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SMPDV – A NEW JITTER ESTIMATOR PROPOSAL

Summary. Multimedia streaming transmission might be affected by jitter – short term packet delivery delay variation. To avoid transmission errors there are de-jitter buffers used, therefore it is necessary to estimate the current jitter although it is impossible to calculate straight value on-line. The situation is complicated by various jitter causes and its appearances. In the paper authors propose their own approach to measuring jitter and a simple estimator that is compared to already existing ones.

Keywords: jitter, estimation, multimedia transmission, real time transmission

SMPDV – PROPOZYCJA ESTYMATORA JITTERA

Streszczenie. Strumieniowanie multimediów w czasie rzeczywistym może zostać zakłócone przez jitter – krótkotrwałą zmienność opóźnień czasu transmisji. Aby zapobiegać zakłóceniom transmisji, stosowane są odpowiednie bufory. Niewykonalne jest wyliczanie na bieżąco jittera z jego definicji – stąd konieczność oszacowania chwilowej wartości. Sytuację komplikuje fakt, iż jitter może mieć różne przyczyny i postaci. Autorzy proponują własne podejście do pomiaru zmienności opóźnienia i prostą miarę zmienności opóźnienia, która jest porównywana z innymi metodami.

Słowa kluczowe: jitter, estymacja, transmisja multimedialna, transmisja w czasie rzeczywistym

1. Introduction

Streaming paradigm that applies well for multimedia transmission and communication is a smart idea that consists in consuming transmitted data as they arrive. It appears to be sensitive to transmission delay variation called jitter [11]. Failure occurs when a packet is delayed more than some expected and acceptable interval. Such the packet is assumed to be lost and therefore transmission quality is degraded – this is real time demand for the data transmission. In fact such packets might be simply delayed and a little delay of the presentation would allow gathering all the delayed data enough to obtain fine media quality. Usually to avoid ‘hiccups’ in transmission, jitter buffers large enough to compensate delay variation are used (see fig. 1). On the other side the buffers should be rather small, otherwise users would lose feeling of interactivity in case of two-sided transmission like VoIP. Size of the jitter buffer might be fixed or, in more advanced solutions there are used adaptive jitter buffers (AJB) – in such cases there might be a need to estimate current jitter value.

Nowadays there are several running estimators of the jitter value. As all of them have some disadvantages, so there is still room for research and improvements. Before further analysis one should ask a few very general and basic questions related to jitter.

- What do we need to measure?
- Is the delay variation a jitter or a longer term trend in transmission delay?
- Is supported information useful?

In our opinion there is following answer the above questions: estimator should detect jitter – that means variance of packet transmission delay, but also it should be an early detector for a short term delay variation that can appear in case of network congestion. Meanwhile it should be long term delay change insensitive. The goal of such a tool is to collect information about jitter (at best upper bound) – predicting future jitter values what can be useful for adaptive resizing of de-jitter buffers.

There are a few classical solutions [2,4,5,6] of the jitter estimating problem that is shown above, although our preliminary tests demonstrated that there is still room for improvements both in the details of classical estimator models and in general concept of the estimator.

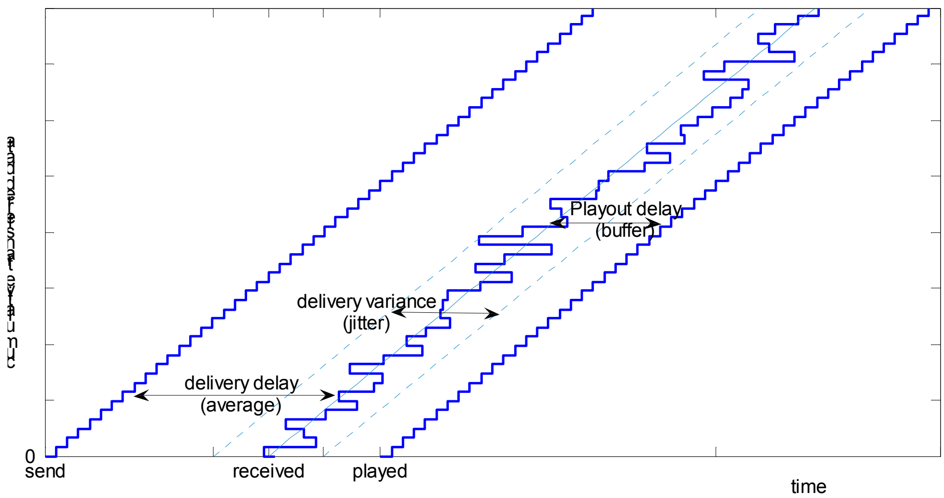

Fig. 1. Overview of transmission, delivery and consumption of constant bitrate stream
Rys. 1. Transmisja, dostarczanie i konsumpcja strumienia o stałej przepływności bitowej

Clark [1] identifies three kinds of jitter (fig. 2) that might be caused by various reasons:

a) constant jitter flawless transmission with roughly constant packet delay variation;

b) transient jitter – a single packet can be significantly delayed to the other packets in a stream. It is observed in numerous cases and it has various reasons like routing table updates, LAN congestion, router packet scheduling, route flapping and others;

c) short term delay variation – occurring when a burst of packets has increased transmission delay. It is usually connected with access link congestion or route change.

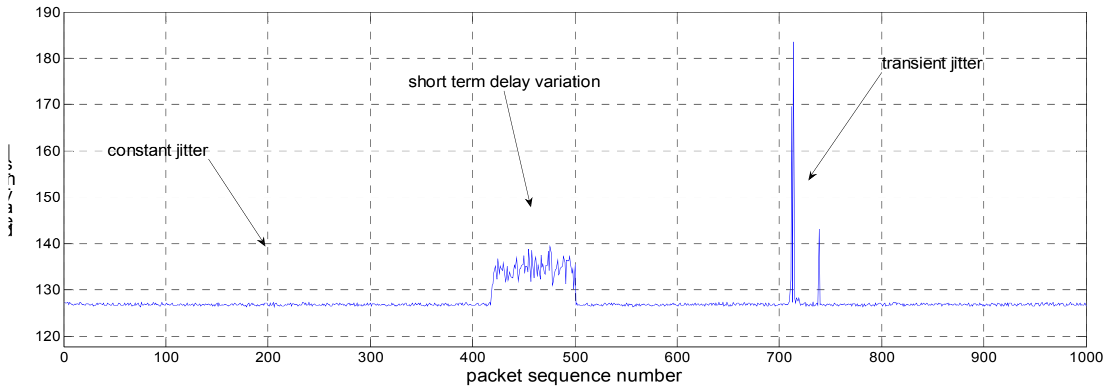

Fig. 2. Examples of jitter types
Rys. 2. Przykłady rodzajów jittera

2. Review of jitter estimators

There most of estimators are defined in documents of two interested institutions: Internet Engineering Task Force and International Telecommunication Union. They can be divided into two main groups: PDV (Packet Delay Variation) related to some absolute reference and IPDV (Inter Packet Delay Variation) with a preceding packet used as reference.

2.1. Inter packet delay variation

The basic form of this measure appears in two variants: simple IP Delay Variation, IPDV [2], given with (1) and Mean Packet to Packet Delay Variation, MPPDV [1], given with (2):

$$IPDV(n) = t(n) - t(n-1) \; , \tag{1}$$

$$MPPDV = \text{mean}(|t(n) - t(n-1)|) \; , \tag{2}$$

where: t_n, t_{n-1} is transmission delay of n^{th} and n^{th-1} packet. For MPPDV mean value is computed for the most recent 16 packets.

In the family of IETF multimedia protocols transmission of audiovisual data is performed using a pair of protocols RTP/RTCP [6]. For RTCP protocol there is a receiver report packet (RR) that supports information about current inter packet jitter according to formula:

$$J(n) = 1/16 \; |D(n)| + 15/16 \; J(n-1) \; , \tag{3}$$

where: J_n is current jitter estimate, J_{n-1} is previous jitter estimate, D_n is current packet delay change computed as a difference of transmission time of two packets using formula:

$$D(n) = t(n) - t(n-1) = [R(n) - S(n)] - [R(n-1) - S(n-1)] = \\ = [R(n) - R(n-1)] - [S(n) - S(n-1)] \quad (4)$$

where: $t(n)$, $t(n–1)$ is transmission delay, $R(n)$, $R(n–1)$ is arrival time and $S(n)$, $S(n–1)$ is timestamp (time of sending) of n^{th} and n^{th-1} packet. Such an approach is a first order autoregressive-like (AR-like) estimator that is claimed by Schulzrinne et al. [6] to enable well noise reduction and to be convergent to real values of jitter.

2.2. Packet Delay Variation

There are also two widely known variants of measure that refer to estimated local mean both authored by Clark [1,4]: Simple Mean Absolute Packet Delay Variation, MAPDV (5) and its more sophisticated version MAPDV2 (7):

$$MAPDV = \text{mean}(|t(n) - a(n)|) \ , \quad (5)$$

where: $t(n)$ is current packet transmission delay and $a(n)$ is nominal (average) transmission time. One can easily notice that in this approach a prior knowledge about default transmission time is required so it determines using local mean or median estimate.

MAPDV2 is more complicated and hard to describe using a single formula. First, one needs to estimate mean delay value a_n using Jacobson's estimator [5] with gain set to 1/16:

$$a(n) = 1/16\, t(n-1) + 15/16\, a(n-1) \ , \quad (6)$$

where: $t(n\text{-}1)$ transmission delay of recent packet; $a(n)$, $a(n\text{-}1)$ new and former estimate of mean transmission delay. Such approximate value is used in the following computations given below as meta code for the last 16 packets:

```
for (i=n-16; i<n; i++)         // n - current packet number
   if t(i)>a(i)                // i - former packet numbers
         P(i)=P(i)+(t(i)-a(i))   // positive deviation
   elseif t(i)<a(i)
         N(i)=N(i)+(a(i)-t(i))   // negative deviation
   end if                      // if t(i)==a(i) do nothing !!!
end for
```

Finally MAPDV2 is calculated as sum of mean values of positive and negative delays:

$$\text{MAPDV2} = \text{mean}(P(i)) + \text{mean}(N(i)) \, . \quad (7)$$

3. Proposed estimator concept

The estimator of jitter value proposed by the authors [9] combines both parts: measuring inter packet delay and absolute delay. We wanted it to have advantages of both of these types: sensitivity to separable peaks of inter packet estimator and ability to detect ramp-alike delay variations of absolute delay estimators so we propose a name SMPDV as an acronym of **S**witched **M**easure **P**acket **D**elay **V**ariation. The idea is similar to classical estimators (3, 6), however, the key fact is that estimating of jitter is based on a larger value chosen from two packet delay variation values as given below:

$$D(n) = \max\left\{ D_{pp}(n), D_{ap}(n) \right\}, \tag{8}$$

where: $D(n)$ is current delay value, $D_{pp}(n)$ is inter packet delay, $D_{ap}(n)$ is absolute packet delay referring to estimate of base transmission time. They are described with the formulas:

$$D_{pp}(n) = \left| t(n) - t(n-1) \right| , \tag{9}$$

$$D_{ap}(n) = \left| t(n) - a(n) \right| , \tag{10}$$

where: $t(n)$, $t(n–1)$ are delays of current and previous packets, $a(n)$ is base transmission time.

Our preliminary tests demonstrated that ARMA-like models work better for single peaks when they return faster to the constant jitter so we decided to estimate packet delay variance using ARMA-like model. Finally, our estimator can be described using the formula below:

$$J(n) = \underbrace{A\left|D(n)\right|}_{current\ delay} + \underbrace{B\left|D(n-1)\right|}_{previous\ delay\ (MA\ part)} + \underbrace{R\,J(n-1)}_{previous\ estimate\ (AR\ part)} , \tag{11}$$

where $J(n)$, $J(n\text{-}1)$ are current and preceding SMPDV estimators, $D(n)$, $D(n\text{-}1)$ are current and previous packet delay values and A,B,R are model coefficients.

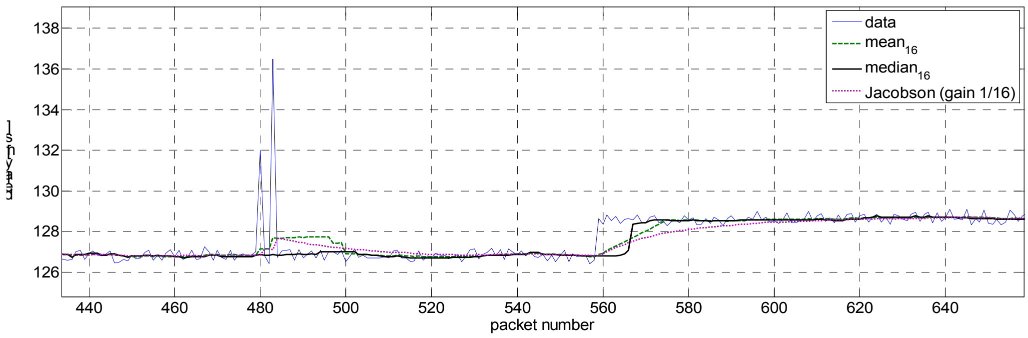

Fig. 3. Example behavior of analyzed base delay estimators for semi-synthetic data
Rys. 3. Zachowanie estymatorów opóźnienia bazowego na przykładzie danych semisyntetycznych

4. Engineering the estimator

4.1. Base delay estimate problem

We verified the behavior of several common estimators of transmission value using a set of round trip real data measures. We tested Jacobson’s estimator used in MAPDV2, moving

average and moving median ones, both using 16 packet uniform window. The interesting is shown in fig. 3 for real round trip delay data with added artificial peaks and step.
As we want the base delay estimator to be slightly 'conservative':

- there should be delayed but unhesitating response to step changes,
- it should exhibit minor sensitivity to single strong delay peaks,
- it should slightly underestimate the base delay time to obtain larger jitter values,

therefore we chose median estimator for using in $D_{ap}(n)$ PDV:

$$a(n) = \text{median}\{t(n-1), t(n-2), ..t(n-17)\} \ , \tag{12}$$

where: $t(n–1)$, $t(n–2)$,…, $t(n–17)$ are transmission delay values of 16 previous packets.

4.2. Fitting model coefficients

The final part of building the estimator was to fit coefficients of the model. It was solved in a problem optimization manner. Using a pool of real round trip packet delay measures we solved the problem numerically applying nonlinear optimization function of MATLAB optimization toolbox [6]. The objective function mean squared error between predicted delay value (jitter plus base delay) and the real transmission delay of next packet was minimized - the square error was minimized when SMPDV anticipated future delay (error of prediction to next three packets). The problem was posed as the nonlinear minimization program:

$$\begin{aligned} &\textit{objective function}: \quad MSE = \frac{1}{N}\sum_{n=1}^{N}\Big(\big(SMPDV(n)\big) - \max_{i-n..n+2}\big(|t(i) - a(i)|\big)\Big)^2 \rightarrow \min \\ &\textit{constrains}: \qquad 0 \le A, B, R \le 1, \quad A + B + R = 1 \end{aligned} \tag{13}$$

where: MSE is mean squared error, $a(i)$ is base delay estimate (12), i indices three following packets and A,B,R are coefficients of ARMA like estimator (11).

The test data consisted of four sets of round trip delay measures from our division in the university to four locations (websites) listed below from the closest to the most distant one:

- POLSL – Silesian University of Technology– very fast local connection,
- MIT – Massachusetts Institute of Technology– fast remote connection,
- UoA – University of Auckland– New Zealand – slower, remote connection,
- DWU – Divine Word University– Papua New-Guinea – slower and distant connection.

The data was collected by sending 1050 packets using true ping application [10] that fills ICMP packet with random contents and has more reliable time measurement than conventional ping. For optimization purposes we removed lost packets and used only 1000 of packets colleted after the estimator reached steady state after initial transient wavering.

One can easily notice that the results obtained with the use of the nonlinear optimization function strongly depend on the data. For the POLSL, UoA and DWU data the proposed value for the A coefficient is around 1 while for the MIT data the obtained value is small. Inverted dependence is observed for the R coefficient. For all cases the B coefficient estimated value is around 0. Authors suspect that such results are caused by the construction of the object function which prefers the estimator to be well fit to the data that results in rejecting the B value as it reason of the delayed response of the model. In authors opinion such model is over trained and it is necessary to consider the previous delay in the estimator.

Table 1

Obtained SMPDV coefficients and statistical parameters for real round trip data

	A	B	R	MSE	mean t [ms]	variance [ms]	packets lost
POLSL	1	0	0	0.1392	0.8314	0.08	0
MIT	0.2488	0	0.7512	72.1474	127.0392	24.1883	0
UoA	0.9134	0	0.0866	25.8581	331.7736	15.4495	8
DWU	0.8644	0	0.1356	62.1247	369.8570	38.6673	2

Based on the results above we decided to verify empirically our estimator for several arbitrarily chosen sets of coefficients. In tab. 2 we proposed three sets of coefficients that can be used to compute current value of the SMPDV estimator. In the *Set 1* we overestimated the influence of the previous estimator value almost not considering the current and previous packet delays. In the second case, *Set 2,* previous estimator was also the most important part of the new value, but there was also influence of the current packed delay. In the last case we almost equally considered the values of the previous estimator and current packet delay combined with a slight influence of the previous delay.

Table 2

Proposed coefficients for the SMPDV estimator and MSE for collected data sets

Coefficient values				MSE			
	A	B	R	POLSL	MIT	UoA	DWU
Set 1	1/16	0	15/16	0.147	72.7478	43.8459	74.9284
Set 2	1/4	0	3/4	0.1474	72.1476	31.332	66.0611
Set 3	1/2	1/16	7/16	0.1469	73.6968	28.4516	64.5638

The proposed sets of coefficients were tested for the four collected data sets and for the semi-synthetic data representing several different types of jitter. We also computed *MSE* value for the real round trip data. One can notice that *MSE* obtained for the *Set 3* is better in certain cases (UoA, DWU) meanwhile in other cases the prediction error remains still very close to the optimal result (see tab. 1). As mentioned in the p. 4.1 we expect estimator to immediately recognize the delay change while in case of the short term delay change to mark the change and then support a jitter value referring to new base delay. Long term changes should not appear in the estimator response. Three SMPDV estimators computed for the sets of coefficients proposed in table 2 are presented in the fig. 4 and compared against the semi-synthetic data which is example of both: transient jitter and short term delay variation. Please note that for visual convenience we shifted estimator values by adding mean transmission time. The most interesting estimator is the one obtained for the third set of coefficients. One can notice that for both types of jitter the estimator can quickly respond to the delay change. In the case of transient jitter it quickly detects the delay change and also returns very fast to its normal level (**a**). A similar situation can be observed in case of the short term delay variation. When the time of response increases, the estimated delay also increases, showing the ability to detect ramp-like delay characteristic of access link congestion (**b**), but when the transmission becomes fixed on a new level, the estimated delay variation is immediately decreased. In all three cases SMPDV do not react to slow delay change (**c**) as it is expected.

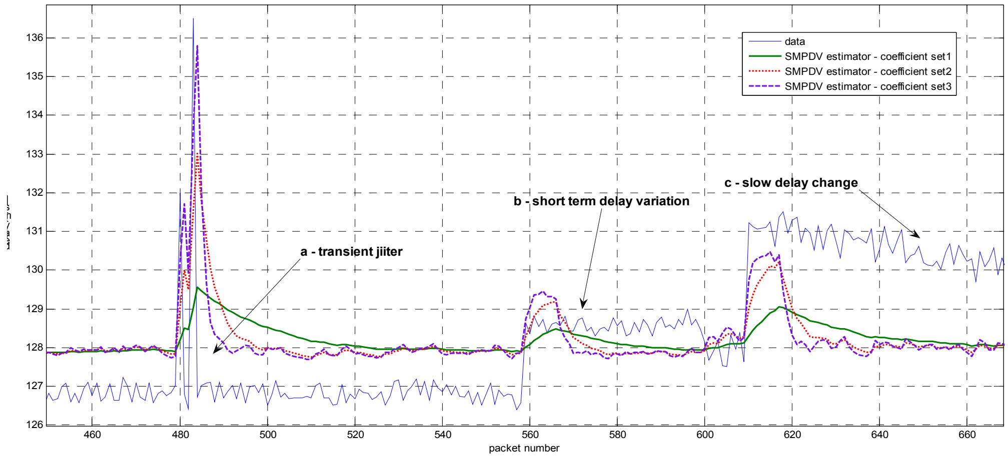

Fig. 4. Comparison of the SMPDV estimators computed for three different sets of coefficients
Rys. 4. Porównanie estymatorów SMPDV przy trzech różnych zestawach współczynników

We also compared our estimator (*Set 3*) with the MAPDV2 – results are shown in fig. 5 where estimated values are added to moving median used as estimator of base transmission delay. One can notice that proposed estimator responds faster to the delay change and also faster returns to base value after single peaks and is able to follow the data more precisely.

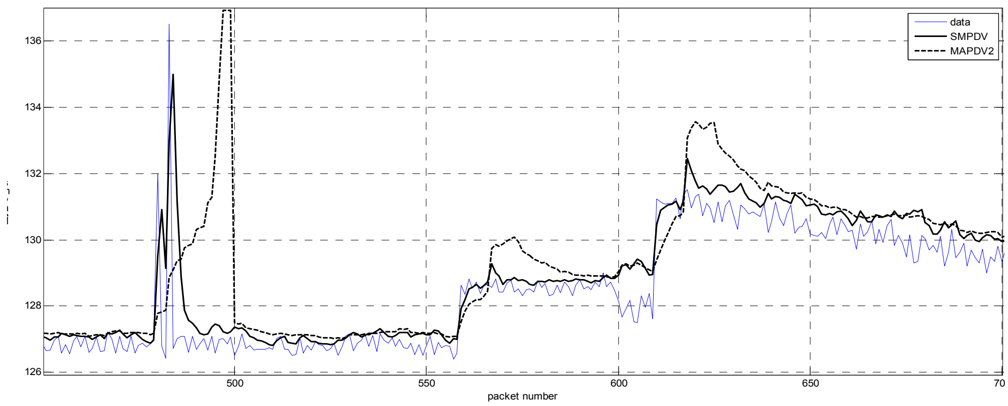

Fig. 5. Comparison of the SMPDV and MAPDV2; estimated values added to moving median
Rys. 5. Porównanie estymatorów SMPDV i MAPDV2, wartości dodane do mediany ruchomej

4.3. Application study of SMPDV in adaptive jitter buffer

Currently AJB sizing algorithms are based on a simple scenario: if there is a packet lost, size of the buffer is significantly increased and then if transmission time is smaller than some threshold value the buffer size is slowly decreased thus we obtain sawtooth shaped graph of the buffer size. One can easily imagine such adaptive jitter buffer to be based on precise estimator where jitter buffer size is set according to values returned by an estimator.

Table 3

Lost packets and average latency for various simulated adaptive jitter buffer sizes

margin / data	pure SMPDV		pure 2*SMPDV		rough SMPDV		rough 2*SMPDV		pure MAPDV2	
	lost	latency	lost	latency	lost	latency	lost	latency	lost	latency
POLSL	146	0.169	69	0.3679	0	19.950	0	19.950	80	0.286
MIT	223	0.375	53	1.048	1	19.941	1	20.261	112	1.370
UoA	245	1.526	163	4.217	2	19.035	2	19.315	127	4.085
DWU	108	4.792	54	9.208	14	21.115	11	22.7353	52	8.021

To verify applicability of our solution we simulated transmission of 1000 VoIP packets using our pool of test data according to specifications of Speex codec over RTP [3] at the basic level – sampling at 8 kHz, packet transmitted every 20 ms. Simulated buffer size was scaled according to SMPDV values in two ways: pure time value and rough time as delay with proper packet resolution (division of delay by 20 ms duration rounded towards infinity). Number of lost packets and average latency obtained as results are shown in tab. 3. They are ambiguous when compared to MAPDV2 - advantage of the estimators depends on a case but both estimators seem to be comparable so the topic seems to be worth of further study.

5. Conclusions

The proposed estimator – SMPDV has good abilities to recognize different types of jitter and can easily and quickly adapt to the delay changes that occur during the transmission. Results of tests, comparison to other solutions and applicability study are very promising. However, in our opinion there is still area for future research which can be conducted in the following directions: to test the SMPDV estimator against benchmark data, to verify the method against one-way datagram transmission like OWAMP [8] and to implement SMPDV in real client software to proof its usability for real de-jitter buffer size adaptation.

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Omówienie

W artykule przedstawiono propozycję nowej miary szacowania zmienności opóźnienia (jittera). Zdefiniowano problem zmienności opóźnienia oraz porównano i przedstawiono różne typy zmienności opóźnienia (rys. 2). Dokonano podziału istniejących już miar zmienności opóźnienia na miary oparte na opóźnieniu między pakietami (1)-(4) oraz na miary oparte na absolutnej zmienności opóźnienia (5)-(7). Zdefiniowano nową miarę zmienności opóźnienia – SMPDV, która łączy w sobie cechy obydwu tych podejść (8)-(11).

Dla danych testowych reprezentujących różne typy zmienności opóźnienia porównano (rys. 5) działanie zaproponowanej miary z miarą MAPDV2, uzyskując lepsze dopasowanie do zmienności transmisji przy użyciu miary SMPDV. Na podstawie zaproponowanej miary zasymulowano działanie adaptacyjnego bufora zmienności opóźnienia. Wyniki symulacji dla danych testowych przedstawiono w tabeli 3.

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