Analytical Model for Virtual Link Provisioning in Service Overlay Networks

Piotr Krawiec, Andrzej Bęben and Jarosław Śliwiński

***Abstract*—In this paper, we propose analytical model of Virtual Link used in the Service Overlay Networks. The Virtual Link exploits Selective Repeat ARQ scheme with time constrained number of retransmission and the playout buffer mechanism. Our model allows deriving equations that express trade-off between loss and delay characteristics experienced by packets transferred through VL. The main innovation of our model is the ability to cope with variable delay experienced by packets transferred by underlying network. Following the analytical model, we propose a method for Virtual Link dimensioning. The accuracy of the proposed model and dimensioning method is illustrated by simulation results.**

I. Introduction

THE Service Overlay Networks (SON) [1] operate at the application layer to offer new services in the Internet, such as QoS [2], reliability [3], multicast [4], privacy [5], etc. The nodes in SON, so called overlay nodes that are connected using underlying network, perform service specific functions related to both packet forwarding and service control. Since transfer characteristics of underlying network are usually not adequate for SON requirements, the overlay nodes engage additional mechanisms to adjust packet transfer characteristics to specific SON needs. This concept, called Virtual Link (VL), was introduced in [2] and then it was extended by several authors, e.g., [6], [7]. These studies show how the ARQ (Automatic Repeat reQuest) and/or FEC (Forward Error Correction) mechanisms applied at VL recover lost packets and finally improve the quality of transferred VoIP or video streams. The VL concept was enhanced in [8], where authors applied hybrid ARQ scheme jointly with the playout buffer mechanism to not only recover lost packets, but also to mitigate packet delay variation. Such improvement was achieved at the expense of reduced capacity and increased packet transfer delay.

In this paper, we introduce analytical model for VL with the Selective Repeat ARQ scheme and the playout buffer mechanism. Although the anaysis of delay characteristics of ARQ schemes have been already presented in literature, e.g., in [9], [10], [11], [12], [13], [14], they are based on the assumtion of constant round trip time. The main novelty of our model are: (1) the ability to cope with variable transfer delays between sender and receiver (2) time limited number of retransmissions. These features originate from characteristics of underlying network, where packet transfer delays are usually described by parameters of a random variable. As a consequence, VL behaves similar to a queueing system with randomly delayed feedback. Such model has not been solved yet, therefore different approximations are considered, e.g., [15]. Our model allows to approximate the distribution of packet transfer time starting from the moment when a packet arrives to VL at the sender side until the moment when the packet leaves the receiver side or until it is lost due to exceeding the threshold of packet transfer delay. On that basis, we are able to express VL characteristics as a function of packet transfer characteristics of underlying network and the assumed transfer time threshold. Using this analysis we propose a method for dimensioning of the VL.

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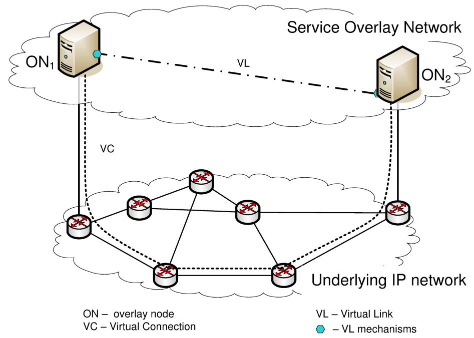

Fig. 1. The Virtual Link concept.

The paper organisation is the following. In Section II, we introduce the VL concept. Then in Section III, we present proposed analytical model of the VL jointly with simulation results showing its effectiveness. In Section IV, we propose VL dimensioning method, which takes advantages of proposed analytical model. Finally, Section V summarises the paper and gives outline of further works.

II. Virtual link

The SON concept assumes that overlay nodes connect each other by Virtual Connections (VC), which are offered by underlying network, as presented on Fig. 1. Since usually, there is no direct relation between SON and underlying network, the overlay nodes engage additional mechanisms, called Virtual Link (VL), to adjust packet transfer characteristics offered by VC to SON needs. As proposed in [2], [8], the VL engages the selective repeat ARQ mechanism supported by the playout buffer mechanism. The selective repeat ARQ mechanism recovers lost packets at the expense of increased packet transfer delay and reduced link capacity. On the other hand, the playout buffer mechanism enforces the same delay for each packet transferred through VL, which emulates the behaviour of an ordinary synchronous link.

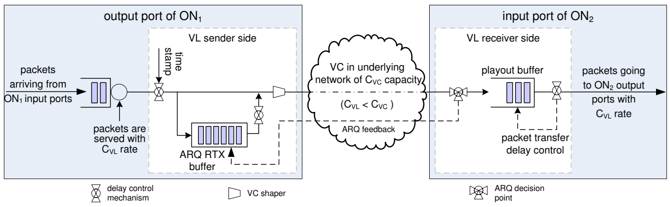

Fig. 2. The mechanisms of Virtual Link.

Fig. 2 presents exemplary VL, which is established between neighbouring overlay nodes ON_1 and ON_2. The packet arriving to the output buffer in ON_1 enters the queue, which is served with the rate of VL, named C_{VL}. The VL begins packet service by the ARQ mechanism, where the copies of particular packets are stored in ARQ retransmission buffer (ARQ RTX). To each packet, we add time stamp with its arrival time to VL. We used this time stamp at the receiver side to recover traffic profile in the playout buffer. The receiving side controls the sequence of the incoming packets and it sends acknowledgements for received packets as well as requests for retransmissions for lost packets. If a packet is lost, then the sending side retransmits it upon receiving retransmission request or expiration of the time-out. The number of retransmissions is limited by value D_{max}, which defines the time limit for successful packet delivery. At the receiver side, we put received packets into the playout buffer. The playout buffer delays the packet departure to allow for retransmissions of previously lost packets and mitigate the variable packet transfer time in VL. Moreover, playout buffer recovers the sequence and the inter-packet gaps of transferred packets, based on their timestamps. When a packet arrives too late, the playout buffer simply drops it. More detailed description of VL mechanisms is presented in [8].

The starting point in the VL analysis are characteristics of VC. They correspond to available capacity, denoted as C_{VC}, and the packet transfer characteristics expressed in terms of QoS metrics [16] such as: 1) minimum IP Packet Transfer Delay, $minIPTD_{VC}$, 2) IP Packet Delay Variation, $IPDV_{VC}$, jointly with random variable describing random part of IP Packet Transfer Delay, as well as, 3) IP Packet Loss Ratio, $IPLR_{VC}$. These data may come from contracts agreed with Internet Service Provider or from the measurements performed by overlay nodes. Anyway, in our analysis, we left the problem of gathering VC characteristics for further studies, assuming that VC characteristics are known *a priori*.

Summarising, the VL mechanisms give a trade-off between packet transfer delay and packet transfer loss characteristics provided by VL. In principle, greater value of $IPTD_{VL}$ allows VL mechanisms for more retransmissions what improve packet loss characteristics but on the other hand might not be acceptable for delay sensitive traffic. This trade-off may be expressed by generic equations (1), (2) and (3). Equation (1) defines the value of $IPTD_{VL}$ experienced by packets transferred through VL

$$IPTD_{VL} = minIPTD_{VC} + D_{max} = const, \qquad (1)$$

where $minIPTD_{VC}$ is the minimum value of packet transfer delay experienced on VC and D_{max} denotes the time limit for successful packet delivery through VL. This time limit determines number of packet retransmissions. Equation (2) defines worst case limit of $IPLR_{VL}$ as a function of packet loss ratio on VC, defined by $IPLR_{VC}$ and the minimum number of packet retransmissions

$$IPLR_{VL} = IPLR_{VC}{}^{1+\left\lfloor \frac{D_{max}}{RTT_{max}} \right\rfloor}, \qquad (2)$$

where RTT_{max} is the maximum value of round trip time experienced by packets transferred through VC and turned around to the source by reverse VC.

Note that every retransmission reduces the effective value of C_{VL}. Therefore, we can approximate the allowed capacity of VL by the value corresponding to the infinite number of retransmissions

$$C_{VL} \leq C_{VC} * (1 - IPLR_{VC}). \qquad (3)$$

Taking into account that implementation of the VL requires introducing some overhead in the form of a VL header, we obtain the final value of allowed capacity for the VL by multiplying C_{VL} from equation (3) by expression ${}^{L_d}/_{(L_d+L_{VL})}$, where L_d denotes size of packet arriving to the VL, and L_{VL} is the VL header length.

Remark that presented above equations are valid for the VL, which uses selective repeat ARQ scheme and playout buffer mechanism.

III. Proposed model

The Virtual Link features a retransmission scheme combined with delay based decision process. Consequently, the model for performance analysis has to take into account the correlation between retransmissions and the packet transfer characteristics between sender and receiver (in both directions). Our analysis aims to derive the packet transfer characteristics of the Virtual Link expressed by packet transfer delay $IPTD_{VL}$ and packet loss ratio $IPLR_{VL}$ with regard to its assumed capacity C_{VL} and retransmission delay limit D_{max}.

A. Definition of model

We assume that Virtual Connection has the capacity C_{VC} and has different propagation delays for data and acknowledgement packets, which are t_{pd} and t_{pa}, respectively. Furthermore, we assume that data (acknowledgement) packets are of constant size L_d (L_a) with transmission delay $t_{Xd} = {}^{L_d}/_{C_{VC}}$ ($t_{Xa} = {}^{L_a}/_{C_{VC}}$). Sum of propagation delay t_p and transmission delay t_X equals $minIPTD_{VC}$ metric. Moreover, the random variable X_i (Z_i) defines the variable part of packet transfer delay in the *data* (*acknowledgement*) direction. We presume that capacities of VCs are significantly lower than capacities avaiable in underlying network, and hence there is no correlation between handling, in underlying nodes, of two consecutive packets transferred through given VC. It allowes us to neglect correlation between losses and delays of consequtive packets transferred through VC. Therefore, the packet losses may follow independent model with loss probability

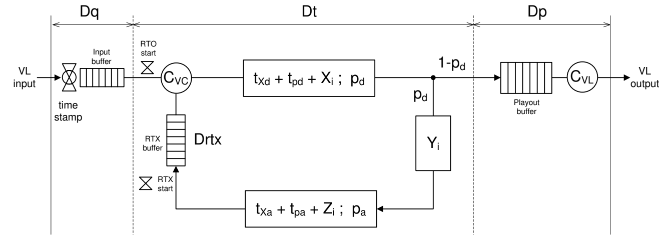

Fig. 3. The Virtual Link model.

p_d (p_a) in *data* (*acknowledgement*) direction. Additionally, the above assumptions implies that random variables defining packet transfer delay (X_i and Z_i) and packet losses are all independent of each other.

The packets arriving to the Virtual Link have constant bit rate profile with bit rate C_{VL} (constant packet inter-arrivals). We assume, that the sequence number space ARQ mechanism utilises, is large enough to not stop the packet sending process.

The overview of the Virtual Link model is presented in Fig. 3. The variables used in the model are the following:

- Dq – random variable which describes the waiting time in the input buffer for "fresh" packets. Dq duration is defined by the time instant when a packet enters the input buffer and by the time instant when its service begins, i.e., the first transmission in Virtual Connection.
- Dt – random variable which describes the packet transfer delay from the sender to the receiver (in Virtual Connection). It is defined as time interval from the start of first transmission of the "fresh" packet in Virtual Connection, until the moment of arrival of the first packet copy to the receiver. Virtual Link drops packets having no chance for reception before the deadline D_{max}; therefore, we assume that for dropped (i.e., lost) packets Dt takes infinite value.
- Dp – random variable which describes the duration of packet's stay in the playout buffer.
- $Drtx$ – random variable which describes a queueing delay of retransmitted packets. We assume that retransmitted packets access the Virtual Connection with higher priority than "fresh" packets.

According to the Virtual Link concept, assuming the packet is not lost, its total transfer delay in VL is constant and it is given by (see eq. 1)

$$IPTD_{VL} = t_{Xd} + t_{pd} + Dmax = Dq + Dt + Dp \quad (4)$$

Random variables Dq and Dt are constrained by $IPTD_{VL}$

$$Dq + Dt \leq IPTD_{VL}. \quad (5)$$

Duration Dp, which complements the sum of Dq and Dt to constant time $IPTD_{VL}$, varies from 0 to $Dmax$

$$\begin{aligned} Dp &= IPTD_{VL} - Dq - Dt, \\ Dp &\in [0; Dmax]. \end{aligned} \quad (6)$$

Let P_s defines the probability of the event that packet is successfully transferred by Virtual Link in time t i.e., the packet's copy arrives to the receiver before t

$$P_s = Pr\{Dq + Dt \leq t\}. \quad (7)$$

In the Virtual Link, the rate of incoming packets is limited by capacity C_{VL}. Next, the packets are taken for service from input queue with rate $C_{VC} > C_{VL}$ or they wait in the input queue until service of retransmitted packets from RTX buffer is finished. Because packet losses higher than a few percent are rather rare in normal operation of network ([17]), for further analysis we assume that queueing time Dq is negligible compared to Dt. Consequently, we can state that

$$P_s = Pr\{Dt \leq t\}. \quad (8)$$

Thanks to the assumed independence between packet delay and packet losses characteristics, we can write the following formula which defines the conditional probability of packet's successful delivery in time not longer than t, assuming that there were i^{th} transmission attempts

$$\begin{aligned} Pr\{Dt \leq t\} &= \sum_{i=1}^{\infty} Pr\{Dt \leq t \wedge Tr = i\} \\ &= \sum_{i=1}^{\infty} Pr\{Dt \leq t \mid Tr = i\} Pr\{Tr = i\}, \end{aligned} \quad (9)$$

where Tr is the random variable describing number of packet transmissions (including retransmissions) until successful packet delivery to the receiver. Assuming independent packet loss model, Tr has a geometric distribution

$$Pr\{Tr = i\} = (1 - p_d) p_d^{(i-1)}. \quad (10)$$

Next, we analyse duration between time instant when the first transmission of a packet starts, until the successful reception after i^{th} transmission. Probability that packet transfer time is not greater than t, assuming success on the first attempt ($Tr = 1$), is given by

$$\begin{aligned} & Pr\{Dt \leq t \mid Tr = 1\} = \\ &= \frac{Pr\{t_{Xd} + t_{pd} + X_i \leq t \wedge Tr = 1\}}{Pr\{Tr = 1\}} \\ &= \frac{Pr\{t_{Xd} + t_{pd} + X_i \leq t\} Pr\{Tr = 1\}}{Pr\{Tr = 1\}} \\ &= Pr\{t_{Xd} + t_{pd} + X_i \leq t\} \end{aligned} \quad (11)$$

In this case the time Dt has two components: 1) data packet transmission time on VC link (t_{Xd}), and 2) variable propagation time through VC link ($t_{pd} + X_i$).

Probability that packet transfer time is not greater than t, assuming success on the second attempt ($Tr = 2$), equals

$$\begin{aligned} & Pr\{Dt \leq t \mid Tr = 2\} = \\ &= Pr\{min[RTO; Y_i + t_{Xd} + t_{pd} + X_i \\ &\quad + t_{Xa} + t_{pa} + Z_i] + Drtx + \\ &\quad + t_{Xd} + t_{pd} + X_i \leq t\} \end{aligned} \quad (12)$$

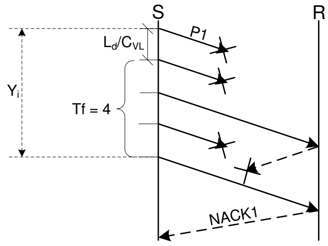

Fig. 4. An Y_i random variable.

In this case time Dt consists of the four components: 1) time required to detect lost packet (ARQ decision delay; it is defined as a minimum of a retransmission timeout RTO and time, after which NACK for lost packet arrives to a sender), 2) waiting time in the retransmission buffer ($Drtx$), 3) data packet transmission time on VC link (t_{Xd}), and 4) variable propagation time of retransmitted data packet through VC link ($t_{pd} + X_i$).

Time, after which NACK arrives to a sender, is a sum of following values (see Fig. 4): 1) time Y_i, 2) data packet transfer time ($t_{Xd} + t_{pd} + X_i$), and 3) transfer time ($t_{Xa} + t_{pa} + Z_i$) of acknowledgement for data packet, which carries NACK for lost packet.

Random variable Y_i denotes time required for the sender to transmit consecutive data packets, which are essential in order to receive an acknowledgement indicating the packet loss. Because packets arrive to VL with rate C_{VL}/L_d, r.v. Y_i can take the following discrete values (see Fig. 4):

$$Y_i = \frac{j \cdot L_d}{C_{VL}}, \quad j \in \{1, 2, \ldots\} \tag{13}$$

Let Tf be a random variable which describes number of consecutive packets, which must be sent by the sender to receive information about packet loss. Fig. 4 illustrates the case for $Tf = 4$. The Tf depends on the data and the ACK packets loss probabilities (p_d and p_a, respectively) and has geometric distribution (see Appendix A):none

$$Pr\{Tf = j\} = (1 - p_d)(1 - p_a)[p_d + (1 - p_d)p_a]^{(j-1)} \tag{14}$$

Using the law of total probability, we can rewrite the formula (12) as:

$$\begin{aligned} &Pr\{Dt \le t \mid Tr = 2\} = \\ &= \sum_{j=1}^{\infty} Pr\left\{min\left[RTO; \frac{j \cdot L_d}{C_{VL}} + \right.\right. \\ &\quad \left. +t_{Xd} + t_{pd} + X_i + t_{Xa} + t_{pa} + Z_i\right] + \\ &\quad \left. +Drtx + t_{Xd} + t_{pd} + X_i \le t \right\} \cdot \\ &\quad \cdot Pr\{Tf = j\} \end{aligned} \tag{15}$$

The SR ARQ scheme used in Virtual Link assumes, that only the first packet retransmission is controlled by NACK, while the second, third, etc., occurs after retransmission interval RTX, as it is shown on Fig. 5. Such approach allows us to maintain high responsiveness for the first retransmission and, at the same time, to impose the limit on the additional traffic generated due to retransmissions. Taking this feature into account, we can express the probability that packet transfer time is not greater than t, assuming that $i-1$ packet transmission attempts fail and there is a success in the i^{th} attempt ($i \ge 2$,) by

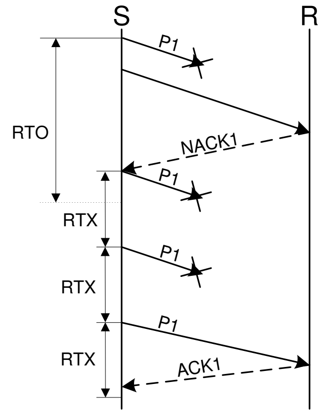

Fig. 5. The Virtual Link retransmission schema.

$$\begin{aligned} &Pr\{Dt \le t \mid Tr = i\} = \\ &= \sum_{j=1}^{\infty} Pr\left\{min\left[RTO; \frac{j \cdot L_d}{C_{VL}} + \right.\right. \\ &\quad \left. +t_{Xd} + t_{pd} + X_i + t_{Xa} + t_{pa} + Z_i\right] + \\ &\quad +(i-2)RTX + Drtx + t_{Xd} + t_{pd} + \\ &\quad \left. +X_i \le t \right\} \cdot Pr\{Tf = j\} \end{aligned} \tag{16}$$

Finally, according to the formula (none9) the probability, that Dt (successful packet transfer from the sender to the receiver) is not greater than t, assuming unlimited number of retransmission, has the following form

$$\begin{aligned} &Pr\{Dt \le t\} = \\ &= Pr\{t_{Xd} + t_{pd} + X_i \le t\}(1 - p_d)p_d^0 + \\ &\quad + \sum_{i=2}^{\infty}\left[\sum_{j=1}^{\infty} Pr\left\{min\left[RTO; \frac{j \cdot L_d}{C_{VL}} + \right.\right.\right. \\ &\quad \left. +t_{Xd} + t_{pd} + X_i + t_{Xa} + t_{pa} + Z_i\right] + \\ &\quad +(i-2)RTX + Drtx + t_{Xd} + t_{pd} + \\ &\quad \left.\left. +X_i \le t \right\} \cdot Pr\{Tf = j\}\right](1 - p_d)p_d^{(i-1)} \end{aligned} \tag{17}$$

where $Pr\{Tf = j\}$ is given by formula (14).

B. Model evaluation

We evaluate the analytical results, which are obtained using the proposed model, with results of simulations. The following assumptions are taken:

- since RTO timer is usually set to relatively high value, we consider RTO timeout expiration as exceptional, rare event, therefore the sender obtains information about packet loss mainly thanks to receiving negative acknowledgements;
- taking into account, that retransmitted packets have higher priority and packet losses at VC are usually a few percent at most, we consider queueing delay $Drtx$ of retransmitted packets as a negligible part of total packet transfer time Dt.

Therefore, we can write formula (17) as:

$$\begin{aligned} &Pr\{Dt \le t\} = \qquad (18)\\ &= \quad Pr\left\{t_{Xd} + t_{pd} + X_i \le t\right\}(1-p_d)p_d^0 + \\ &\quad + \sum_{i=2}^{\infty}\left[\sum_{j=1}^{\infty} Pr\left\{\frac{j \cdot L_d}{C_{VL}} + t_{Xd} + t_{pd} + X_i + \right.\right. \\ &\quad + t_{Xa} + t_{pa} + Z_i + (i-2)RTX + t_{Xd} + t_{pd} + \\ &\quad \left.\left. + X_i \le t\right\} \cdot Pr\left\{Tf = j\right\}\right](1-p_d)p_d^{(i-1)} \end{aligned}$$

Let consider the expression under the summation:

$$\begin{aligned} &\frac{j \cdot L_d}{C_{VL}} + t_{Xd} + t_{pd} + X_i + t_{Xa} + t_{pa} + \qquad (19)\\ &\quad + Z_i + (i-2)RTX + t_{Xd} + t_{pd} + X_i \end{aligned}$$

For fixed i and j we can write it as a sum of independent random variables and constants:

$$X_i + Z_i + X_i + a + \frac{j \cdot L_d}{C_{VL}} + (i-2)RTX \qquad (20)$$

where a is a constant value, which equals to $2(t_{Xd} + t_{pd}) + t_{Xa} + t_{pa}$.

Probability density function of Dt can be obtained as a convolution of *pdf*s for X_i and Z_i random variables and the constants, by exploitation of the Laplace transform ($\delta()$ denotes the Dirac delta function):

$$\begin{aligned} &f_{Dt}(t) = \mathcal{L}^{-1}\Big\{\mathcal{L}\{X_i\} \cdot \mathcal{L}\{\delta(t - t_{Xd} - t_{pd})\}\Big\} \cdot \qquad (21)\\ &\cdot Pr\{Tr = 1\} + \sum_{i=2}^{\infty}\left[\sum_{j=1}^{\infty} \mathcal{L}^{-1}\Big\{\mathcal{L}\{X_i\} \cdot \right. \\ &\mathcal{L}\{Z_i\} \cdot \mathcal{L}\{X_i\} \cdot \mathcal{L}\{\delta(t - a - \frac{j \cdot L_d}{C_{VL}} - \\ &\left. - (i-2)RTX)\}\Big\} \cdot Pr\{Tf = j\}\right] \cdot Pr\{Tr = i\} \end{aligned}$$

We calculate distribution of Dt using (21) for the case, when random variables which describe variable part of packet transfer delay through VC link are exponentially distributed with mean equals none$1/m_X$ for direction sender-to-receiver and $1/m_Z$ for direction receiver-to-sender, respectively: $X_i \sim Exp(m_X)$, $Z_i \sim Exp(m_Z)$. We assume that VC links for both directions are the same: $m_x = m_z$, $t_{pd} = t_{pa}$, $p_d = p_a$, with $C_{VC} = 1\,Mbps$ and two values of packet loss ratio: 5% and 0.5%. The VL bit rate $C_{VL} = 650\,Kbps$ and packet size $L_d = 200\,B$. We consider two cases:

- case#1: VC is characterised by low packet delay variation ($t_{pd} = t_{pa} = 25\,ms$, $m_x = m_z = 1\,ms$, $IPDV_{VC} = 7\,ms$);
- case#2: VC is characterised by relatively high packet delay variation ($t_{pd} = t_{pa} = 5\,ms$, $m_x = m_z = 5\,ms$, $IPDV_{VC} = 34.5\,ms$).

During simulation experiments we generate at least $4{\cdot}10^8$ packets for each considered case.

Fig. 6 and Fig. 7 depicts complementary cumulative distribution function of time Dt, calculated on the basis of formula (21), as well as obtained experimentally (solid line and dashed line, respectively), for both cases. We observe that the proposed model closely approximates the simulation results. The reason of small shift between the appropriate

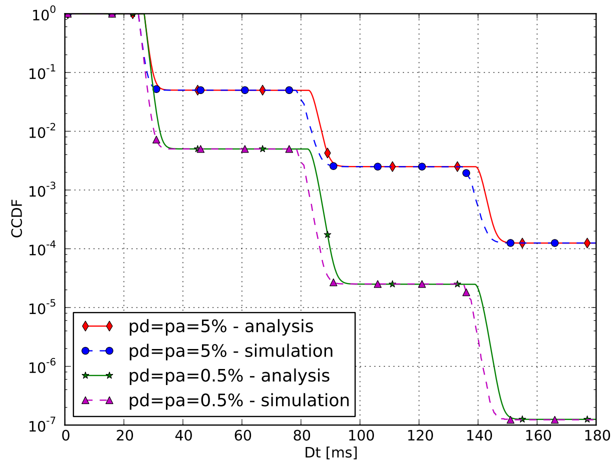

Fig. 6. Complementary CDF of time Dt for case#1: VC with low packet transfer delay variation.

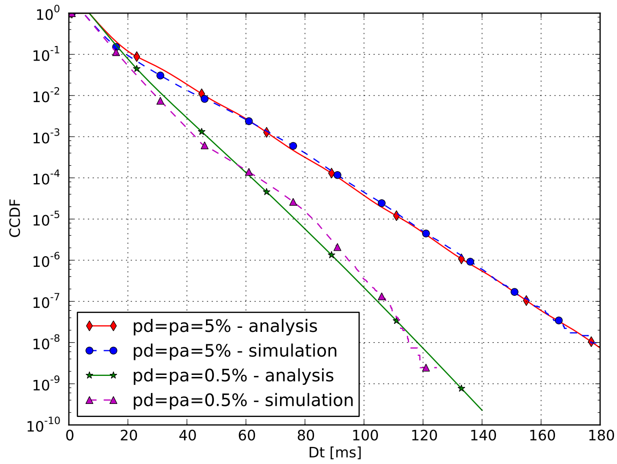

Fig. 7. Complementary CDF of time Dt for case#2: VC with high packet transfer delay variation.

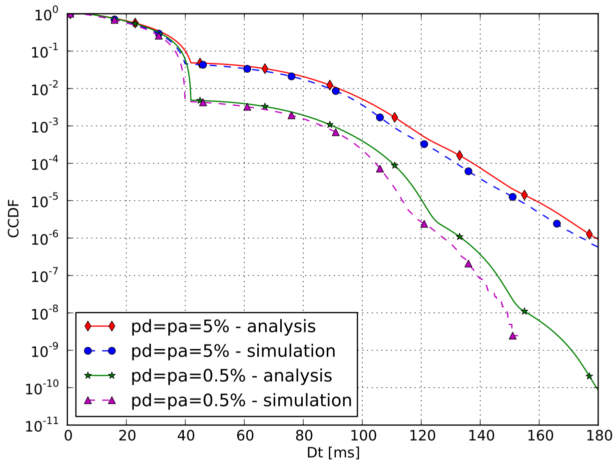

Fig. 8. Complementary CDF of time Dt, in case when X_i and Z_i random variables have uniform distribution.

curves, is that the approximation takes the worst case from the point of view of packet transmission time. We assume, that each packet is transmitted with rate limited by the VC shaper to C_{VC} (see Fig. 2). During simulations, such limitation occurred rarely, and transmission time for most packets was much shorter, because they were sent with physical link bit rate, a hundred times greater than VC bit rate. Consequently, the proposed model can be treated as an upper bound for Dt time distribution.

Notice, that in some intervals the simulated curves are above the calculated ones for greater values of Dt in case#2 (see Fig. 7). This is due to packet stream integrity condition, which we assumed during simulations. To avoid packet reordering, which is possible in high packet delay variation case, we allow only such values of single packet delay, generated according to exponential distribution, which keep packet order. In this way, distribution of X_i and Z_i random variables used in simulations differ slightly comparing to calculation.

In Fig. 8 we present complementary cumulative distribution function of time Dt for case#2, i.e. when VC is characterized by high packet delay variation, but when the random variables X_i and Z_i have a uniform distribution over interval [0 ms, 35 ms]. In this case, a distribution of packet transfer delay through VC link used for calculation was identical with the distribution applied in simulation experiments. Consequently, the analytical curve is above the simulated one, and can be treated as an upper bound for numerical results, as in case#1 (see Fig. 6).

IV. Application for Virtual Link dimensionig

The Virtual Link concept assumes, that packet transfer is successful, if its copy arrives to the receiver in time not greater than $IPTD_{VL}$:

$$P_s = Pr\{Dt \leq IPTD_{VL}\} \tag{22}$$

Therefore, the packet loss probability on VL link can be defined as:

TABLE I

$IPLR_{VL}$ for VC with low packet transfer delay variation (case#1).

$pd\,(=pa)$	$IPTD_{VL}$	$IPLR_{VL}$ analytical model	$IPLR_{VL}$ simulation
5%	$40\,ms$	$5{\cdot}10^{-2}$	$5{\cdot}10^{-2} \pm 4{\cdot}10^{-5}$
	$70\,ms$	$5{\cdot}10^{-2}$	$5{\cdot}10^{-2} \pm 3\cdot10^{-5}$
	$110\,ms$	$2.5{\cdot}10^{-3}$	$2.5{\cdot}10^{-3} \pm 5\cdot10^{-6}$
0.5%	$40\,ms$	$5{\cdot}10^{-3}$	$5{\cdot}10^{-3} \pm 6{\cdot}10^{-6}$
	$70\,ms$	$5{\cdot}10^{-3}$	$5{\cdot}10^{-3} \pm 9{\cdot}10^{-6}$
	$110\,ms$	$2.5{\cdot}10^{-5}$	$2.5{\cdot}10^{-5} \pm 3{\cdot}10^{-7}$

$$IPLR_{VL} = 1 - P_s = 1 - Pr\{Dt \leq IPTD_{VL}\} \tag{23}$$

However, exact computation of $Pr\,\{Dt \leq IPTD_{VL}\}$ is not trival, since in the VL number of possible transmission attempts for each packet is limited by time. Sender transmit packet only if it has chance to be received by receiver in the assumed time interval limit $IPTD_{VL}$. Othervise, packet is droped. Therefore, we propose to approximate an $IPLR_{VL}$ metric by calculating distribution $Pr\,\{Dt \leq t\}$ with assumption of unlimited number of retransmissions, as defined by equation (17). Next, we determine packet loss probability on the VL as a fraction of packets, for which transfer time Dt was greater than $IPTD_{VL}$:

$$IPLR_{VL} \approx Pr\{Dt > IPTD_{VL}\} \tag{24}$$

In tables I and II we present values of $IPLR_{VL}$ metric obtained from simulations and calculated analytically according to rule presented above. Simulations were performed for the same scenarios and the values of parameters as in section III-B. Table I refers to case#1, with low delay variation, whereas table II refers to case#2, with greater delay variation. Parameter $Dmax$ had appropriate values to obtain packet transfer delay in the VL equals to 40, 70 and $110\,ms$. The results were obtained by repeating the simulation tests 10 times and calculating the mean values with the corresponding 95% confidence intervals. For each iteration, we simulated at least $50{\cdot}10^6$ packets.

Results presented in table I show, that proposed method allows to determine packet loss ratio in the Virtual Link with high accuracy in the case, when variation of delay is relatively low comparing to constant part of packet transfer delay through underlying VC. In the case#2, for tests with higher values of $IPTD_{VL}$ and packet losses in VC equal to 0.5%, the $IPLR_{VL}$ measured in simulations is above the calculated values (see table II). This mismatch is similar to the values observed for analytical and simulated curves depicted in Fig. **??**. However, analytical results still constitute a good approximation for the $IPLR_{VL}$ obtained by simulations.

Summarizing, presented analytical method helps us to approximate packet loss ratio provided by the Virtual Link, taking as input data the packet transfer characteristics of VC, such as: bit rate C_{VC}, packet loss ratio p_d and p_a, constant (t_{pd}, t_{pa}) and variable (X_i, Z_i) transfer delay, as

TABLE II
$IPLR_{VL}$ FOR VC WITH HIGH PACKET TRANSFER DELAY VARIATION (CASE#2).

$pd\,(=pa)$	$IPTD_{VL}$	$IPLR_{VL}$	
		analytical model	simulation
5%	$40\,ms$	$1.9{\cdot}10^{-2}$	$1.4{\cdot}10^{-2} \pm 3 \cdot 10^{-5}$
	$70\,ms$	$9.1{\cdot}10^{-4}$	$1.1{\cdot}10^{-3} \pm 4{\cdot}10^{-6}$
	$110\,ms$	$1.3{\cdot}10^{-5}$	$1.7{\cdot}10^{-5} \pm 1 \cdot 10^{-7}$
0.5%	$40\,ms$	$2.8{\cdot}10^{-3}$	$1.7{\cdot}10^{-3} \pm 4 \cdot 10^{-7}$
	$70\,ms$	$2.8{\cdot}10^{-5}$	$5.4{\cdot}10^{-5} \pm 7 \cdot 10^{-7}$
	$110\,ms$	$4.0{\cdot}10^{-8}$	$6.6{\cdot}10^{-8} \pm 1.5 \cdot 10^{-8}$

well as assumed parameters of the VL: bit rate C_{VL} and delay $IPTD_{VL}$. Jointly with formula (3), which defines maximum allowed capacity of VL, it can be used for dimensioning of the Virtual Link. A procedure of the VL dimensioning we present below. Notice, that the VL admits a controlled trade-off between packet loss level, offered capacity and constant delay introduced by the VL.

A. Virtual Link dimensioning algorithm

Using the formula (24) to determine upper bound of packet loss ratio in the VL, together with the formula (3) for approximation of the allowed VL capacity, we can dimension the Virtual Link according to the following algorithm:

- Step 1: determine values of parameters for Virtual Connection, which is used for establishing the VL (bit rate of VC, packet loss ratio, minimum packet transfer delay on VC, distribution of the variable part of packet transfer delay on VC).
- Step 2: for given value of the VL capacity C_{VL}, which satisfies the condition described by formula (3), calculate distribution of packet transfer time between sending and receiving overlay node $Pr = \{Dt \leq t\}$ (according to formula (18)).
- Step 3: according to formula (24), find such value T_{VL} in obtained Dt time distribution, for which probability $Pr = \{Dt > T_{VL}\}$ equals required value of $IPLR_{VL}$. In case when the input parameter is value of $IPTD_{VL}$, from obtained Dt time distribution find value of $IPLR_{VL}$ as probability $Pr = \{Dt > IPTD_{VL}\}$.
- Step 4: calculate value of D_{max} parameter for the VL as the difference between time T_{VL} and minimum packet transfer delay on VC: $D_{max} = T_{VL} - minIPTD_{VC}$. In case when the input parameter is value of $IPTD_{VL}$, then $D_{max} = IPTD_{VL} - minIPTD_{VC}$.

Using the following steps, at the top of given VC we can establish the VL with bit rate C_{VL}, constant packet transfer time $IPTD_{VL}$ (= T_{VL}) and packet loss ratio not greater than $IPLR_{VL}$.

V. SUMMARY

In this paper we considered the performance analysis of Virtual Link. The Virtual Link is established between nodes of the Service Overlay Network in order to improve packet transfer characteristics of the underlying network. The presented analysis focuses in the packet transfer delay distribution as observed during handling in the VL when the retransmission delay limit is infinite. Comparing to other analytical methods for ARQ systems, our model features variable transfer delays between sender and receiver, and moreover, it uses delay-bound scheme for number of retransmissions. The accuracy of the proposed model was verified by means of simulation in exemplary scenario with exponentially distributed delay characteristics of the underlying network. The analytical and numerical results differ slightly due to worst case assumptions and due to the method of packet transfer delay emulation that maintains the order of packets in the underlying network (for the case with greater value of delay variation). Finally, we proposed a method for dimensioning of VL with finite retransmission delay limit that allows for controlled use of trade-off between packet transfer delay and packet losses.

One of the remaining problems, which is not directly related to the VL analysis, is the reliable characterisation of packet transfer characteristics in the Virtual Connection. In the situation when operator of underlying network do not provide required parameters of the VC, for example, in the form of an SLA (Service Level Agreement) contract, the SON owner can obtain them using measurements. Those measurements can be performed by external tools, such as OWAMP (One-Way Active Measurement Protocol) tool [18], or by internal measurement module integrated with the VL. In the latter case, the packet transfer characteristics of the VC can be measured by means of a passive measurement method. For this purpose, we can use timestamps and sequence numbers, which are carried in each VL packet header, to determine packet transfer delay and loss characteristics.

In the further work we will extend the proposed model to include the FEC mechanism.

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APPENDIX A: TF R.V. DISTRIBUTION

Random variable Tf describes number of consecutive packets, which must be sent by the sender to receive information about lost packet.

R.v. Tf takes value 1, if the first packet sent after lost packet, reaches a receiver, and sender receives acknowledgement for that packet (which contains NACK for lost packet) - see Fig. 9.

Probability, that Tf is equal 1, is:

$$Pr\{Tf = 1\} = (1 - p_d)(1 - p_a) \tag{25}$$

Probability, that Tf is equal 2, is (see Fig. 9):

$$\begin{aligned} Pr\{Tf = 2\} &= p_d(1 - p_d)(1 - p_a) + \\ &\quad + (1 - p_a)p_a(1 - p_d)(1 - p_a) \\ &= (1 - p_d)(1 - p_a)[p_d + (1 - p_d)p_a] \end{aligned} \tag{26}$$

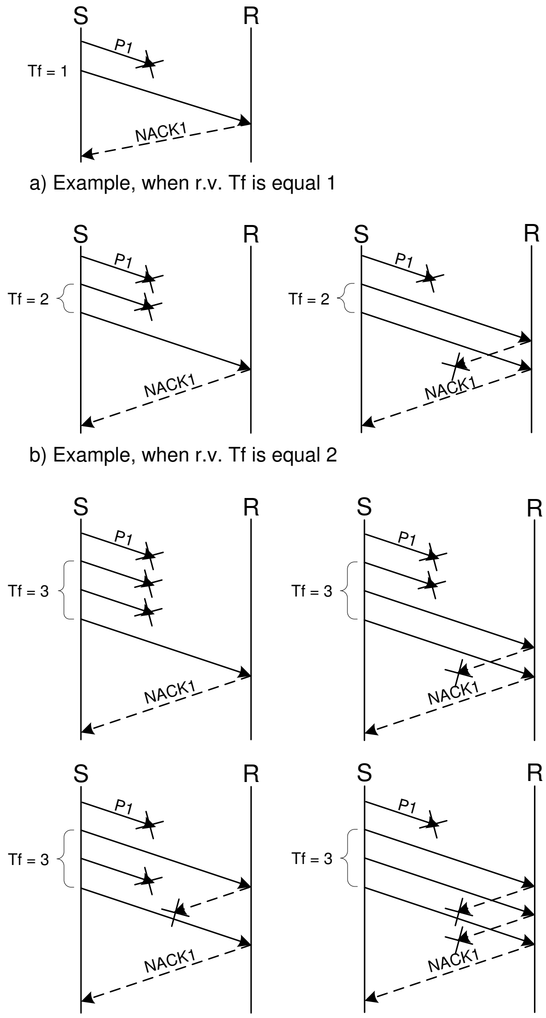
Probability, that Tf is equal 3, is (see Fig. 9):

$$\begin{aligned} Pr\{Tf = 3\} &= p_d^2(1 - p_d)(1 - p_a) + \\ &\quad + p_d(1 - p_a)p_a(1 - p_d)(1 - p_a) \\ &\quad + (1 - p_d)p_a p_d(1 - p_d)(1 - p_a) \\ &\quad + (1 - p_d)p_a(1 - p_d)p_a(1 - p_d)(1 - p_a) \\ &= (1 - p_d)(1 - p_a)[p_d + (1 - p_d)p_a]^2 \end{aligned} \tag{27}$$

Finally, random variable Tf has the geometric distribution:

$$Pr\{Tf = j\} = (1 - p_d)(1 - p_a)[p_d + (1 - p_d)p_a]^{(j-1)} \tag{28}$$

where p_d and $p_a$denotes packet loss probability for direction sender-to-receiver and receiver-to-sender, respectively.



a) Example, when r.v. Tf is equal 1

b) Example, when r.v. Tf is equal 2

c) Example, when r.v. Tf is equal 3

Fig. 9. A Tf random variable.

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