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Packet Loss Rate

The packet loss rate refers to the number of sent packets minus the number of received packets in a portion of time [44–46].

From: Internet of Things, 2020

Chapters and Articles

Modeling and performance evaluation of resource allocation for LTE femtocell networks

Ying Loong Lee, ... Teong Chee Chuah, in Modeling and Simulation of Computer Networks and Systems, 2015

9.3 Packet loss rate

The PLR is an important performance measure for real-time flows such as video and VoIP. Since these data flows are guaranteed certain data rates for smooth and seamless transmission, the number of packets lost or dropped during transmission must be kept low. In a transmission interval, the PLR can be calculated as follows:

$$\mathrm{PLR} = \frac{N^{tx} - N^{rx}}{N^{tx}} \times 100\% \tag{24.10}$$

where N^{tx} and N^{rx} are the total number of transmitted and received packets, respectively. This evaluation can be easily performed by extracting all the real-time packet sizes which are both transmitted and received respectively and one of the evaluation tools in LTE-Sim can be used to calculate the PLR. The PLRs of video and VoIP flows are

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Adaptive FEC-Based Error Control for Internet Telephony

Jean-Chrysostome Bolot, ... Don Towsley, in
Readings in Multimedia Computing and Networking, 2002

Abstract

Excessive packet loss rates can dramatically decrease the audio quality perceived by users of Internet telephony applications. Recent results suggest that error control schemes using forward error correction (FEC) are good candidates for decreasing the impact of packet loss on audio quality. With FEC schemes, redundant information is transmitted along with the original information so that the lost original data can be recovered at least in part from the redundant information. Clearly, sending additional redundancy increases the probability of recovering lost packets, but it also increases the bandwidth requirements and thus the loss rate of the audio stream. This means that the FEC scheme must be coupled to a rate control scheme. Furthermore, the amount of redundant information used at any given point in time should also depend on the characteristics of the loss process at that time (it would make no sense to send much redundant information when the channel is loss free), on the end to end delay constraints (destination typically have to wait longer to decode the FEC as more FEC information is used), on the quality of the redundant information, etc. However, it is not clear given all these constraints how to choose the “best” possible redundant information.

We address this issue in the paper, and illustrate our approach using a FEC scheme for packet audio recently standardized in the IETF. We show that the problem of finding the best redundant information can

be expressed mathematically as a constrained optimization problem for which we give explicit solutions. We obtain from these solutions a simple algorithm with very interesting features, namely i) *the algorithm*

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Measurements and Sustainability

Eric Rondeau, ... Gérard Morel, in Green Information Technology, 2015

Service of Data Transport

The data transport service uses three specific metrics: the bandwidth, the jitter, and the packet loss rate (Michaut and Lepage, 2005). There are two types of bandwidth. The first type is the *total path bandwidth*, also called *path capacity*. It expresses the maximum total throughput accepted by the path. The second type is the *available bandwidth* expressing the maximum throughput offered to the user. Thus, the available bandwidth is the path capacity minus the used traffic (concurrent traffic). The bandwidth measure is based on interval packet times.

Jitter is the delay variation between the same successive requests. It is not a measured value but is obtained by subtracting the delays of two successive requests.

The reliability of a communication network path is expressed by the *packet loss rate*. This metric is equal to the number of packets not received divided by the total number of packets sent. It should be noted that erroneous packets are not generally considered outside the lost packets in computer networks because most applications require data integrity. Indeed, the frame reception driver discards each erroneous

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Phone-to-Phone Configuration for Internet Telephony

Yiu-Wing Leung, in Advances in Computers, 2011

3.3.4 Simulation Model and Results

Lightweight piggybacking involves interdependent steps, as shown in Figs. 12 and 13. Because of this interdependency, it is very difficult to mathematically analyze the performance of lightweight piggybacking. Therefore, we conduct computer simulation for performance evaluation.

We consider a dependent loss model in which packet loss is modeled by a two-state Markov chain [54]. This model is commonly adopted in the literature because packet losses in the Internet exhibit temporal dependency. We let the average burst length be 3, the number of voice streams n be 60 (i.e., n is large for effective utilization of the redundancy, while the resulting coding/decoding time is acceptably small, as shown in Table VI), the size of a voice packet be one unit, and the size of a FCVP be one third of the original size (e.g., this corresponds to multilayer coding with one base layer and two enhancement layers). For convenience, we let R be the ratio of

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Receiver-driven Layered Multicast

Steven McCanne, ... Martin Vetterli, in
Readings in Multimedia Computing and Networking, 2002

8 Future Work

RLM is the first comprehensive instance of a receiver-driven multicast adaptation algorithm and we have just scratched the surface of this problem. While we have evaluated RLM in terms of packet loss rates, the ultimate evaluation metric is the level of quality perceived by the user. We will soon carry out qualitative performance measurements both in a controlled environment as well as by fielding an implementation in the Internet. The litmus test will be whether or not the user community adopts the RLM and the layered codec as the preferred configuration.

We also plan to experiment with algorithms that dynamically adjust the bit-rate allocation of the different compression layers. Our compression scheme produces an *embedded code*, which has the property that any prefix of the compressed bitstream remains a valid representation at a lower quality. In other words, a given video frame can be successively refined at a very fine granularity. Using this property, we can partition the bit-rate arbitrarily among layers and

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Wireless Communications

Michele Zorzi, A. Chockalingam, in
Encyclopedia of Physical Science and Technology (Third Edition),
2003

V.C Quality of Service Support

With the advent of broadband networking and ATM, quality of service (QoS) provisioning has become a key concept in telecommunications (Schwartz, 1996). Typical QoS metrics in ATM networks are the packet loss rate, delay, and delay jitter (Onvural, 1995). Packet networks, unlike circuit-switched networks, are based on statistical multiplexing and resource sharing, and are prone to congestion, which occurs when the instantaneous users' demand collectively exceeds the network's capabilities. A major concern in broadband networking is to try to avoid congestion situations as much as possible and to cope with them when they occur. Admission control techniques are used in order to determine whether or not incoming connection requests can be accepted (Acampora, 1994a). A connection request can only be accepted if its QoS requirements can be met and if the QoS requirements of all connections already admitted can be maintained. This concept of admitting users based on a QoS calculation at the time

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Congestion Control in High-Speed Networks

Subir Varma, in Internet Congestion Control, 2015

5.2 Design Issues for High-Speed Protocols

In networks with large link capacities and propagation delays, the congestion control protocol needs to satisfy the following requirements:

1. It should be able to make efficient use of the high-speed link without requiring unrealistically low packet loss rates. As pointed out in the introduction, this is not the case for TCP Reno, because of its conservative window increase and aggressive window decrease rules.
2. In case of very high-speed links, the requirement of intraprotocol fairness between the high speed TCP protocol and regular TCP is relaxed because regular TCP is not able to make full use of the available bandwidth. Hence, in this situation, it is acceptable for the high speed TCP variants to perform more aggressively than standard TCP.
3. If connections with different round trip latencies share a link, then

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Control of Networks

Jean Walrand, Pravin Varaiya, in
High-Performance Communication Networks (Second Edition),
2000

8.1.5 Quality of Service

We said earlier that with better control strategies the network can provide more connections or calls with the same quality of service. We will make the concept of QoS more precise.

In the best case, the QoS guarantees very small cell (or packet) loss rate and delay. By very small cell loss rate, we mean a loss rate comparable to the loss rate due to unavoidable transmission errors. For example, suppose the bit error rate along the fibers is about 10^{-12} and that a cell has 424 bits (53 bytes). Then the fraction of cells lost because of transmission errors is on the order of $424 \times 10^{-12} \approx 10^{-10}$. This is the least cell loss rate that could be promised to users. The smallest delay that could be promised would be comparable to the propagation delay. For example, the propagation time of a cell from San Francisco to Boston is on the order of 10 ms. Thus, in the best case, the network could promise a cell loss rate of about 10^{-10} and a delay on the order of 10 ms for cells going from San Francisco to Boston.

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Physical layer design

Husheng Li, in
Communications for Control in Cyber Physical Systems, 2016

8.1 Introduction

We now consider the lowest layer of the communication network in cyber physical system (CPSs), namely the physical layer. Usually the main tasks in the physical layer consist of modulation and coding, which were introduced in Chapter 2. In modern communication systems, modulation and coding are usually independent of upper layer activities. However, as we will see, if the modulation and coding take into consideration the current physical dynamic state, this can improve the performance of controlling the physical dynamics, at the expense of a more complex system design. In this chapter, we ignore the complexity of system design, which is an important concern in practical system design, and study how the physical layer can be made adaptive to the physical dynamics and how much performance gain we can achieve compared with existing communication systems.

8.1.1 Modulation

As mentioned previously, the task of modulation is to convert information bits into physical symbols. Modern communication systems, particularly wireless systems, can make the modulation scheme adaptive to the communication channel condition, thus making a tradeoff between the communication rate (a denser constellation results in a higher rate) and reliability (a denser constellation results in less reliable transmission). In the context of a CPS, the different communication rates and reliabilities can be interpreted as different packet delays and different packet loss rates. In Section 8.2 we will study the effects of delay and packet loss rate on control performance and then propose a scheme to select the appropriate modulation scheme in an adaptive manner.

8.1.2 Coding

In communications, coding consists of source and channel coding. Both will be discussed in the context of CPSs in this chapter.

Source coding

We know that the purpose of source coding is to convert the original information source into information bits. The main challenge of source coding in a CPS is how to adapt the source coding scheme to the physical dynamics. In Section 8.3 we consider the point-to-point case,

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eHealth: Enabling technologies, opportunities and challenges

Hela Makina, ... Abderrezak Rachedi, in Advances in Computers, 2023

4.5.1 5G QoS indicators

In this section, we describe the key performance indicators that have been defined by the ITU to specify, quantify, and measure the characteristics of 5G systems [72]. Then, we will use these KPIs to characterize eHealth services. Table 4 defines the eight basic indicators and gives their associated traffic category.

Table 4. ITU key performance indicators for 5G networks.

KPI	Description	Use case
Peak data rate (in Gbit/s)	Maximum attainable data rate per user/device under perfect conditions.	eMBB
User experienced	Reachable data rate that is accessible	eMBB

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Related terms:

Internet of Things, Routing Protocol, Wireless Sensor Network, Congestion Control, Quality-of-Experience, Round-Trip Time, Transmission Control Protocol, Sensor Node, Energy Consumption, cluster-head.

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