2d} + Z_{n-2d} * Z^{-1}_{n-d}. \end{aligned} \tag{2}$$

Let us observe that the first term of the addition equals to 0 by definition and the last term equals to the sampling of the Z_n white noise with its own delayed signal. The autocorrelation of the white noise is zero for all nonzero time shifts [7]; therefore, R'_n can be described as a sequence of an independent random variable which satisfies the Markov property. $X_n \overset{\text{def}}{=} R'_n$ is a homogeneous Markov chain.

Let $X_n = 0$ if the nth packet is received correctly; $X_n = 1$ if the nth packet is lost and has not been retransmitted; finally $X_n = 2$ if the nth packet is successfully retransmitted after loss. We analyzed and described a three-state Markov model in our previous publication; therefore, we simply list most important properties. For a detailed discussion including the resolution of (3)–(6) and for the meaning of the probabilities, please refer to our former paper [8].

The transition matrix

$$\begin{bmatrix} 1 - p_{01} - p_{02} & p_{01} & p_{02} \\ p_{10} & 1 - p_{10} - p_{12} & p_{12} \\ p_{20} & p_{21} & 1 - p_{20} - p_{21} \end{bmatrix}. \tag{3}$$

The steady-state packet loss rate

$$\begin{aligned} P_{\text{LOSS,steady}} &= (p_{02}p_{21} + p_{01}p_{20} + p_{01}p_{21}) \\ &\quad \times (p_{01}p_{20} + p_{01}p_{21} + p_{02}p_{21} + p_{10}p_{02} + p_{12}p_{01} \\ &\quad + p_{12}p_{02} + p_{21}p_{10} + p_{20}p_{10} + p_{20}p_{12})^{-1}. \end{aligned} \tag{4}$$

The steady-state packet retransmission rate

$$\begin{aligned} P_{\text{RET,steady}} &= (p_{10}p_{02} + p_{12}p_{01} + p_{12}p_{02}) \\ &\quad \times (p_{01}p_{20} + p_{01}p_{21} + p_{02}p_{21} + p_{10}p_{02} + p_{12}p_{01} \\ &\quad + p_{12}p_{02} + p_{21}p_{10} + p_{20}p_{10} + p_{20}p_{12})^{-1}. \end{aligned} \tag{5}$$

And the probability of loss burst with length l

$$P_{\text{RET,burst}}(l) = (1 - p_{20} - p_{21})^{l-1}(p_{20} + p_{21}). \tag{6}$$

4.2. The Model of Bandwidth Limitation. The previous section introduced how the wireless channel affects the packet transmission, and now we are going to analyze the the bandwidth allocation in IPTV solutions.

We introduce three planes of the IPTV packet transmission. The *transmitter* plane represents the provider's network, *receiver* plane represents the OITF, and *playout* plan represents the content presentation within the OITF. The events of the packet loss usually show a burstiness in wireless communication [9–11]. Therefore, we investigate an intraburst loss on Figure 4 after the first k consecutive packet losses (b) the receiver requests them for retransmission (c) which packets are delivered within the allocated bandwidth for packet transmission (d) to the presentation device (e). For a successful retransmission, all retransmitted packet should

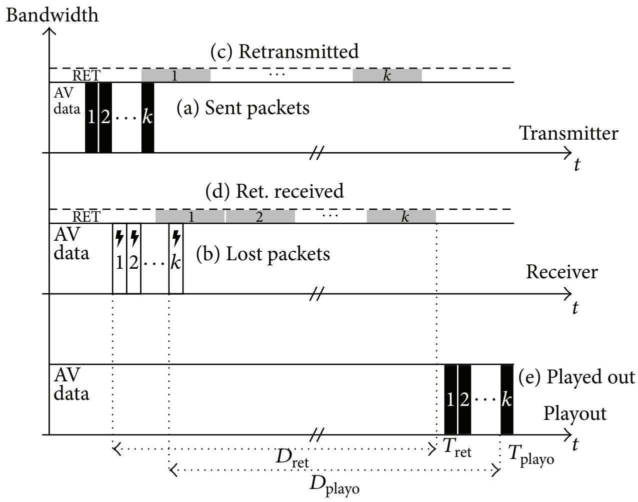

Figure 4: Intraburst retransmission.

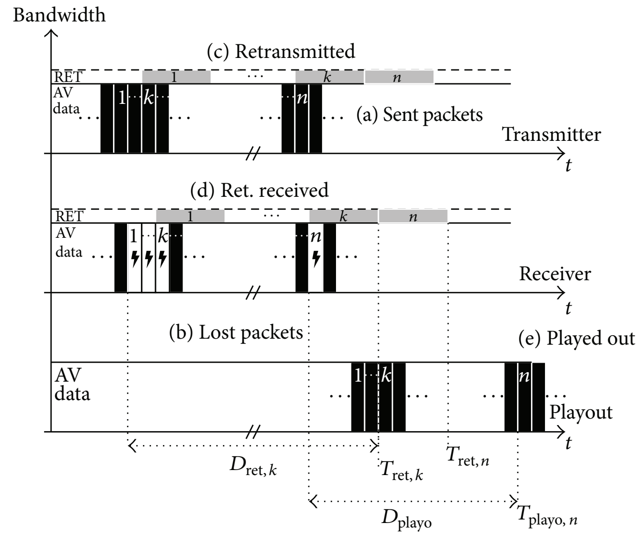

Figure 5: Interburst retransmission.

arrive earlier than their presentation time ($T_{\text{ret}} \leq T_{\text{playo}}$). Expressing this condition with the durations

$$D_{\text{pkg}} + \text{RTT} + k\frac{B_{\text{AV}}}{B_{\text{RET}}} D_{\text{pkg}} < (k-1) D_{\text{pkg}} + D_{\text{playo}}, \quad (7)$$

where D_{pkg} is the average transmission time for a packet, RTT is the round-trip time, B_{AV} and B_{RET} are the allocated bandwidths on the communication channel, and D_{playo} is the packet playout buffer in the OITF. The formula of the maximal number of consecutive packets which can be successfully retransmitted is defined as

$$k_{\text{RET,intra,max}}\left(\frac{B_{\text{RET}}}{B_{\text{AV}}}\right) = \frac{D_{\text{playo}} - 2D_{\text{pkg}} - \text{RTT}}{D_{\text{pkg}}\left(\left(1/\left(B_{\text{RET}}/B_{\text{AV}}\right)\right) - 1\right)}. \quad (8)$$

The actual throughput of the AV stream may vary by installations; therefore, we expressed this value as a ratio of B_{RET} and B_{AV}.

Second, we highlight the barrier of interburst behavior on Figure 5. After a loss of long burst, the retransmission bandwidth is occupied by the traffic of the retransmitted packets even if there is no other packet loss at the time in the video stream. This means that a loss event *blocks* the retransmission channel. We are interested in the following question: assuming a $k < k_{\text{RET,intra,max}}$ long burst of loss, after how many packets (n) can a new loss burst occur which would be also successfully retransmitted (e.g., what is the minimal distance ($n-k$) between two loss bursts if the first burst lasts for k packets?). Now, $T_{\text{ret},k} \overset{\text{def}}{=} T_{\text{playo},k}$, and $T_{\text{ret},n} \leq T_{\text{playo},n}$. Figure 5 shows that

$$D_{\text{pkg}} + \text{RTT} + (k+1)\frac{B_{\text{AV}}}{B_{\text{RET}}} D_{\text{pkg}} < (n-1) D_{\text{pkg}} + D_{\text{playo}}, \quad (9)$$

where $n > k$. Expressing n

$$n_{\text{RET,inter,min}}\left(k, \frac{B_{\text{AV}}}{B_{\text{RET}}}\right) = (k+1)\frac{B_{\text{AV}}}{B_{\text{RET}}} + \frac{\text{RTT} - D_{\text{playo}}}{D_{\text{pkg}}} + 2. \quad (10)$$

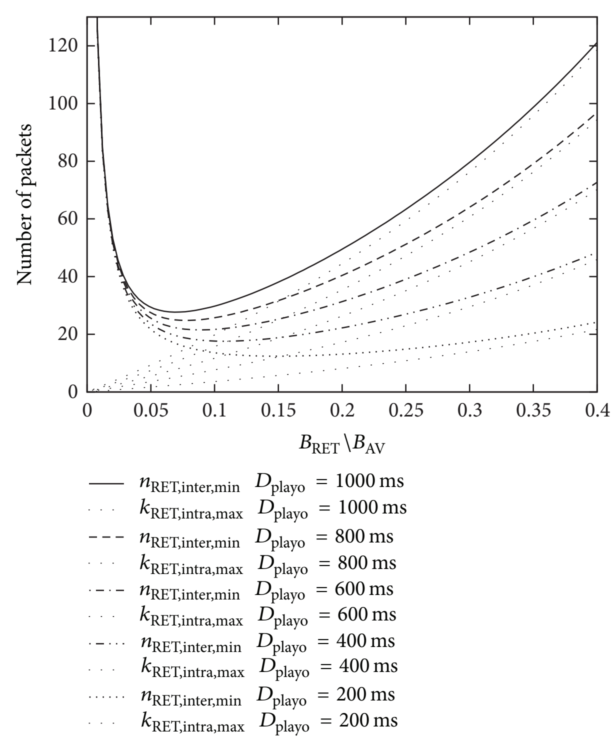

Figure 6: Intra- and interburst limitation.

We present the above declared functions (8)–(10) on Figure 6. This graph shows that at small values of $B_{\text{AV}}/B_{\text{RET}}$ (0%–15%), the effect of the inter-burst blocking is greater. For example, at 5% the maximal consecutive burst length is 10 packets, and the retransmission channel is blocked by this traffic for 30 packets. Above 20%, this effect becomes insignificant.

We also state that with grater playout buffer (D_{playo}), RET is able to correct larger loss bursts at the price of a greater end-to-end delay.

5. The Optimal Retransmission

In this section, we introduce a new method for retransmission bandwidth allocation based on our models with the aim of achieving a better video quality. The method is realized in a test environment, and our hypothesis is measured and proved.

Traditional RET 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 done by packet queuing which increases the overall packet retransmission time; therefore, the probability of a retransmission packet arrives late (after its playout time) is great. Several papers addressed this problem [12], introducing a selective retransmission protocol by evaluating the traffic on the application level, and assigning priority and importance for each packet retransmissions.

The main advantages of our method are the minimal additional delay, the low resource needs, and the consideration of the wireless channel. Our method assess the RET mechanism on the network layer, skips (forbids) the retransmission requests of a lost packet according to the above described intra- and interburst channel blocking, and takes the special properties of the wireless channel into consideration. Our algorithm consists of three steps.

(1) The packet arrival process is continuously monitored for packet loss.

(2) $k_{\text{RET,intra,max}}$ and $n_{\text{RET,inter,min}}$ are calculated.

(3) A lost packet is requested retransmission only if the intra- and inter-burst channel blocking do not forbid the retransmission; otherwise, the retransmission request is skipped.

5.1. Empirical Evaluation. To evaluate our model and methods, we followed the OIPF system architecture [13] and implemented the following OIPF functions in a test environment (source codes are available on [14]).

(1) Multicast content delivery function, *ser*, c++ application generates UDP/RTP multicast traffic and implements a simple control protocol hosted by an x86 Linux server connected to the core network of Deutsche Telekom (DT).

(2) RET server, *ret*, c++ application stores the multicast traffic in a circular buffer and implements a simple retransmission request protocol hosted by the same x86 Linux server.

(3) Unit-17 interface, part a was realized by the ADSL2+ access network of DT and was provided by a DLink ADSL modem.

(4) Unit-17 interface, part b was realized by a 802.11 b WLAN network and was provided by a Cisco 1200 series wireless access point (AP) connected to the ADSL modem.

(5) OITF: *cli*, c++ application implements a simple multicast receiver and controls functions of the multicast content delivery function and RET server. Hosted on a x86 Linux laptop connected to the Cisco AP.

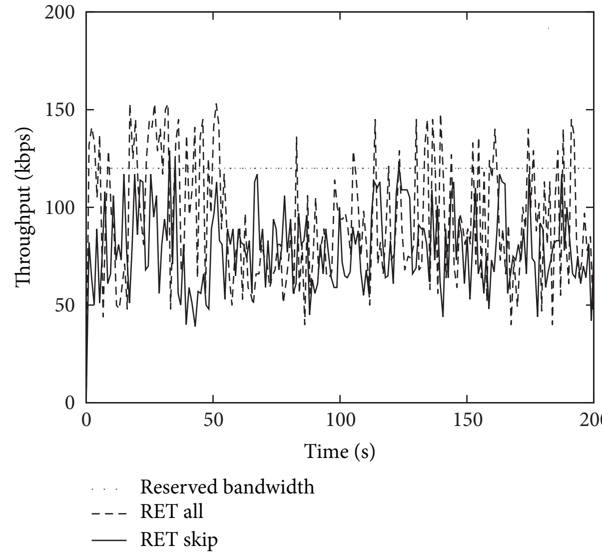

Figure 7: Retransmission throughput.

Figure 7 shows the throughput of the retransmission stream in two cases: traditional retransmission (*RET all*: all lost packets were requested for retransmission) and our new retransmission method (*RET skip*: retransmission requests may be skipped based on the actual parameters of the channel). It can be clearly seen that our algorithm kept the retransmission throughput under its dedicated bandwidth which ensured the in-time delivery of the retransmission packets however we intentionally skipped those retransmission requests which in time delivery would not be ensured due to the channel blocking (intra- and inter-burst effect).

The main benefit of our method is showed by Figure 8. We compared the total packet loss rate in the above mentioned two cases, and we found that with the smart skipping of packet retransmission requests, we were able to achieve a better (smaller) loss rate then retransmitting all of the packets. Our method avoided the effect of late retransmission. IF a packet is requested for retransmission without ensuring the necessary transport bandwidth, then it may delay further retransmission requests which may arrive to late after their playout time. This causes an inefficient retransmission bandwidth utilization which increases the overall packet loss rate (on the *playout* plane).

5.2. The Effect of the Intraburst Limitation. Let us analyze our results theoretically as well. In this and in the upcoming section, we characterize the Unit-17 interface with the transition matrix of our Markov model and the design attributes of the access network. Applying the intra- and inter-burst limitations on our model, we derive the probability of the retransmission skip caused by our algorithm. Finally, we express the overall packet loss rate which is a key indicator for the quality of the video transmission.

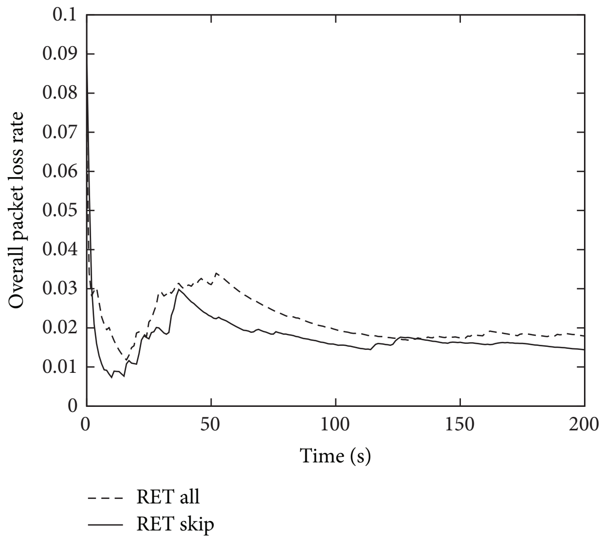

FIGURE 8: Overall packet loss rate.

The intra-burst limitations have a significant short-time effect if the distance of the burst losses is great ($p_{00} \to 1$). Our question is about the probability of retransmission packet skip. First, we calculate the maximal number of consecutive retransmission requests

$$k_{\mathrm{RET,intra,max}} = \frac{D_{\mathrm{playo}} - 2D_{\mathrm{pkg}} - \mathrm{RTT}}{D_{\mathrm{pkg}}\left(\left(B_{\mathrm{AV}}/B_{\mathrm{RET}}\right) - 1\right)}. \tag{11}$$

The probability of l long retransmission request is given by the Markov model (6). We calculate the probability of m packet skips if the retransmission burst is greater than $k_{R,\mathrm{burst,max}}$, which is

$$P_{\mathrm{skip}}(m) = \left(p_{20} + p_{21}\right)\left(1 - p_{20} - p_{21}\right)^{k_{\mathrm{RET,intra,max}}-1+m}. \tag{12}$$

The overall probability of a packet skip is given by

$$\begin{aligned} P_{\mathrm{skip,intra}} &= \lim_{i\to\infty}\sum_{i=1}^{\infty}\frac{P_{\mathrm{skip}}(i)}{i} \\ &= \lim_{i\to\infty}\sum_{i=1}^{\infty}\frac{\left(p_{20} + p_{21}\right)\left(1 - p_{20} - P_{21}\right)^{k_{\mathrm{RET,intra,max}}-1+i}}{i} \\ &= \left(p_{20} + p_{21}\right)\left(1 - p_{20} - p_{21}\right)^{k_{\mathrm{RET,intra,max}}-1} \\ &\quad \times \lim_{i\to\infty}\sum_{i=1}^{\infty}\frac{\left(1 - p_{20} - p_{21}\right)^{i}}{i}. \end{aligned} \tag{13}$$

Let us observe that the last sum can be expressed as a special form of the polylogarithm (also known as Jonquière's function)

$$Li_s(z)\big|_{s=1} = \sum_{k=1}^{\infty}\frac{z^k}{k^s}\Big|_{s=1} = \sum_{k=1}^{\infty}\frac{z^k}{k}, \tag{14}$$

for every $-1 \le z < 1$. ($1 - p_{20} - P_{21}$ satisfies this criteria. Therefore using the well-known formula of $Li_1(z) = -\ln(1 - z)$, the equation can be expressed in a closed form

$$\begin{aligned} P_{\mathrm{skip,intra}} &= \left(p_{20} + p_{21}\right)\left(1 - p_{20} - p_{21}\right)^{k_{\mathrm{RET,intra,max}}-1} \\ &\quad \times (-1)\ln\left(1 - \left(1 - p_{20} - p_{21}\right)\right) \\ &= \left(p_{20} + p_{21}\right)\left(1 - p_{20} - p_{21}\right)^{k_{\mathrm{RET,intra,max}}-1} \\ &\quad \times (-1)\ln\left(p_{20} + p_{21}\right). \end{aligned} \tag{15}$$

The overall packet loss can be expressed as a sum of the probability of packet skip (15) and the steady state probability of packet loss (4).

5.3. The Effect of the Interburst Limitation. We ask the same question as in the previous section, what is the probability of packet skip? Let us assume that the first burst is small enough to be retransmitted ($k < k_{\mathrm{RET,intra,max}}$). The probability of retransmission burst is of l size is given by our Markov model (6). The probability of k retransmission burst followed by $n-k$ good transmission burst and a second retransmission

$$\begin{aligned} f(k,n) = {} & \left(p_{20} + p_{21}\right)\left(1 - p_{20} - p_{21}\right)^{k-1}p_{20} \\ & \times \left(p_{01} + p_{02}\right) \\ & \times \left(1 - p_{01} - p_{02}\right)^{n-k-1}p_{02}. \end{aligned} \tag{16}$$

The first packet of the second retransmission burst is skipped if $n < n_{\mathrm{RET,inter,min}}(k)$. From this, we can calculate the probability of one packet skip for k

$$P_{\mathrm{skip,inter,k}} = \sum_{i=k}^{n_{\mathrm{RET,inter,min}}(k)} f(k,i). \tag{17}$$

For the overal packet skip probability, we have to summarize (17) for every $k < k_{\mathrm{R,burst,max}}$

$$P_{\mathrm{skip,inter}} = \sum_{j=1}^{k_{\mathrm{RET,intra,max}}}\sum_{i=j}^{n_{\mathrm{RET,inter,min}}(j)} f(j,i). \tag{18}$$

The overall packet loss can be expressed as a sum of the probability of packet skip (18) and the steady-state probability of packet loss (4).

6. Conclusion

In this paper, we highlighted the relevance of the wireless transport in today's IPTV solutions, and we pointed out that it's and the access network's combined effect on the bandwidth management is not discussed deeply by publications. Our general research project targets this specific area by introducing several optimization methods from which we presented one, the optimization of the packet retransmission on the previous pages.

We created a new discrete-time channel model to describe the effect of the burst losses on the IPTV service quality and the role of the packet retransmission. We proved that it is a Markov Chain, and as part of our results, we expressed and evaluated its most important quantitative parameters. Using our model, we also introduced an algorithm for retransmission optimization in IPTV solutions over WLAN home networks.

As a further evaluation of our results, we created a testbed in alignment with the OPIF system architecture, and we performed the empirical analysis of our channel model and methods. We showed that our concept improved the overall packet loss characteristics. Furthermore, we enclosed a mathematical analysis of our algorithm, and we derived the theoretical packet loss probabilities to support our measurements.

In the next research phases, we are going to investigate, introduce, and leverage our theoretical results of throughput management in the adaptive bitrate streaming technologies for IPTV solutions, and we are going to evaluate our channel model in the media access control layer of the wireless access technologies.

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