OPTIMAL RATE ALLOCATION AND LOST PACKET RETRANSMISSION IN VIDEO STREAMING

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Abstract:*We consider the problem of optimal rate allocation and admission control for adaptive video streaming sessions in wireless networks with user dynamics. The major aim is to achieve an optimal tradeoff between several key objectives: maximizing the average rate utility per user, minimizing the temporal rate variability, retransmission of lost packets and maximizing the number of users supported. We identify the structure of algorithms that achieve asymptotically optimal performance in large-capacity systems, and exploit the insight into this structure to devise parsimonious and robust online algorithms. Extensive simulation experiments demonstrate that the proposed online algorithms perform well, even in systems with relatively small capacity.*

KEYWORDS *Network simulator2, VMware, fedora, Linux, Video streaming ,QoS, VoD, Rate Adaption*

1.INTRODUCTION

Video traffic is experiencing tremendous growth, fueled by the proliferation of online video content and the steady expansion in transmission bandwidths. The amount of video traffic is forecast to double annually in the next several years, and is expected to account for the dominant share of wire line as well as wireless Internet traffic soon[1].The available hardware and technology for consumers and service providers today allow for advanced multimedia services over IP-based networks. Hence, the popularity of video and audio streaming services such as Video-on-Demand (VoD), advanced on-line gaming, and video chatting and conferences are increasing.[8] The demand for resource efficiency and robustness in the network follows. The current commercially available data transfer technology for streaming media does not adjust well to the best effort heterogeneous Internet architecture, and QoS demands are impossible to guarantee.[3] A central problem with this development is that the Internet itself is extremely dynamic and unpredictable, while the real-time services we want to lay on top of it demand stable conditions for constant and uninterrupted throughput.[7] It is not only the demand for QoS that increases. As CPU and memory get physically smaller and more efficient, the number of applications that can be run on small devices increases as well. With this new flora of handheld computers, PDAs and cell phones, the wish for terminal mobility and easy wireless access to the same advanced services follows.[2] Now, while the Internet itself has unpredictable throughput, the wireless access networks add even more uncertainty to the QoS-possibilities. Most Internet based VoD services today rely on pre-compressed media, allowing consumers to choose between two or three different qualities to fit the given end to end connection (figure 1).[9] For video conferencing and live streaming, the media is compressed on the fly, but the compression rate is normally controlled by an offline codec. This means that the video codec has no direct contact with the network and transport layer, where the actual transmission status is found. Thus, these approaches do not fully utilize the available resources at any given time and they might not provide the chosen quality during the entire session because of sudden changes in the network load somewhere along the path[6]. With an online codec solution, the communicating parties of a multimedia transfer could adopt to the network state at any given time in the session. This is extra desirable for a mobile terminal which has a constantly changing radio coverage, or for a system that supports session-mobility, allowing the user to change terminal equipment without interrupting or restarting an ongoing session. Figure the concept behind a solution based on continuous network feedback upstream from the client to the media server[4].

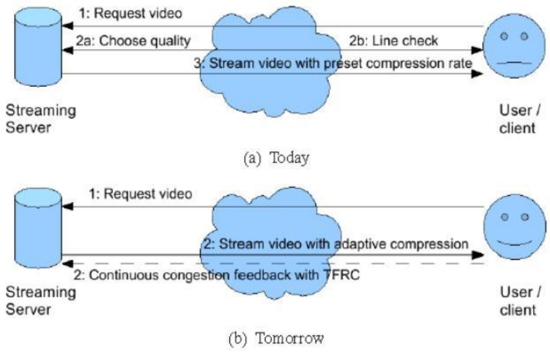


Figure 1.Typical streaming service with (b) and without (a) congestion feedback.

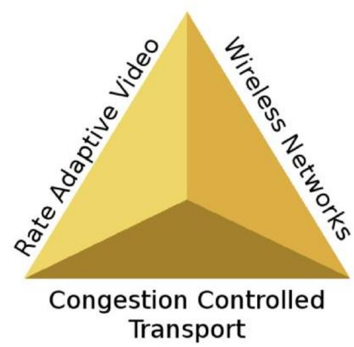
1.1 PROBLEMS FACED

In the development of robust, scalable and intelligent new technologies for multimedia transfer over present and future wireless IP-networks, many concerns need to be taken into account. The solutions can be found by improvements in many levels of the ISO stack, or by novel protocol design. The challenges include:

- Calculating desired rates from current status and collected statistics.
- Transporting network feedback to the sender.
- Interpreting the feedback and notifying the transmitting application.
- Scaling the multimedia output rate from the sender.
- Providing stable transmission conditions for mobile users.
- Minimizing delay and jitter for real-time applications.
- Staying fair to concurrent traffic on the link.
- Differentiating between congestion loss and link error loss.

All of the challenges above have been addressed by separate research groups around the world, but where one solution is brilliant in the aspect discussed in that study, it fails to solve problems discussed in another. This thesis aims to summarize the problems found and the suggested solutions, and to account for the complexity of multimedia streaming over wireless networks. An extensive part of the thesis is used to explain how the technologies can be tested with simulations.

Illustrations of the features and problems of current and new technologies will be presented through the described simulation framework.



2. PROPOSED CONCEPT

2.1. LIMITATIONS

The topic of this thesis touches all protocol layers involved in a video streaming service. However, the presented simulation, status summary and analysis have limitations both in scope and possibilities.

2.2. SCOPE

The methods, mechanisms and protocols that are presented, tested and analyzed are discussed given the need and demand for a set of services with a certain quality. How the services are built up, and how the technology will and can be used in different scenarios and settings, is mentioned but not emphasized any further. The focus of this work is mainly on the transport and MAC layers of the protocol stack, and does not prioritize details from the physical or application layers.[5] The cost-efficiency of each technology in real-life implementation and business-related topics such as time to market, commercial support, industry cooperation and market potential, which are fundamental for a thorough understanding of a standard development process, are left for future studies. The thesis seeks to enlighten the technological possibilities in terms of quality and performance, but an important factor such as security is left out of the scope. For every new product and service intended for online use today, security is a big concern that may be regarded as a part of the total QoS. [3]Understanding topics such as service authentication, authorization and confidentiality, and how current technology in this field can be integrated with new media transport without breaking their performance and intention, is very important. Without this knowledge, technology that can replace today's well-established protocol stack (or parts of it) may not succeed.

C.BLOCK DIAGRAM

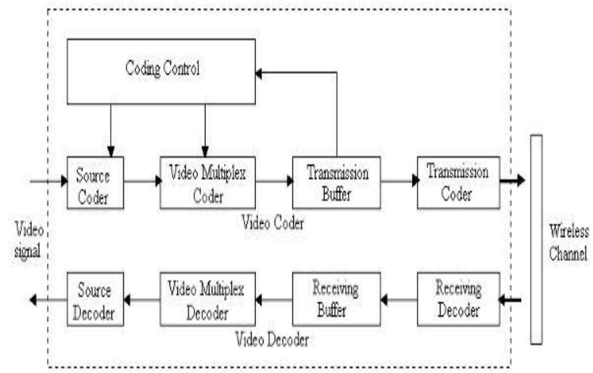


Figure2.block diagram of video transmission

Figure 2. illustrates an overview of the transmission system described above. Broadband wireless networks are known to be susceptible to variations in wireless link quality. To counter the variation in link quality, a user-defined link adaptation scheme is used in wireless networks such as IEEE 802.11 and IEEE 802.16 networks. This mechanism is also known as rate adaptation in other standards (e.g., 3GPP). The link adaptation scheme adapts transmission parameters to dynamic channel conditions, based on some measured parameters. These parameters are generally measured at the receiver and are delivered to the transmitter through a feedback mechanism. The transmitter uses this information to determine a specific modulation and coding scheme (MCS) for the next packet to be transmitted. Under good channel conditions, more information bits and spectrally efficient modulation schemes can be used; whereas, under bad channel conditions, link adaptation adds resilience to the transmitted signal, reducing the number of delivered information bits per symbol. Several transmitter parameters can be modified by the link adaptive scheme.

2.4. FORMULAE

We take the function $G(\cdot)$ to be affine,

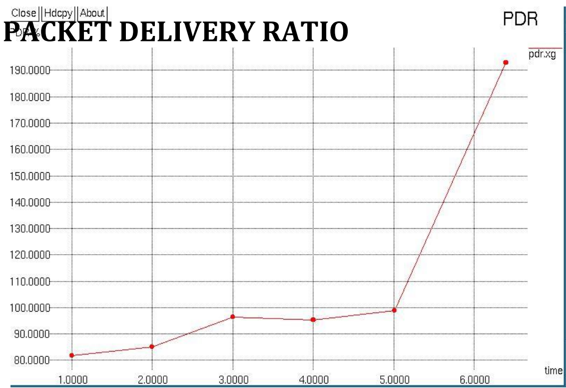
$Hk (wk) = wk (QR(k) − QR(k − 1))$

Let $G(\cdot)$ be a function on W (as defined in Section V) of the form

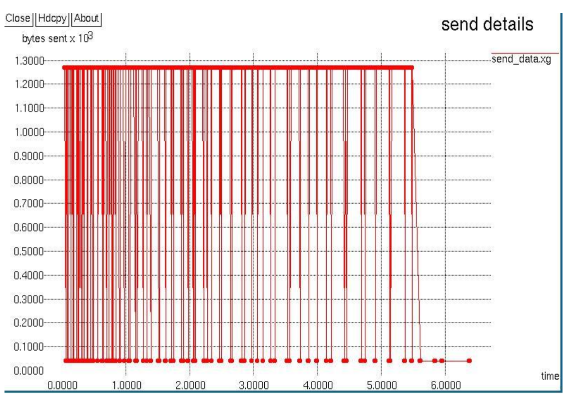
$G(w) = h0 +\_KRk=1 hk(wk).$

2.5.GRAPH

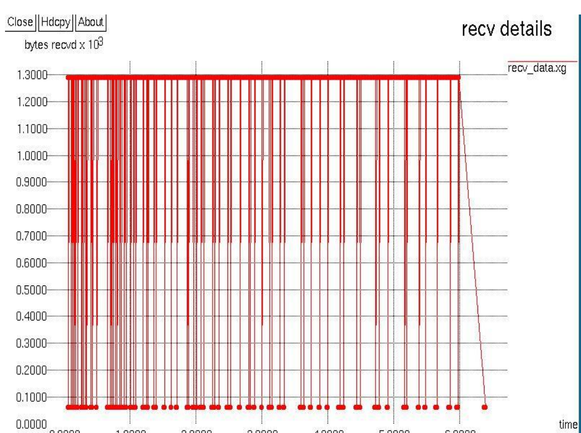
PACKET DELIVERY RATIO



SENT PACKETS



RECEIVED PACKET



3. CONCLUSION

We have addressed the problem of optimal rate allocation for video streaming in wireless networks with user dynamics, and developed online, parsimonious algorithms with provable convergence properties and performance guarantees. In future work we plan to extend the work to account for capacity variations due to slow fading as caused by user mobility.

4. ACKNOWLEDGEMENT:

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