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On the patterns of the nonstationary datagram based fast communication processes*

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Abstract. Nowadays expectations against modern communication services involve not just Quality of Service (QoS) enhancement for real-time applications but also increased transmission rate between the storing and processing of Big Data nodes. Transmission Control Protocol (TCP) has strict flow control of the data stream providing automatic adaptation to the path load of the process-to-process communication. User Datagram Protocol (UDP) based solutions are proposed to settle the communication efficiency. In this paper, we analyse the effect of three independent communication parameters on the efficiency of looped UDP communication: the size of the Maximum Transfer Unit (MTU), the bandwidth of the end-to-end session, and the segment size of the UDP protocol data unit. The usage of nonstationary multi-resolution methods helps to identify three characteristic patterns offering identification of the objective qualitative features of the looped datagram communication services.

Keywords: datagram, high-speed network, nonstationarity, Fourier transform, nonlinearity, Wavelet transform, Empirical Mode Decomposition, Variational Mode Decomposition

AMS Subject Classification: 62J02, 62P30, 68M11, 68M12, 93B17

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1. Introduction

Intensive flow control of the TCP transport layer mechanism provides process-to-process service with low-efficiency usage of the communication paths resources like links, buffers and processors of the intermediary nodes [9]. The lack of justification for this strict connection-oriented mechanism implies technological reconsideration in practice. The strict limit of the TCP segment header of 40 bytes, the extreme complexity of multipath TCP evolution, the slow upgrade method of the new stacks, and the missing TCP header encryption make this forty years old mechanism not optimal for the next period. These issues opened development steps toward the usage of simpler and faster UDP extended with reliable services [7]. UDP is a connectionless transport layer mechanism and functions like any other datagram type service on the other positions of the communication stack: IEEE 802.3 on the datalink layer, Internet Protocol (IP) on the network layer and Simple Network Management Protocol (SNMP) on the application layer. All of these mechanisms function without prior setup or later acknowledgment phase of the corresponding protocol data unit (PDU) transmission. The absence of the PDU acknowledgment makes faster the transmission but leaves the layer service without dataflow control possibility. Communication mechanisms based on any of the datagram services require extra intelligence to provide reliability and security. The complexity and resource usage intensity of these extra solutions determines the efficiency and behavior of the applications.

The performance research works of the UDP show the necessity of a new transport layer mechanism combining the advantages of the TCP and UDP network services. The potential solution is QUIC with future development possibilities allowed by the Request for Comments (RFCs) standards created by the Internet technology responsible organization, Internet Engineering Task Force (IETF).

Methods of classical statistical time series analysis can evaluate the properties of data communication services by considering stationary features of the data. Time series which is captured from fast communication sessions shows nonstationarity of the real data transfers frequently. This property is caused by the network conditions changing rapidly and unpredictably over time. This is due to a variety of factors such as network congestion, varying traffic loads, and changes in the topology of the network. In high-speed networks the amount of traffic can vary significantly from moment to moment, leading to changes in network congestion and packet loss rates. Similarly, the network topology can change due to the addition or removal of network nodes or links, leading to changes in the routing paths and delays. Moreover, high-speed networks often use sophisticated network protocols and mechanisms such as congestion control and flow control to manage the flow of data and ensure efficient services. These mechanisms can also introduce variability and nonstationarity into the network conditions, as they can adjust the behaviour of the network dynamically based on the current traffic and congestion levels.

High-speed networks are nonlinear because their behaviour is not proportional to the input signal or traffic load. Instead, the response of the network can be highly nonlinear, meaning that small changes in the input signal can result in large and unpredictable changes in the behaviour of the network. One of the primary causes of non-linearity in high-speed networks is congestion. When a network becomes congested, packets can be dropped or delayed, leading to nonlinear changes in network behaviour. A small increase in the traffic load can lead to a disproportionate increase in packet loss rates or delays, resulting in a nonlinear response. Another factor that can contribute to nonlinearity in high-speed networks is the use of complex network protocols and mechanisms such as congestion control and flow control. These mechanisms can introduce non-linearities into the network behaviour by adjusting the flow of traffic based on the current network conditions.

Overall, the nonstationary and nonlinear nature of high-speed networks poses significant challenges for network design and management. That is why the use of advanced techniques and algorithms to ensure reliable and efficient communication over time is dispatched requirement today. Advanced modelling and simulation techniques are needed to predict the behaviour of the network accurately and optimize its performance. It also requires the development of novel algorithms and protocols that can manage the nonlinear response to changes in traffic load and congestion.

Highlights of this paper are the following:

- General, nonlinear and nonstationary properties of the packet-switched datagram mechanisms belonging to different logical layers (L2-datalink: Ethernet; L3-network: IP; L4-transport: UDP) are proved.
- Overview is given of the nonlinear and nonstationary methods used to analyse time series of the network traffic.
- Properties of the looped communication mechanism based on User Datagram Protocol are presented using captured data series from real network traffic.
- Clusterization of loop-based UDP traffics is made in function of the maximum transfer unit, the bandwidth of the end-to-end session and the segment size.

The structure of the rest of the paper is described as follows: related works on the efficiency of transport layer mechanisms are listed in section two. Section three gives an overview of the applied methodologies that have been considered and proven to be useful for us in this context of analysing nonstationary and nonlinear time series. The results and interpretation of the analysis are listed in chapter four. Finally, we conclude and give the possible continuation of the problems related to this research work.

2. Related work of the UDP performance evaluation

There have been developed several studies on the performance of the UDP in various scenarios. Testbed is applied to compare the performance of TCP and

UDP in terms of throughput, delay, and packet loss for different network conditions [16]. It was found that UDP outperforms TCP in terms of throughput for small data transfers, while TCP is more efficient for large data transfers. However, TCP experiences significant delay and packet loss under heavy network load, while UDP suffers from higher packet loss in all network conditions. It discusses the impact of Quality of Service (QoS) mechanisms on the performance of TCP and UDP, as well. It is found that QoS mechanisms can improve the performance of both TCP and UDP, but their effectiveness depends on the specific application and network conditions.

Another comparison of the performance of TCP and UDP protocols in various simulation scenarios was executed with ns-2 simulator [4]. The study evaluates the performance of TCP and UDP in terms of throughput, delay, and packet loss under different network conditions, including different traffic loads and network topologies. It was stated that the choice of the congestion control algorithm and the buffer size can significantly affect the performance of TCP and UDP and that different algorithms and buffer sizes may be more suitable for different network scenarios.

An experimental study of the throughput performance for UDP and VoIP traffic in IEEE 802.11 wireless networks is given in [10]. The study aims to investigate the impact of network parameters such as distance, packet size, and channel conditions on the throughput performance of UDP and VoIP traffic. The paper conducts a series of experiments using a testbed consisting of a set of wireless access points and clients. They measure the throughput performance of UDP and VoIP traffic under different network conditions, including varying the distance between the access point and the client, the packet size, and the channel conditions. The results show that the throughput performance of UDP and VoIP traffic is significantly affected by the network parameters. The throughput performance of UDP traffic decreases as the distance between the access point and the client increases, while the throughput performance of VoIP traffic remains relatively stable.

A research work proposes a new protocol called Performance Adaptive UDP (PA-UDP) for high-speed bulk data transfer over dedicated links [6]. The study aims to overcome the limitations of traditional UDP, which suffers from high packet loss and delay under heavy network load. PA-UDP is designed to adapt its performance to the network conditions by dynamically adjusting the packet size and transmission rate based on feedback from the receiver. The protocol uses a feedback mechanism that allows the receiver to inform the sender about the status of the network and adjust the packet size and transmission rate accordingly. The results of the work show that PA-UDP outperforms both TCP and UDP in terms of throughput and delay under heavy network load while maintaining low packet loss.

The research paper [13] proposes a new UDP-based protocol called UDT (UDP-based Data Transfer) for high-speed data transfer over wide area networks (WANs). This study aims to overcome the limitations of traditional TCP, which is not well-suited for high-speed data transfer over WANs due to its congestion control mechanism and its reliance on a reliable transport layer. UDT is designed to provide reliable and high-speed data transfer over WANs by using a congestion control algorithm that is optimized for high-speed networks, and by integrating several features such as error detection and recovery, flow control, and adaptive data rate control. The protocol also uses a feedback mechanism that allows the receiver to inform the sender about the status of the network and adjust the transmission rate accordingly.

A new protocol called Reliable Blast UDP (RBUDP) for high-speed bulk data transfer over wide area networks (WANs) is proposed [14]. The study aims to address the limitations of traditional UDP, which suffers from high packet loss and delay under heavy network load, and traditional TCP, which is not well-suited for high-speed data transfer over WANs due to its congestion control mechanism. RBUDP is designed to provide reliable and predictable high-speed bulk data transfer over WANs. It uses a combination of several techniques, including a reliability mechanism that ensures the reliable delivery of data, and an adaptive congestion control mechanism that adjusts the transmission rate based on the feedback from the receiver. The packet-blasting technique allows the sender to send multiple packets without waiting for an acknowledgment.

A paper that evaluates the security and performance of the QUIC (Quick UDP Internet Connections) protocol is [19]. The study aims to provide a comprehensive analysis of the protocol security features and performance characteristics to better understand the benefits and limitations of using QUIC. It provides an overview of the QUIC protocol and its key features, including the use of encryption, multiplexing, and congestion control. Then the security of QUIC is evaluated by analyzing its resistance to various types of attacks, including network-level attacks, cryptographic attacks, and protocol-level attacks. It compares the security of QUIC to that of other transport layer protocols such as TCP and Transport Layer Security (TLS). The performance of QUIC is evaluated in terms of throughput, latency, and fairness. They compare the performance of QUIC to that of TCP and other transport layer protocols using a testbed and several real-world scenarios. They analyze the impact of various network conditions, including network congestion and loss, on the performance of QUIC. The results show that QUIC provides better security than TCP and TLS, as it is less vulnerable to attacks such as network-level attacks and cryptographic attacks. The authors also found that QUIC performs better than TCP, especially in scenarios with high packet loss and network congestion. However, they observe that QUIC can be unfair in some scenarios, as it may prioritize traffic from certain connections over others.

Novelty in this paper is the usage of nonlinear and nonstationary empirical evaluation methods to evaluate looped data transfer traffic running on a datagram communication stack. Interpretation of the number of zero-crossings of the decomposed signal using the Fourier transform is another result of this research work.

3. Applied methodology

Since the UDP-based communication mechanisms are nonstationary and nonlinear processes we give an overview of the most important related statistical analysis methods of such time series. We will include Discrete Fourier Transform (DFT), Short Time Fourier Transform (STFT), Discrete Wavelet Transform (DWT), Empirical Mode Decomposition (EMD), and Variational Mode Decomposition (VMD) methods. In each case, we consider a time series $x(t) \in \mathbb{R}$ and $x[k], k = [0, 1, \ldots, N-1]$ denotes its observations in N equidistant discrete time points. The common approach of these methods is to decompose the inter-arrival time (IAT) series of the UDP-based traffic denoted by $x(t)$ into a sum of orthogonal or approximately orthogonal modes and residual $\mathrm{res}(t)$. The exact or approximate orthogonality of the modes depends on the method applied and in the case of exact orthogonality, the residual is null. The number of modes k also depends on the method applied and the proper value of it can determine exactly with closed formula or approximate with an algorithm. The decomposition formula is given in the following equation:

$$x(t) = \sum_{i=1}^{k} \mathrm{mode}_i(t) + \mathrm{res}(t). \tag{3.1}$$

These modes belong to frequency bands being disjunct or nearly disjunct depending on the decomposition method applied [2, 18]. The main properties of the discussed decomposition methods are given in Table 1.

Table 1. Main properties of the decomposition methods.

Property	**DFT**	**STFT**	**DWT**	**EMD**	**VMD**
Time domain aspect	No	Yes	Yes	Yes	Yes
Frequency domain aspect	Yes	Yes	Yes	Yes	Yes
Filtering aspect	Global	Linear	Dyadic	Dyadic	Linear

It should be mentioned that each method listed except DFT has both time and frequency domain aspects. Despite this fact, the importance of the DFT is indisputable in the characterization of stationary modes. The discrete Fourier transform of signal $x(t)$, is:

$$\mathbb{F}\{x[n]\} = X[k] = \sum_{n=0}^{N-1} x[n] e^{-i2\pi kn/N}, \quad k = 0, 1, \ldots, N-1,$$

see [3]. The inverse discrete Fourier transform of the sequence $X[k]$ is:

$$\mathbb{F}^{-1}\{X[k]\} = x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] e^{i2\pi kn/N}, \quad n = 0, 1, \ldots, N-1.$$

The module square $|X[k]|^2$ of $X[k]$ is called periodogram which leads to the power spectral density (spectrum) estimate of $x(t)$. Terms $W_N = e^{i2\pi/N} \in \mathbb{C}$ are the complex N^{th} roots of unity, where $i = \sqrt{-1}$. FFT is the fast algorithm for computing the DFT taking 2^r-points ($r \in \mathbb{N}$) for sequences with length $N = 2^r$. DFT has many important features that we mention here just the Convolution Theorem for two sequences $x_1[n]$ and $x_2[n]$:

$$\mathbb{F}\{x_1[n] * x_2[n]\} = \mathbb{F}\{x_1[n]\} \cdot \mathbb{F}\{x_2[n]\}.$$

The STFT is a signal processing technique that analyzes the frequency of signal content over time [12]. It does this by breaking the signal into small, overlapping segments and performing a Fourier transform on each segment. It uses a window function w[n] that is brief in duration. It has value one in the interval $[0, N_w - 1]$ and zero elsewhere. The SFTF of the signal $x(t)$ is given by the following formula:

$$X[n,k] = \sum_{m=n-(N_w-1)}^{n} w[n-m]x[n]e^{-i\frac{2\pi}{N}km}.$$

STFT examines the frequency content of a signal over time, which can be useful for analyzing changes in the frequency content of the signal.

The DWT is a type of transform that involves breaking a signal down into different frequency bands or scales using wavelet functions making it possible to analyze the signal in both time and frequency domains simultaneously [11]. DWT is based on the mother wavelet $\Psi(t) \in L^2(\mathbb{F})$, which fulfills following condition:

$$\int_0^\infty |\mathbb{F}\{\Psi(k)\}|^2 \frac{\mathrm{d}k}{k} < \infty,$$

where $\mathbb{F}\{\cdot\}$ represents the Fourier transform. Another requirement against the mother wavelet is to have vanishing moments for $M \geq 1$, i.e.:

$$\int_{-\infty}^{\infty} t^m \Psi(t)\,\mathrm{d}t = 0, \quad m = 0, 1, 2, \ldots, M.$$

Discrete wavelets are created from the mother wavelet by the following formula:

$$\psi_{j,k}(t) = \frac{1}{\sqrt{2^j}} \Psi\left(\frac{t}{2^j} - k\right),$$

where j and k are called scaling factor and displacement in time, respectively. DWT of the signal $x(t)$ is given by the following formula:

$$DWT\{x(t)\}[j,k] = \int_{-\infty}^{\infty} x(t)\psi_{j,k}(t)\,\mathrm{d}t.$$

The orthogonal wavelet basis functions are typically designed to be localized in both time and frequency, which makes them well-suited for analyzing signals at multiple resolutions. This method is useful for detecting and analyzing localized features of different sizes, shapes, spikes or transitions.

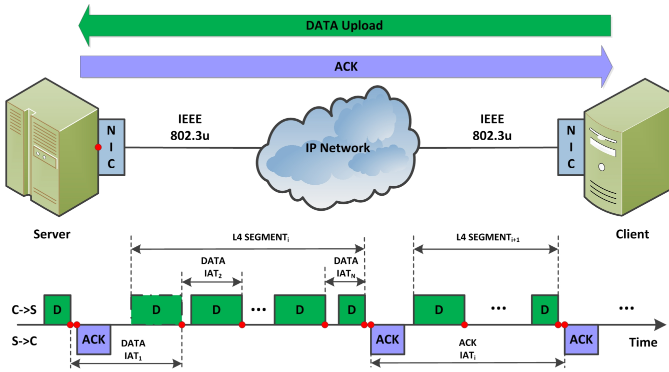


Figure 1. Measurement scenario of the looped UDP traffic.

EMD is a signal processing technique used to extract the underlying oscillatory components of a complex signal [15]. It decomposes a signal $x(t)$ into a set of intrinsic mode functions (IMFs) that represent the different oscillatory modes of the signal, from high-frequency components to low-frequency components. The EMD algorithm identifies the local maxima and minima of the signal and then fits an envelope to the signal by connecting these extrema using cubic splines. The difference between the original signal and the envelope is called a "residue" or "detail" signal, which represents the high-frequency components of the signal. The process is repeated on the residue signal, generating a new IMF and a new residue signal. This process continues until the residue signal can no longer be decomposed into further IMFs. The resulting IMFs are ordered by their frequency content, with the highest frequency components appearing in the first IMFs and the lowest frequency components appearing in the last IMFs. The Hilbert transform $\mathbb{H}\{x(t)\}$ of the signal $x(t)$ is given by the following formula:

$$\mathbb{H}\{x(t)\} = y(t) = \frac{1}{\pi} P \int_{-\infty}^{\infty} \frac{x(\tau)}{t-\tau} \,\mathrm{d}\tau,$$

where P is the Cauchy principal value. Hilbert transform is finite for each real function of class L^p. Based on $x(t)$ and $y(t)$ an analytic signal, $z(t) \in \mathbb{C}$ is created as:

$$z(t) = x(t) + iy(t) = a(t)e^{i\theta(t)},$$

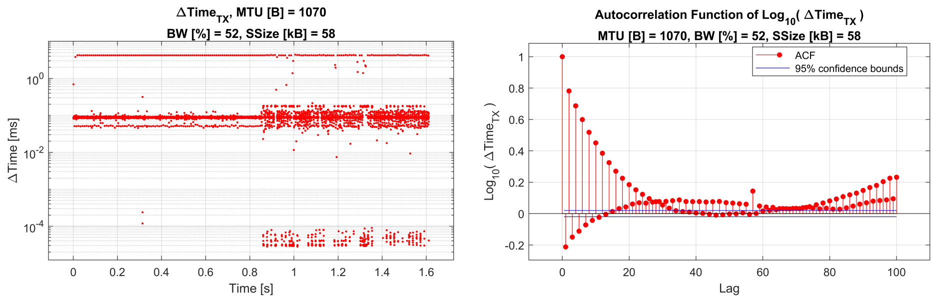
where $a(t)$ and $\theta(t)$ are the amplitude and the phase, respectively defined by:

$$a(t) = [x^2(t) + y^2(t)]^{1/2}, \quad \theta(t) = \arctan\left(\frac{y(t)}{x(t)}\right).$$

The instantaneous frequency $\omega(t)$ of the signal $x(t)$ is given by the formula:

$$\omega(t) = \frac{\mathrm{d}\theta(t)}{\mathrm{d}t}.$$

The resulting decomposition of signal $x(t)$ has the expression that conforms to equation (3.1). The number of modes k usually is less than twenty and these modes are just approximately orthogonal. The last residual is a monotone function in time. EMD is a data-driven method and does not rely on any predefined basis functions, making it suitable for analyzing nonstationary and nonlinear signals, such as those found in networking, biomedical and financial data, etc.



(a) **(b)**

Figure 2. a) Data Interarrival time series $x(t)$; b) Autocorrelation function of $x(t)$. File transfer case: (MTU, BW, SSize) = (1070 B, 52 %, 58 kB).

Variational Mode Decomposition (VMD) is another signal processing technique that, like EMD, it is used to extract the underlying oscillatory components of a complex signal [5]. The signal $x(t)$ is decomposed into a set of stationary functions by solving an optimization problem that seeks to find a set of complex-valued modes that satisfy a sparsity constraint given by:

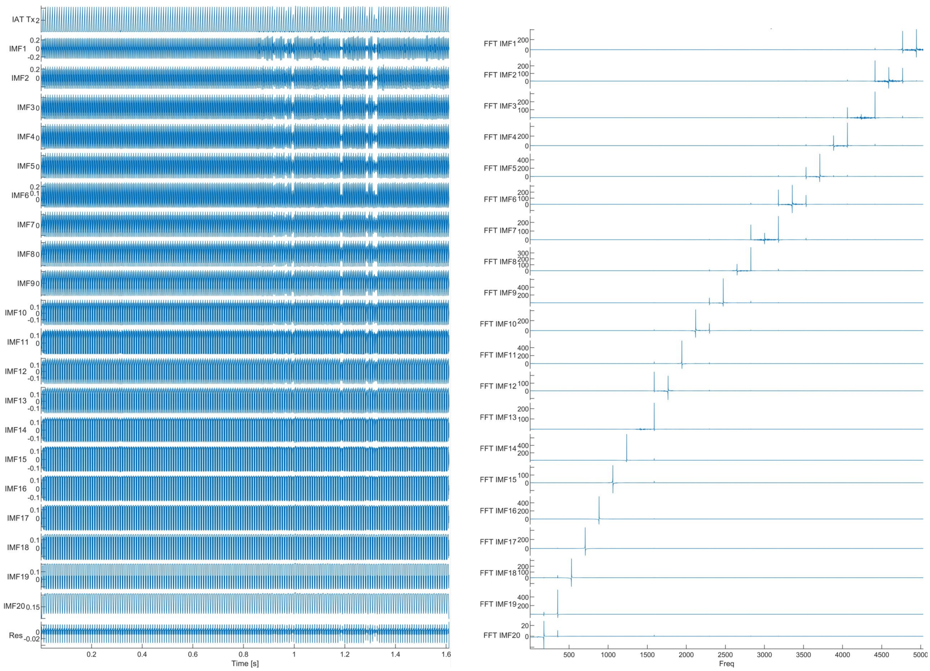
$$\min_{\{\mathrm{mode}_k\},\{\omega_k\}} \left\{ \sum_k \left\| \partial_t [(\delta(t) + \frac{i}{\pi t}) * \mathrm{mode}_k(t)] e^{-i\omega_k t} \right\|_2^2 \right\},$$

where ω_k are instantaneous frequencies, ∂_t is the partial derivative with respect to time, $\delta(t)$ is the Dirac delta function and $\|\cdot\|_2^2$ represents the square of Frobenius norm 2. The optimization problem is solved using an iterative algorithm that alternates between updating the mode functions and updating the sparsity constraint.

The resulting VMD modes are ordered by their frequency content, with the highest frequency components appearing in the first modes and the lowest frequency components appearing in the last modes. In this case, the modes conform to equation (3.1) are orthogonal. VMD has been shown to be effective in extracting the intrinsic modes of nonstationary and nonlinear signals, such as those found in audio, image, and biomedical data. VMD also has some advantages over EMD, such as better noise suppression and faster convergence.

4. Measurement scenario and basic features of the looped UDP data traffic

A UDP-based communication session with a loop was used to upload 1080 times a fixed-size data file to a test server with different combinations of the independent parameter triplets: Maximum Transfer Unit (MTU) of the interface card, Bandwidth at the application layer (Bw) and UDP segment size (SSize). The rule of the UDP loop is the following: at the reception of the data segment by the server one acknowledgment is sent back to the client. This control information sent to the client contains two main elements: i) binary status information about the successful reception or failure of the IP packets belonging to the UDP segment; ii) timestamp of the server. The status information is used for the retransmission of the wrong IP packets. The timestamp keeps track of the application-level bandwidth usage.



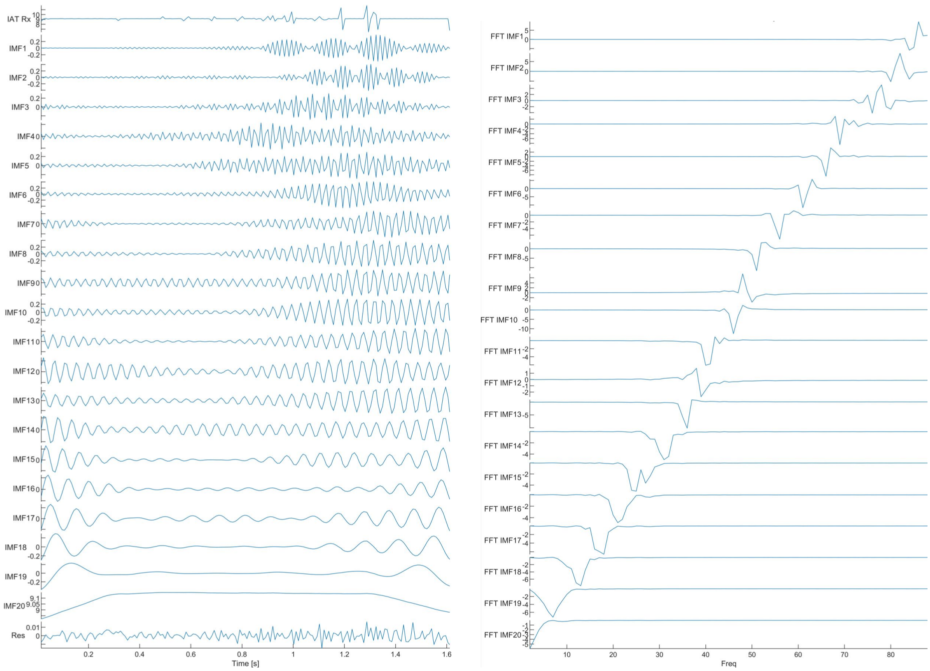
(a) **(b)**

Figure 3. a) VMD (in [ms] scale) of data IAT series; b) Spectrum (frequency in [Hz]) of VMD of data IAT series. File transfer case: (MTU, BW, SSize) = (1070 B, 52 %, 58 kB).

The size of the file was 10 MB and the values of the network parameters were: $\mathrm{MTU}[i] = 510 + i \cdot 80$ [B], $i = 1, 2, \ldots, 12$, $\mathrm{Bw}[j] = 37 + j \cdot 6$ [%], $j = 1, 2, \ldots, 9$ and $\mathrm{SSize} = -2 + k \cdot 6$ [kB], $k = 1, 2, \ldots, 10$. IAT time series of the fast communication services (see figure 2a) have nonstationary characters (see figure 2b). Decomposed modes of the IAT serve to determine characteristic time patterns in multi-resolution frequency scales.

To characterize IAT processes in time-frequency domains we used the Hilbert transform of the modes. To extract the DC component applied the following property of the Hilbert transform: the $\mathbb{H}\{x(t)\}$ is the phase shift by $\pi/2$ of the original signal $x(t)$. The remaining AC components serve to determine the spectrum of the signals and extract patterns. We processed the EMD and VMD of the data and acknowledged IAT time series of the traffics. The number of modes is limited, $k \leq 20$ and based on it mode components were determined (see VMD figures 3a and 4a).

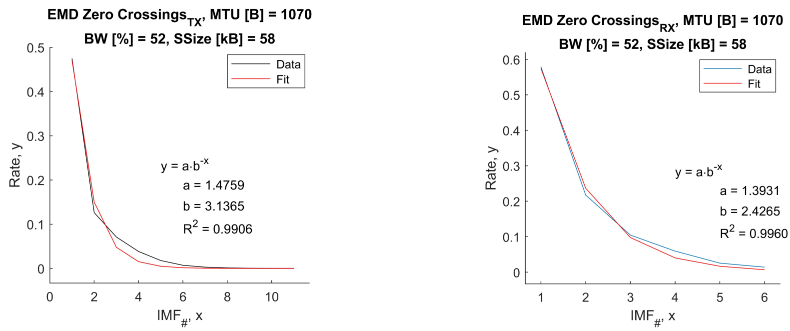
It is an important feature of the VMD that the Fourier transform of the modes have disjunct intensities in the function of the frequency and this dependence is close to being linear (see VMD figures 3b and 4b). The EMD of the data and acknowledgment traffic have different behaviour from the VMD. The EMD modes have nonlinear dependence on time and frequency. They are mentioned by the majority of scientific papers to be dyadic filters [1, 8, 17, 20].



(a) **(b)**

Figure 4. a) VMD (in [ms] scale) of acknowledgment IAT series; b) Spectrum (frequency in [Hz]) of VMD of acknowledgment IAT series. File transfer case: (MTU, BW, SSize) = (1070 B, 52 %, 58 kB).

Because the EMD and VMD modes are symmetrical, independent and stationary time functions the zero-crossing rates represent a proper measure of the frequencies. For a clear view of this dependence see figures 5a and 5b.



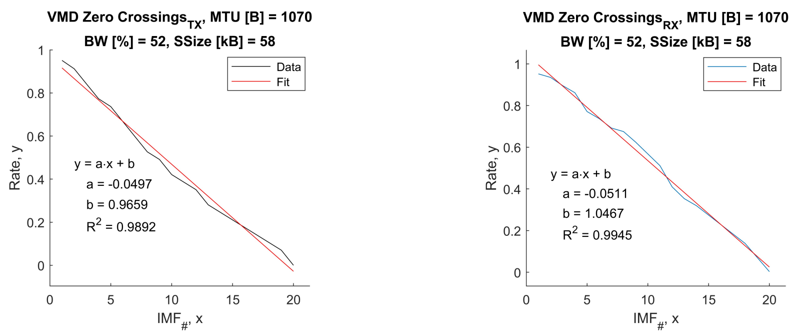
(a) **(b)**

Figure 5. a) Zero-crossings of EMD of data IAT series; b) Zero-crossings of EMD of observed IAT series. File transfer case: (MTU, BW, SSize) = (1070 B, 52 %, 58 kB).

It was found that for all cases the zero-crossing rate of EMD and VMD modes for both data and acknowledgment traffic have their own exponential and linear relations, respectively:

$$\begin{aligned} \mathrm{EMDRate}_{Tx}(k) &= a_{Tx} \cdot e^{-k \cdot b_{Tx}}, \quad & \mathrm{EMDRate}_{Rx}(k) &= a_{Rx} \cdot e^{-k \cdot b_{Rx}}, \\ \mathrm{VMDRate}_{Tx}(k) &= a_{Tx} \cdot k + b_{Tx}, \quad & \mathrm{VMDRate}_{Rx}(k) &= a_{Rx} \cdot k + b_{Rx}, \end{aligned}$$

where fit parameters a_{Tx}, b_{Tx}, a_{Rx} and b_{Rx} depend on the traffic case determined by the triplet (MTU, BW, SSize).

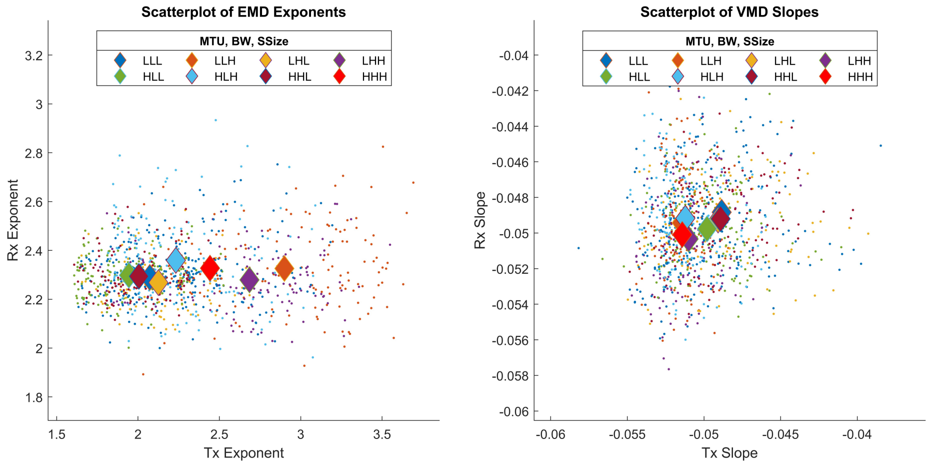


(a) **(b)**

Figure 6. a) Zero-crossings of VMD of data IAT series; b) Zero-crossings of VMD of acknowledgment IAT series. File transfer case: (MTU, BW, SSize) = (1070 B, 52 %, 58 kB).

Each range of the MTU, BW and SSize was divided into lower (L) and higher (H) half, resulting in eight 3D subspaces of the independent parameter combinations: LLL, LLH, LHL, LHH, HLL, HLH, HHL, HHH (see figures 7a and 7b). Should mention here a very important aspect: the functions of zero-crossing rates of the EMD representing the frequencies of the modes are exponential functions of base different than 2 (see figure 7a). Most bases of exponent of transmit data are in the range $(1.7, 3.5)$ and bases values of acknowledgments are in the range $(2, 2.8)$ with the mean 2.3. This fact modifies the slightly the dyadic filter property of the EMD method. The corresponding slope of the linear fitting parameters of VMD is converged around $a^* = 0.05 = 1/20 = 1/k_{\max}$ which is caused by the maximum value of VMD modes $k \leq 20$ (see figure 7b).

Based on the position of mass points of each parameter subspace three clusters as group of traffic patterns can be identified: $[C_1, C_2, C_3] = [xxL, LxH, HxH]$, where character x has the meaning of neuter effect of the corresponding parameter (i.e. $xxL = LLL \vee LHL \vee HLL \vee HHL$, where $\vee$ is the logical operator OR).



(a) **(b)**

Figure 7. a) Scatter plot of data and acknowledgment EMD exponents; b) Scatter plot of data and acknowledgment VMD slopes.

Having this clusterization property of the zero-crossing rate fitting parameters we can affirm that in the case of looped UDP services, we have three traffic patterns groups in the function of the maximum transfer unit, the bandwidth of the traffic and UDP segment size:

C_1) When the segment size, SSize is low the value of the other two parameters has no significant effect on the traffic. In this situation, the intensity of IP packet fragmentation is low at the data sender. The character of the traffic is stochastic with the bandwidth close to the limit set by the Bw parameter.

C_2) When the maximum transfer unit, MTU is low and the UDP segment size, SSize is high the bandwidth parameter, Bw has no strong effect on the traffic.

Intensive fragmentation is executed at the IP layer because large data segments are sent through short Ethernet frames. The character of the UDP traffic is dominated by fragmentation. The ratio of data frames over acknowledgment frames is the largest in these cases.

C_3) When the maximum transfer unit, MTU, and UDP segment size, SSize are high, the bandwidth parameter, Bw has no significant effect on the traffic behaviour. The intensity of the fragmentation at the IP layer is moderate, and the fluctuation of the bandwidth below the maximum is mostly reduced.

5. Summary of the results

A data file of 10 MB size was uploaded 1080 times with different traffic parameter triplets: maximum segment size, application bandwidth, and segment size of looped UDP traffic. An overview of the nonlinear and nonstationary analysis methods based on the decomposition of the interarrival times of the data upload and acknowledgment download traffics was given in the paper. It was found that the modes determined by Empirical Mode Decomposition and Variational Mode Decomposition belong to different frequency bands making it possible to characterize these stationary modes by the ratio of zero-crossing. The zero-crossing rates have an exponential and linear dependence on the number of modes of the Empirical Mode Decomposition and Variational Mode Decomposition, respectively. The Empirical Mode Decomposition is in the general b-base filter with $bget2$. The upload data traffic of the looped UDP has three groups of traffic patterns in the function of traffic parameter triplets (MTU, Bw, SSize), while the download acknowledgment traffic is a result of b-base filter with $b^* = 2.3$ instead of a dyadic filter ($b = 2.0$) published in a lot of scientific papers until now. More analyses are required to determine the dependence of the value b of the b-basis filter in the case of Empirical Mode Decomposition. Similarly, the dependence of the slope a in Variational Mode Analysis should continue to interpret.

References

[1] M. B. Abd-el-Malek, S. S. Hanna: *Using filter bank property to simplify the calculations of Empirical Mode Decomposition*, Communications in Nonlinear Science and Numerical Simulation 62 (2018), pp. 429–444, issn: 1007-5704, doi: 10.1016/j.cnsns.2018.02.035.

[2] R. Bazi, T. Benkedjouh, H. Habbouche, S. Rechak, N. Zerhouni: *A hybrid CNN-BiLSTM approach-based variational mode decomposition for tool wear monitoring*, The International Journal of Advanced Manufacturing Technology 5 (2022), pp. 3803–3817, doi: 10.1007/s00170-021-08448-7.

[3] P. Bloomfield: *Fourier analysis of time series: an introduction*, John Wiley & Sons, 2004.

[4] F. T. AL-Dhief, N. Sabri, N. A. Latiff, M. Abbas, A. Albader, M. A. Mohammed, R. N. AL-Haddad, Y. D. Salman, M. Khanapi, et al.: *Performance comparison between TCP and UDP protocols in different simulation scenarios*, International Journal of Engineering & Technology 7.4.36 (2018), pp. 172–176.

[5] K. Dragomiretskiy, D. Zosso: *Variational Mode Decomposition*, IEEE Transactions on Signal Processing 62.3 (2014), pp. 531–544, doi: 10.1109/TSP.2013.2288675.

[6] B. Eckart, X. He, Q. Wu: *Performance adaptive UDP for high-speed bulk data transfer over dedicated links*, in: 2008 IEEE International Symposium on Parallel and Distributed Processing, IEEE, 2008, pp. 1–10.

[7] *Fast transport layer protocol: QUIC*, Official web site of IETF QUIC Working Group, url: https://quicwg.org/.

[8] P. Flandrin, G. Rilling, P. Goncalves: *Empirical mode decomposition as a filter bank*, IEEE Signal Processing Letters 11.2 (2004), pp. 112–114, doi: 10.1109/LSP.2003.821662.

[9] Z. Gál, G. Kocsis, T. Tajti, R. Tornai: *Performance evaluation of massively parallel and high speed connectionless vs. connection oriented communication sessions*, Advances in Engineering Software 157-158 (2021), p. 103010, issn: 0965-9978, doi: 10.1016/j.advengsoft.2021.103010.

[10] S. Garg, M. Kappes: *An experimental study of throughput for UDP and VoIP traffic in IEEE 802.11 b networks*, in: 2003 IEEE Wireless Communications and Networking, 2003. WCNC 2003. Vol. 3, IEEE, 2003, pp. 1748–1753.

[11] J. Gilles: *Empirical Wavelet Transform*, IEEE Transactions on Signal Processing 61.16 (2013), pp. 3999–4010, doi: 10.1109/TSP.2013.2265222.

[12] D. Griffin, J. Lim: *Signal estimation from modified short-time Fourier transform*, IEEE Transactions on Acoustics, Speech, and Signal Processing 32.2 (1984), pp. 236–243, doi: 10.1109/TASSP.1984.1164317.

[13] Y. Gu, R. L. Grossman: *UDT: UDP-based data transfer for high-speed wide area networks*, Computer Networks 51.7 (2007), pp. 1777–1799.

[14] E. He, J. Leigh, O. Yu, T. A. DeFanti: *Reliable blast UDP: Predictable high performance bulk data transfer*, in: Proceedings. IEEE International Conference on Cluster Computing, IEEE, 2002, pp. 317–324.

[15] N. E. Huang, Z. Shen, S. R. Long, M. C. Wu, H. H. Shih, Q. Zheng, N.-C. Yen, C. C. Tung, H. H. Liu: *The empirical mode decomposition and the Hilbert spectrum for nonlinear and non-stationary time series analysis*, Proceedings of the Royal Society of London A: mathematical, physical and engineering sciences 454.1971 (1998), pp. 903–995.

[16] H. K. Rath, A. Karandikar: *Performance analysis of TCP and UDP-based applications in a IEEE 802.16 deployed network*, in: 2011 The 14th International Symposium on Wireless Personal Multimedia Communications (WPMC), 2011, pp. 1–5.

[17] G. Rilling, P. Flandrin: *One or Two Frequencies? The Empirical Mode Decomposition Answers*, IEEE Transactions on Signal Processing 56.1 (2008), pp. 85–95, doi: 10.1109/TSP.2007.906771.

[18] R. C. Sharpley, V. Vatchev: *Analysis of the Intrinsic Mode Functions*, Constructive Approximation 24 (2006), pp. 17–47, doi: 10.1007/s00365-005-0603-z.

[19] M. Soni, B. S. Rajput: *Security and performance evaluations of QUIC protocol*, in: Data Science and Intelligent Applications: Proceedings of ICDSIA 2020, Springer, 2021, pp. 457–462.

[20] Y. Yang, J. Deng, D. Kang: *An improved empirical mode decomposition by using dyadic masking signals*, Signal, Image and Video Processing 9.6 (2015), pp. 1259–1263, issn: 1863-1711, doi: 10.1007/s11760-013-0566-7.