Transmission Delay Modeling of Packet Communication over Digital Subscriber Line

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Abstract. *Certain multimedia and voice services, such as VoIP, IPTV, etc., are significantly delay sensitive and their performance is influenced by the overall transmission delay and its variance. One of the most common solutions used in access networks are xDSL lines, especially ADSL2+ or VDSL2. Although these subscriber lines also use packet communication, there are several differences and mechanisms, which influence their resulting delay. Their delay characteristics are also dependent on the individual settings of each xDSL provider, therefore we decided to investigate this area for typical commercially available lines in Czech Republic. Based on the measured values and experiments with real ADSL2+ lines we also developed a potential modeling method, which is presented in this article as well. The parameters for packet jitter based on the generalized Pareto distribution were modeled.*

Keywords

ADSL2+, digital subscriber line, packet jitter modeling, transmission delay.

1. Introduction

Today, the local area data networks and also access telecommunication networks are mostly packet oriented solutions and are usually based on the Ethernet conception. Since the packet oriented networks based on the Ethernet are now dominating the local area networks and access networks, the modeling of their delay characteristics is important today. These networks are also used for transmitting the time-division multiplex signals and services, which are significantly delay sensitive, their delay and jitter characteristics are very important [1], [2]. The Plain Old Telephone Service (POTS) is being replaced by Voice over IP (VoIP) [3], IPTV (Internet Protocol Tele-Vision) service is used for broadcasting TV signals and also a structured E1 signals can be transmitted over packet networks [4].

The digital subscriber lines (xDSL) represent still the most frequent technology used in access networks in Czech Republic. The previous generation of Asymmetric Digital Subscriber Lines (ADSL2+) is now being replaced by modern Very-High Speed Digital Subscriber Lines (VDSL2) [5]. Moreover, both types are considered for hybrid metallic-optical FTTx (Fiber to the "x") access networks, therefore these lines will probably remain one of the most common network solutions in access networks segment. However, the delay characteristics of commercially available xDSL lines on the market are influenced by many factors and mechanisms. First, there are standard packet-based operations, next the specific xDSL processes and finally, the higher communication layer mechanisms, such as the aggregation process or prioritization tasks. The value of the summary delay may differ for various subscriber lines during different time of a day period and is also influenced by the disturbances and noises (crosstalk, pulse interference) in xDSL channel, the current load of xDSL line etc.

However, we focused our investigation on the possibility of using generalized Pareto distribution for modeling of the summary delay of xDSL lines. First, we performed several tests and real measurements with commercially available ADSL2+ lines to obtain the initial data for following statistical processing. Next, we applied the generalized Pareto distribution formula and the conclusions and comparisons are presented within the last section of this paper as well.

2. Typical Parameters of DSL

2.1. Physical Layer

Both ADSL2+ and VDSL2 use Discrete Multitone modulation (DMT), the Vectored DMT (VDMT) modulation for far-end crosstalk elimination will be probably used for upgrading the VDSL2 lines in a future [5]. Both modulation techniques are based on many narrow frequency sub-channels with the width of 4,3125 kHz, which are independently used for transmitting the data bits using QAM (Quadrature-Amplitude Modulation). The sub-channel width of VDSL2 can also be double (8,625 kHz) for the frequency profile to 30 MHz Several methods for error correction and indication can be applied, such as Reed-Solomon forward error correction code, cyclic redundancy check, data interleaving, etc. The negative effect of these methods is increasing the delay necessary for processing the transmitted data at both receiver and transmitter side. The DMT modulation itself is realized by using Fourier transform, which also add some delay.

The Power Spectral Density masks (PSD) define the maximum transmit signal power for a specific frequency band [5]. These masks are defined in ITU-T recommendations and there are several dozen of masks for different regions, so that the individual network operators could select the most optimal. The VDSL2 lines also come with several different spectrum profiles, which specify the maximum frequency band and also the distribution of upstream (US) and downstream (DS) frequency bands. The resulting transmission rates of real ADSL2+ and VDSL2 lines are heavily dependent on the real conditions of used metallic cables and lines [6]. The attenuation limits the maximum used frequency, while noises and disturbances negatively affect the amount of allocated bits in each subchannel [7]. Crosstalk and especially far-end crosstalk (FEXT) is usually the most dominant source of noise [5]. Transmission performance of xDSL lines can be theoretically calculated with analytical models [8].

2.2. Data Performance at Higher Layers

The performance of xDSL lines at the physical layer can be estimated by using models of transmission lines and systems [8], however, the transmission rates at higher layers are lower due to the protocol header and service communication. Today, mostly TCP/IP and Ethernet protocols are used in data networks, therefore the data encapsulation used in xDSL lines is usually based on Point-to-Point over Ethernet (PPPoE) scenario.

The resulting transmission performance of a real xDSL line can be also influenced by other factors, mainly by the process of aggregation. The aggregation network node performs an aggregation task with all data to prevent overloading of backbone telecommunication network of xDSL service provider. Typical aggregation ratio is 1:50 today. All these mechanisms also negatively influence the delay characteristics of xDSL lines and increase their frame loss due to the aggregation process.

2.3. Real Delay and Packet Loss

The typical delay and packet loss characteristic was measured for a real ADSL2+ line to obtain some initial data for following statistical processing and modeling. The ADSL2+ line was a standard and commercially available line with real transmission rates of 8 Mbps in downstream and 384 kbps in the upstream direction. The ADSL2+ line was performed in fast mode without interleave, i.e. with minimal delay on the physic layer. The measurements were performed for a situation presented in Fig. 1.

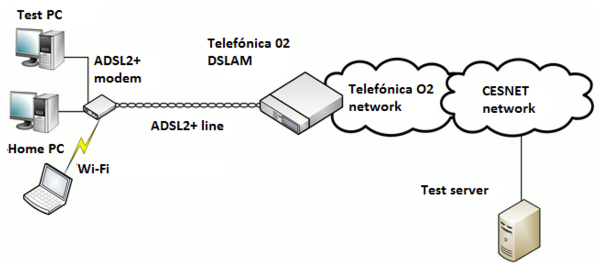

Fig. 1: Test scenario with Telefonica O2 ADSL2+ line.

The real delay and packet loss were continuously measured during 24 hour test. The test simulated real behavior of an ordinary household with 2 PCs connected via ADSL2+ home gateway (via Telefonica O2 ADSL2+ line) during the whole day, such as e-mail communication, watching online video streaming services, online gaming, web browsing, etc.

The summary communication delay in loop (RTT-Round Trip Time) was measured between the test PC and a server connected via gigabit CESNET network using three different lengths of transmitted IP packets - 32 bytes, 500 bytes and 1350 bytes for simulating the influence of a packet length on resulting transmission characteristics.

The resulting RTT for all three packet lengths is presented in Fig. 2, while Fig. 3 contains the illustration of packet loss.

The delay in the loop of real ADSL2+ line used for our experiment was oscillating between 11–13 ms for 32 byte packets, 20–22 ms for 500 byte packets and 38–40 ms for 1350 byte packets during the late night period, when no other data were transmitted. On the other hand, during the day and especially evening period with extensive ADSL2+ line load, the delay significantly increased and moreover it was jumping up and down dramatically.

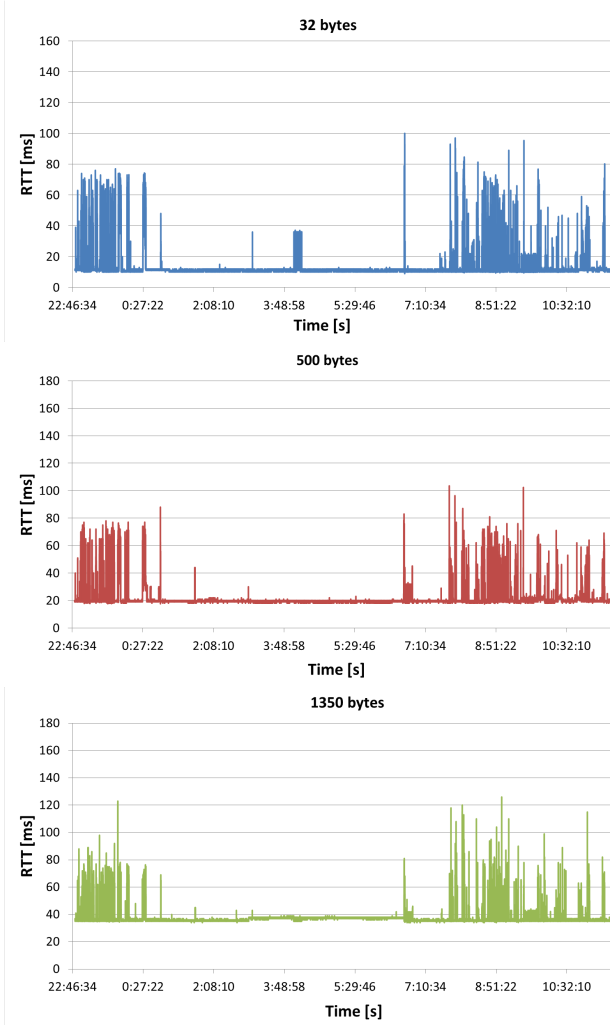

Fig. 2: Measured RTT for different packet lengths.

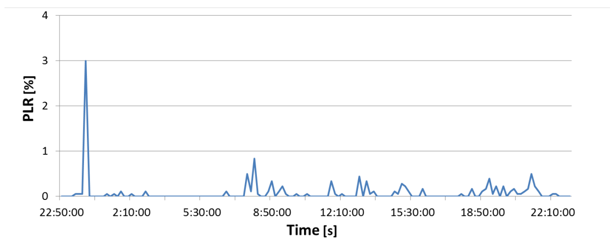

Fig. 3: Illustration of packet loss for all packet lengths.

The measured packet loss was also evident during the late evening and night period, when the backbone network of Telefonica O2 xDSL provider was overloaded and therefore the impact of aggregation rules were noticeable.

3. DSL Packet Delay Analysis

3.1. DSL Layer Delay

The xDSL layer delay is mostly caused by the xDSL mechanisms and operations with transmitted signals. The task of these mechanisms is to provide additional protection of transmitted data against noises (crosstalk, pulse interference) and disturbances, which can easily occur during the transmissions. That is why several different solutions were implemented, such as Reed-Solomon forward error correction, cyclic redundancy check, interleaving, scrambling, etc.

The second reason of additional xDSL layer delay is caused by the operations necessary for DMT modulation. The main source of delay in this category is the Fourier transform, which is performed upon a block of parallel data. Also the trellis coding operation and using cyclic prefix for separating transmitted frames add specific delay. However, these delays are usually constant and independent on the transmitted data and their amount.

Today, thanks to the relatively powerful xDSL processors implemented in DSLAMs and modems, these delays caused by xDSL operations are very low and can be usually ignored.

3.2. Total Process and Transmission Delay

The most significant part of the summary delay consists of operations performed at layer no. 3 (IP packet layer), which especially produces the packet jitter, i.e. delay variation. The summary transmission delay can be estimated using Eq. (1) for upstream or downstream [4] as:

$$t_t = t_s + t_{tx} + t_p + t_b, \tag{1}$$

where t_s is a serialization data link delay and can be calculated by the equation:

$$t_s = \frac{8n_p}{B_l}, \tag{2}$$

in which n_p is a length of packet in bytes (usually between 8 and 1490) and B_l is a link bit rate in bps [10].

t_{tx} is a transmission delay, which can be estimated from the distance d between all network nodes and the velocity of signal propagation v as:

$$t_{tx} = \frac{d}{v}, \tag{3}$$

t_p is a delay necessary for xDSL operations both at the ATU-R modem and ATU-C DSLAM side (typically several or tens milliseconds) including forward error correction (FEC), interleaving, etc.

t_b is a delay used for buffering transmitted packets in network nodes. This delay is a pseudo-random value with probability density function given by the Eq. (4):

$$PDF(t_b) = \frac{1}{\sigma\left(1+\frac{\xi(t_b-\mu)}{\sigma}\right)^{\frac{1}{\xi}+1}}, t_b \geq \mu, \qquad (4)$$

where μ, σ and ξ are parameters of generalized Pareto distribution [1]. Some other references [2] suggest using generalized Gamma distribution for the variation of this delay.

Variable delay can be compensated using the jitter buffer in terminals. This helps to reduce jitter, but on the other hand increasing the constant delay component.

The summary delay of a specific link is usually expressed as a round trip delay (RTT) representing the summary value of a delay in upstream t_{US} and downstream t_{DS} transmission directions:

$$t_{RTT} = t_{US} + t_{DS}. \qquad (5)$$

4. Transmission Delay Measurement and Modeling

4.1. Measured Results

The measurements of a real RTT and packet loss were performed on ADSL2+ line, which was described in chapter 2.3 and was presented in Fig. 2. The RTT tests were performed again during a 24 hours experiment with typical subscriber activity and for the same lengths of transmitted packet - 32 bytes, 500 bytes and 1350 bytes. The resulting RTT in a form of frequency quantity analysis for all three lengths of packets is presented in Fig. 4.

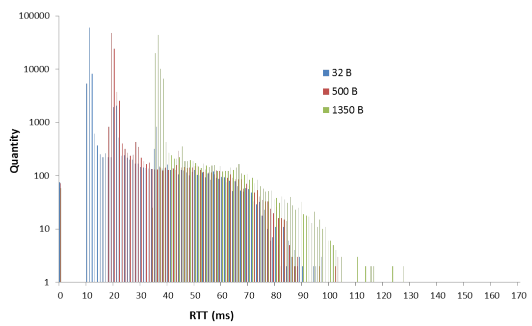

Fig. 4: Measured RTT and its frequency quantity analysis for all three packet lengths.

4.2. Results of Modeling

Next, the generalized Pareto distribution with Eq. (4) was applied with parameters presented in Tab. 1.

Tab. 1: Parameters of generalized Pareto model for packet jitter.

Parameter	μ[ms]	σ[ms]	ξ[-]
Value	$-0,3$	$0,35$	$1,4$

These values were subsequently used for modeling the density of a delay variation, and the result is presented in Fig. 5. The measured values were derivate from RTT (Fig. 4), so that the constant delays ts and tp has been subtracted and very small value of t_{tx} has been ignored. The chart shows that a single model is applicable to all considered lengths of packets.

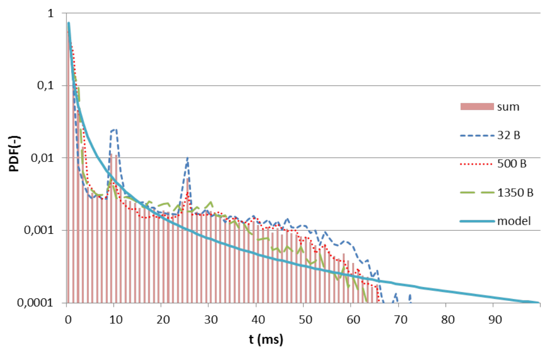

Fig. 5: Modeled probability density function of a delay variation.

The model characteristic was obtained by an approximation of measured values by aggregation of a variable delay part for all lengths of transmitted packets. These values for upstream direction are presented in Tab. 2, while Tab. 3 contains values for downstream transmission direction.

4.3. Impact of Delay on Frame Loss

The cumulative distribution function CDF of generalized Pareto distribution is defined as:

$$CDF(t_b) = 1 - \frac{1}{\sigma\left(1+\frac{\xi(t_b-\mu)}{\sigma}\right)^{\frac{1}{\xi}}}, t_b \geq \mu. \qquad (6)$$

The comparison between measured and modeled results is presented in Fig. 6.

Considering maximum tolerable jitter for a specific multimedia service, in case of its exceeding, the transmitted packets will be dropped and lost.

Tab. 2: Particular values of variable delay characteristic for upstream direction.

Delay	t_t	t_s(32B)	t_s(500B)	t_s(1350B)	t_{tx}	t_p
Value[ms]	$6-26$	$0,5$	$7,7$	21	$0,5$	5

Tab. 3: Particular values of variable delay characteristic for downstream direction.

Delay	t_t	t_s(32B)	t_s(500B)	t_s(1350B)	t_{tx}	t_p
Value[ms]	$5-7$	$0,03$	$0,5$	$1,3$	$0,5$	5

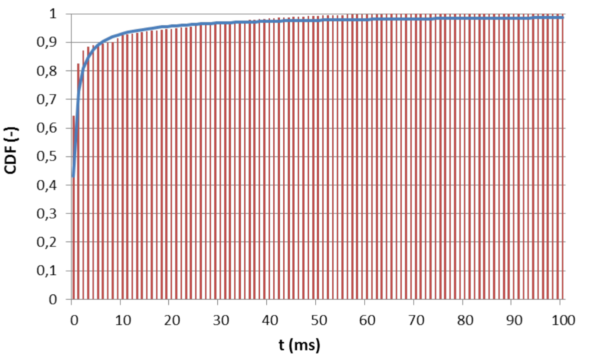

Fig. 6: Cumulative distribution function and its comparison with measured values.

The summary packet loss depends on the maximum jitter excess probability. This can be expressed using CDF as:

$$PLR(t) = 1 - CDF(t). \quad (7)$$

The resulting dependence of estimated packet loss on the delay variance is illustrated in Fig. 7. If for example a maximum tolerable delay in the upstream direction is 50 ms, its variable part is approx. 24 ms, which, according to the graph in Fig. 7, corresponds with a frame loss of 4 %.

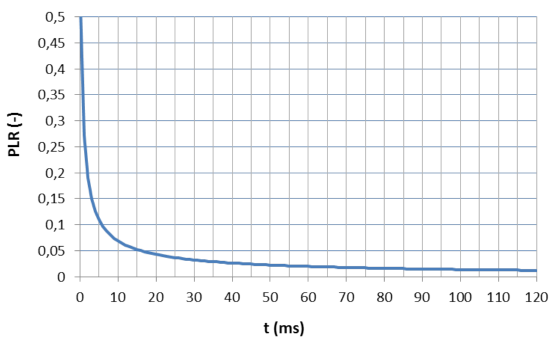

Fig. 7: Estimated dependence of a packet loss on maximum delay variance.

In this way, the calculated loss must be understood as an additional component of packet loss to basic packet loss of data link (for example presented on Fig. 3), where ADSL is operated for real time services.

5. Conclusion

This article dealt with the problem of a delay of digital subscriber lines and its modeling. The first part of the paper was focused on the introduction and analyses of transmission mechanisms and their impact on delay characteristics of digital subscriber lines. Next, several real measurements and experiments with commercially available ADSL2+ subscriber lines were performed. These tests were focused on RTT analysis and its jitter and were subsequently used for calculating statistical parameters used for following models.

These models are based on the generalized Pareto distribution and they verified that the variable part of RTT can be approximated using the general formulas of this distribution. Thanks to that, the impact of a delay on the packet loss could be analyzed, which can significantly help with the estimations of packet loss and its dependence on the delay and jitter characteristic of a specific digital subscriber line. This is useful mainly for various multimedia services, such as VoIP [9], IPTV, etc.

For real-time services, the variable component of delay must be compensated using the jitter buffer in terminals. This helps to reduce jitter, but on the other hand increasing the constant delay component. But also, if the current value of delay exceeds the jitter buffer size, there is an additional packet loss. The packet jitter analysis and modeling can help to correct design of jitter buffer size and other network components.

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Jiri VODRAZKA was born in Prague, Czech Republic in 1966. He joined the Department of Telecommunication Engineering, Faculty of Electrical Engineering, Czech Technical University in Prague in 1996 as a research assistant and received his Ph.D. degree in electrical engineering in 2001. He has been the head of the Transmission Media and Systems scientific group since 2005 and became associate professor in 2008. He participates in numerous projects in cooperation with external bodies. Currently he also acts as vice-head of the Department.

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