Paper

Analysis of Burst Ratio in Concatenated Channels

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Abstract—Burst ratio is a parameter that quantifies packet loss patterns in transmission networks. It has been defined for an end-to-end scenario, therefore burst ratio can be determined only if the characteristics of the whole transmission path are known. In this paper, the burst ratio parameter applicability to cases when the transmission path consists of a series of transmission channels with known packet loss rate and burst ratio values is extended. The paper also presents the results of simulations performed with NS2 software, demonstrating the validity of the burst ratio analysis. Consequently, the research makes it possible to determine the value of the burst ratio parameter in concatenated packet networks, which in turn supports delivering higher quality VoIP services.

Keywords—bursty packet loss, E-model, quality of experience, voice over IP.

1. Introduction

Voice over Internet Protocol (VoIP) applications play a crucial role in connecting people and businesses around the world. It is a huge business for hardware manufacturers, network operators and service providers. In order to assure end customer satisfaction, the transmission networks must be designed well, and the quality of the provided VoIP service must be constantly monitored and maintained. In order to achieve this, all factors that affect the application quality of experience (QoE) [1] must be recognized.

The quality of VoIP carried over packet networks is influenced by multiple factors [2]. They include user-dependent aspects (e.g. user expectations), terminal quality (e.g. microphone sensitivity) and application settings (e.g. audio codec). The quality is also affected by transmission network-dependent factors, which include throughput, round-trip time and packet loss. To some extent, they can be controlled by network design and maintenance.

One of the transmission network-dependent factors that influences the perceived quality of VoIP transmissions is the burst ratio parameter [3]. It quantifies the packet loss pattern by describing the extent to which the packets were lost in bursts. The burstiness of packet loss affects the perceived media quality. If the number of audio packets lost sequentially is low enough not to be noticed by the human cognitive system, or it can be concealed by the packet loss concealment (PLC) technique [4], then the event has no impact on the perceived quality. In contrast, long sequences of lost packets can be easily perceived as an annoying quality deterioration. Therefore, the burstiness (burst ratio) of packet loss can be correlated with the perceived quality of VoIP service [5].

In order to provide a VoIP service of the best possible quality, the burst ratio parameter needs to be well recognized and analyzed. Thus far, it has only been defined for end-to-end transmission scenarios. In this case, in order to calculate the burst ratio of a transmission, the characteristics of the complete, end-to-end transmission path must be measured. This article describes the research into defining the end-to-end value of the burst ratio parameter, when the transmission is carried over multiple concatenated transmission channels and only the characteristics of each individual intermediate channels are determined.

Although extensive research on the influence of bursty packet loss on the QoE of VoIP has been carried out [6], [7], the authors are the first to analyze burst ratio in concatenated channels. In work [8], the results of theoretical studies are presented in which the formula for burst ratio in the concatenated scenario is derived. This article presents results of NS2 simulations [9] performed in order to validate the equations in a real environment. The results demonstrate the validity of the aforementioned theoretical considerations.

The results help control the burst ratio parameter by describing the impact of individual transmission channels on the burst ratio of the complete transmission path. The results will improve the quality and reliability of VoIP applications, thus improving end user satisfaction.

The remainder of this paper is structured as follows. In Section 2 the burst ratio parameter is presented and described in detail. In Section 3 we describe the methodology and features of the simulations that were carried out to validate the theoretical studies. Section 4 presents the results of the validation of the equation for Burst Ratio in concatenated channels. In Section 5 the verification of the simplified form of the equation is presented. Potential applications of the results are presented in Section 6. Finally, the conclusions are given in Section 7.

2. Burst Ratio Overview

This section presents the definition and application of burst ratio. It also contains results of our previous studies in the field of extending the burst ratio parameter applicability to multi-channel scenarios.

In order to describe packet loss of a communication channel, the packet loss rate Ppl is used. It indicates the probability of losing a packet during transmission over the channel. However, it is not a complete channel description as it does not capture packet loss patterns. Under the same packet loss rate, the loss can be evenly distributed over the whole transmission, or take place in bursts if multiple consecutive packets are lost.

The parameter that describes the packet loss pattern is burst ratio (denoted as $BurstR$). It is defined in [3] as the average length of observed bursts in a packet arrival sequence (average burst length) normalized over the length of burst expected for purely random loss (μ):

$$BurstR = \frac{\text{Average measured burst length}}{\mu}. \tag{1}$$

Burst ratio describes the packet loss pattern by expressing how much longer or shorter the measured bursts were than in the hypothetical case when all the packets were lost randomly under the same packet loss rate. Therefore, the burst ratio quantifies the observed packet loss as:

- bursty if $BurstR > 1$,
- random if $BurstR = 1$,
- scattered if $BurstR < 1$.

The length of packet loss burst expected for purely random loss (μ) is given as [10]:

$$\mu = \frac{1}{1 - Ppl}, \tag{2}$$

where Ppl stands for the probability of packet loss. The formula shows that even for purely random loss the observed burst length increases with higher packet loss, in the multiplicative inverse way. This is why the $BurstR$ value can differ dramatically for the same observed packet loss burst length, depending on the packet loss rate μ.

Generally speaking, for the same packet loss rate, higher values of burst ratio indicate that the packets are being lost in series. Conversely, lower values of the parameter mean that the packet loss was distributed more evenly over the transmission.

It is common to model packet loss in digital transmission channels with time-discrete state models, Markov chains [11], [12]. The approaches include two-state Markov chain, Gilbert or Gilbert-Elliot models. When examining the lossy transmission, authors are focusing on two-state Markov chain due to its simplicity and flexibility. In two-state Markov chain the successful transmission of a packet over a channel and losing a packet are marked with two different transmission channel states (Markov chain states). An example of the chain is shown in Fig. 1. In this case, if the channel successfully transmits a packet, it is in the F (found) state. If the packet is lost, the channel is in the L (lost) state. At any given time, the channel can only be in one of these two states.

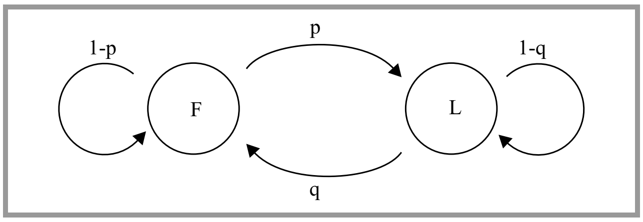

Fig. 1. In two-state Markov loss model F and L represent the found and lost states of a channel, while p and q describe the probabilities of switching the F and L states.

The two-state Markov chain is described with two parameters: p and q probabilities. The probability of losing a packet if the previous packet was successfully transmitted (transition from F to L) is described by p. Similarly, q defines the probability of successfully transmitting a packet if the previous one was lost (transition from L to F). Consequently, probability $1-p$ describes the probability of losing packets in series.

In two-state Markov chains a packet may be lost if the previous packet was successfully transmitted (with probability p) or if the previous packet was lost (with probability $1-q$). Therefore, for two-state Markov chains the probability of losing a packet is determined as:

$$Ppl = \frac{p}{p+q}. \tag{3}$$

For random loss, $q = 1-p$, the probability of losing a packet is equal to p:

$$Ppl = p. \tag{4}$$

A transmission channel modeled with the two-state Markov chain exhibits the burst ratio following the formula [13]:

$$BurstR = \frac{1}{p+q}. \tag{5}$$

Burst ratio is used in E-model [13], a commonly used analytical method of voice quality assessment. E-model uses numerous transmission parameters in order to calculate the transmission ratio factor R, which can then be used to obtain an estimated mean opinion score for the conversational scenario.

Figure 2 presents how the estimated mean opinion score value changes when the burst ratio parameter value varies between 1 and 4. The figure was created with an assumption that the G.711 codec without packet loss concealment (PLC) was used, a 1% packet loss rate was observed and other E-model parameters were used at their default values [14].

Figure 2 shows that there is a clear correlation between the application quality and the burst ratio value. Therefore, in order to calculate estimated mean opinion scores using the E-model, the burst ratio parameter must be accurately determined.

Originally, burst ratio was defined only for scenarios where the transmission is monitored and analyzed end-to-end. In [8] authors studied the burst ratio in a situation

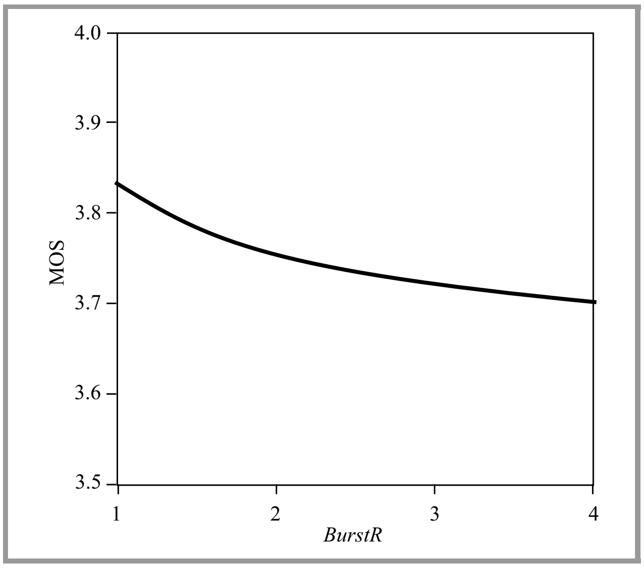

Fig. 2. Based on the E-model relationship between the estimated mean opinion score (MOS) and burst ratio parameter (*BurstR*) for 1% packet loss and the G.711 codec without PLC.

where the transmission path consists of a series of channels, and each is monitored separately. In this case, the burst ratio of the complete path must be calculated using the measured characteristics of separate channels, as presented in Fig. 3.

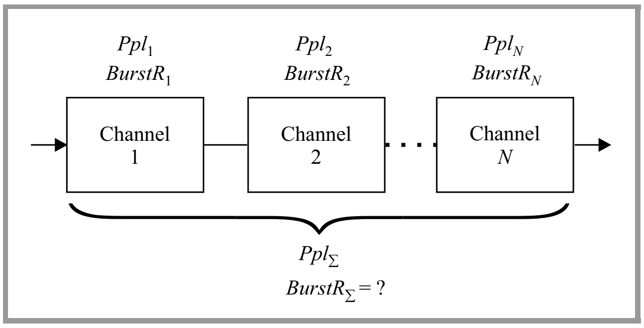

Fig. 3. The problem of burst ratio in concatenated channels network.

It was shown in [8] that if each channel can be modeled with a two-state Markov chain, the burst ratio of the complete transmission path consisting of N channels is described by the formula:

$$BurstR_\Sigma = \frac{1-\prod_{n=1}^{N}(1-Ppl_n)}{1-\prod_{n=1}^{N}\left(1-\frac{Ppl_n}{BurstR_n}\right)}, \tag{6}$$

where Ppl_n and $BurstR_n$ are the parameters of the n-th channel.

The exact value of the burst ratio can be determined with the regular burst ratio equation. However, for channels characterized by low packet loss the following formula can be assumed:

$$\prod_{n=1}^{N} Ppl_n = 0. \tag{7}$$

In this case the packet loss of multiple concatenated channels is as follows:

$$Ppl_\Sigma = \sum_{n=1}^{N} Ppl_n. \tag{8}$$

Based on this assumption, the burst ratio value of concatenated channels can be presented with the following, simpler equation.

$$BurstR'_\Sigma = \frac{\sum_{n=1}^{N} Ppl_n}{\sum_{n=1}^{N} \frac{Ppl_n}{BurstR_n}} \tag{9}$$

Analysis performed in [8] shows that this simplification is a reliable approximation of Eq. (6). The error introduced by the simplification depends on the characteristics of each channel and increases with increasing packet loss rate and burst ratio values.

As the assumption of modeling the channels with two-state Markov chains is a simplification, the authors verified the formula in a simulated network using Network Simulator 2 (NS2). The results of this verification are shown below.

3. Simulation Environment

In this section the methodology used to verify the accuracy of Eqs. (6) and (9) is described. The verification has been performed by running extensive simulations in NS2 [9].

The fundamental part of the simulation environment was designed during a seminar in Telekom Innovation Laboratories [15], which is a recognized research and develop-

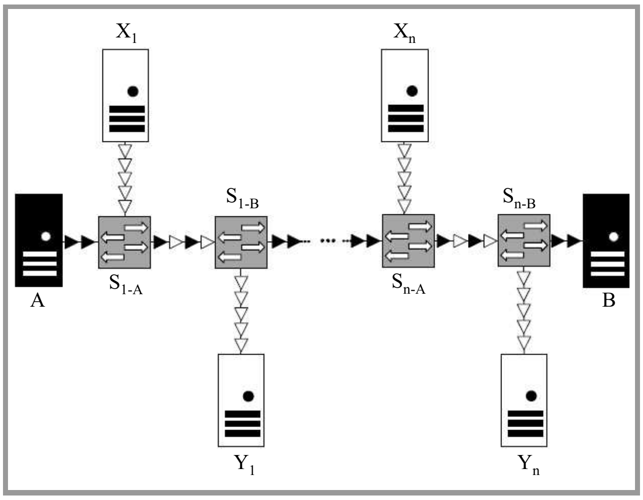

Fig. 4. The generic topology used in the simulations.

Table 1
Simulations parameters

Object	Parameter	Value	Comment
VoIP traffic	Transport protocol	UDP	
	Traffic generator	CBR	
	Packet size	50–1500 bytes	Value selected randomly (uniform distribution)
	Inter-packet interval	0.002–0.06 s	
	Start time delay	0.5–1 s	
Backgroud traffic	Number of streams transported by a single switch	1–10	Value selected randomly (uniform distribution)
	Transport protocol of a stream	TCP, UDP	
	TCP packet size	1000 bytes	
	TCP window size	2–20	
	TCP congestion control algorithm	Tahoe	
	TCP application	FTP	
	UDP traffic generator	Pareto	
	UDP Pareto shape parameter	1.4	
	UDP Pareto burst time	50–5000 ms	Value selected randomly (uniform distribution)
	UDP Pareto idle time	30000–375000 ms	
	UDP Pareto sending rate in burst	400–700 kb/s	
	UDP Pareto packet size	50–1500 bytes	
	Start time delay for each stream	0.5–1 s	
Switches	Number of intermediate switches	2–10	Each simulation repeated for every value
	Queuing scheme of each switch	DropTail, RED, FQ, SFQ	Value selected randomly (uniform distribution)
	Buffer size of each switch	2–20 packets	
Links	Capacity	500–1000 kb/s	Value selected randomly (uniform distribution)
	Propagation delay	0–200 ms	
Simulation	Duration	10, 100, 1000 s	Each simulation repeated for every value

ment institute in the field of quality of audio and multimedia applications.

NS2 is a commonly used [16] simulation environment for testing and studying communication protocols and networks. It can be used to simulate TCP/IP protocol stacks, traffic sources of various distributions and packet queuing and dropping mechanisms.

The release NS2 2.35 was used in this research in order to simulate packet transmission over a series of switches and to analyze packet loss. Each switch serves a number of packet streams and drops packets in case of a buffer overflow. After each simulation the burst ratio calculated at the end of the transmission path using Eq. (1) is compared with the burst ratio value calculated from the transmission parameters of each intermediate switch using Eq. (6). The calculations are performed by analyzing the NAM trace files generated by each NS2 simulation.

The topology used in the simulations is a path presented in Fig. 4. It contains two endpoints (A and B) responsible for a VoIP transmission, n pairs of background traffic servers $(X_1, Y_1, \ldots, X_n, Y_n)$ and n pairs of switches $(S_{1-A}, S_{1-B}, \ldots, S_{n-A}, S_{n-B})$. VoIP traffic, marked with black arrows, is sent from server A to server B. n background traffic streams, marked with white arrows, are sent between servers X_1 and $Y_1, \ldots, X_n$ and Y_n. VoIP traffic and background traffic compete for resources of shared links, which are built up by pairs of switches $S_{1-A} \longleftrightarrow S_{1-B}, \ldots,$ $S_{n-A} \longleftrightarrow S_{n-B}$. Consequently, at switches $S_{1-A}, \ldots, S_{n-A}$ the VoIP packets and the background transmission compete for access to the shared links. If not enough bandwidth is available to serve both streams, the switches drop packets. Therefore, in the simulation the transmission path of the VoIP application consists of a series of links. However, packets may be dropped at shared links only. Other links do not drops packets because they always have enough bandwidth due to transmitting either VoIP or background traffic only. At the end of the simulation, the packet loss analysis of each switch which drops packets is performed.

During the analysis the VoIP application packet loss rate and the burst ratio value are calculated. Using these values and Eq. (6), the burst ratio of the whole transmission (from node A to B) is calculated. The calculated value is compared with the value calculated at node B based on the analysis of VoIP stream packets that were not successfully delivered, Eq. (1). The result of the comparison quantifies the accuracy of Eq. (6).

It should be noted that in the simulations the packet loss takes place in shared links only. Therefore, in the remaining sections the terms "channel" and "shared link" are used interchangeably.

The results of the simulations may depend on the topology as well as transmission and network parameters. The complete list of parameters identified and analyzed during the simulations is presented in Table 1. The parameters were randomly altered within a range of values during each simulation in order to reduce the influence of a specific parameter value on the results. The parameter values and ranges of values were adjusted so the results of the simulations were relevant for the study of burst ratio parameter.

In order to obtain meaningful results it was important that the VoIP traffic was constantly generating packets. Therefore VoIP traffic utilized the user datagram protocol (UDP) with a constant bit rate. Additionally, the randomization of the background traffic was of crucial importance in order to assure a full spectrum of simulation conditions. Therefore, the background traffic used UDP (with the Pareto distribution) and TCP protocols, both selected randomly for each simulation. Moreover, the start time and the total number of transmitted packets within each transmission were also randomized. As a result the VoIP traffic faced different conditions in each simulation run. The wide spectrum of conditions meant the VoIP traffic was characterized by a wide range of parameters values $BurstR$ and packet loss rate Ppl.

This paper presents the results of a total 250,000 simulations, each representing different network conditions. They were carried out in order to demonstrate the validity of the equations. As a result, the validation contains relevant and fully conclusive results.

4. Accuracy of Burst Ratio Calculation

In this section the simulation results run in order to validate Eq. (6) are presented. The equation was numerically verified by the authors in [8], where a transmission channel was modeled by a two-state Markov channel. This section contains simulations results, where the transmission environment was modeled with real networks characteristics, simulated using NS2.

The verification has been performed by comparing two burst ratio values:

- the $BurstR$ value measured at the end of the transmission path using Eq. (1),
- the value calculated using Eq. (6), which incorporates the characteristics of each intermediate transmission channel, denoted below as $BurstR_\Sigma$.

The comparison is presented as relative error δ_Σ, defined as follows:

$$\delta_\Sigma = \frac{BurstR_\Sigma - BurstR}{BurstR}. \tag{10}$$

If δ_Σ is equal to 0, Eq. (6) is perfectly accurate. A positive value of δ_Σ means that the experienced packet loss is less bursty than that estimated using Eq. (6). A negative value of δ_Σ means that the burst ratio value calculated with Eq. (6) underestimated the burstiness of the analyzed traffic.

The number of shared links may have an impact on the final results, because the VoIP traffic needs to compete for resources in each link. The more shared links, the more VoIP packets may be lost. In order to study this impact, each simulation was rerun with two, six and ten shared links.

The results published in this section present the relationship between relative error δ_Σ (in %) and packet loss Ppl, number of transmitted packets or $BurstR$ of the complete transmission. The error is analyzed in the form of a mean and its confidence intervals. The mean value of the relative error is shown using black lines. The 95% confidence intervals of the mean are marked with gray areas.

Figure 5 presents the relationship between the relative error δ_Σ of the burst ratio calculation using Eq. (6) and packet

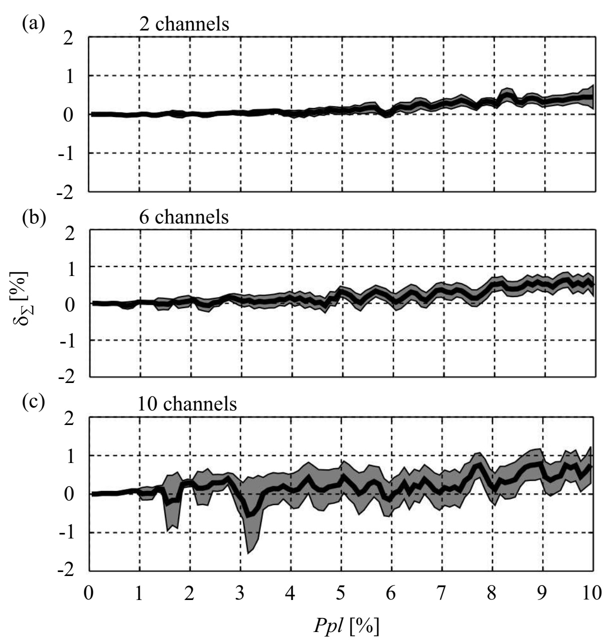

Fig. 5. Relationship between the burst ratio calculation error δ_Σ and the packet loss rate Ppl of the whole transmission. The solid line represents mean relative error while the gray areas present the 95% confidence intervals of the mean. The figures were created with a packet loss range of 0–10%. The subplots presents results for simulations of two, six and ten intermediate channels.

loss Ppl of the whole transmission. It can be observed that for values of packet loss lower than 1%, the relative error is negligible, regardless of how many intermediate channels the transmission contains. As the packet loss increases, the mean error and its confidence interval increase slightly as well. The observed increase is dependent on the number of intermediate channels. The higher the number of channels, the higher the error for the same value of packet loss. However, the relative error never reaches 2%, which indicates a high accuracy of the equation.

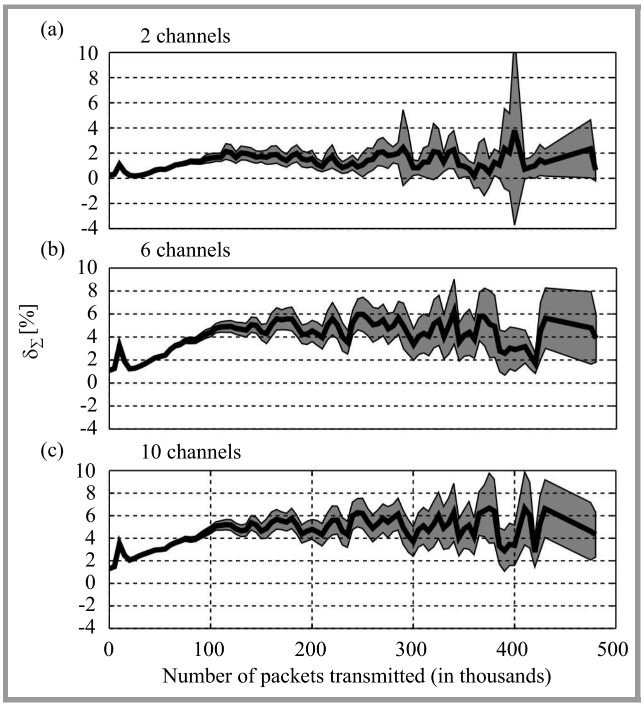

Fig. 6. Relationship between the burst ratio calculation error δ_Σ and the number of transmitted packets. The solid lines represent mean relative error while the gray areas present the 95% confidence intervals of the mean. The subplots presents results for simulations of two, six and ten intermediate channels.

Figure 6 presents the relationship between the burst ratio calculation error δ_Σ and the number of transmitted packets during measurement. The figure shows that the mean error initially slightly increases for the shorter observations and then stabilizes at a level of 2% for two intermediate channels or 5% for ten channels. Figure 6 presents results for up to $500,000$ transmitted packets, which corresponds to approximately 2 hours 45 minutes observation of a transmission. Such a long observation is unrealistic and its results are presented only for reference. More reasonable duration of observation is up to 5 minutes, which corresponds to 0–15,000 of transmitted packets. In this range the error never exceeds 4%, regardless of the number of intermediate channels.

Figure 7 presents the relationship of the relative error δ_Σ of the burst ratio calculation using Eq. (6) and burst ratio value *BurstR* of the complete transmission. It can be seen that regardless how many intermediate channels are used the relative error is low around $BurstR = 1$. For two channels, the error value is negligible, regardless of the burst ratio value. In the case of several intermediate channels, as the burst ratio increases, the error decreases and for $BurstR > 1.5$ the error becomes negative. In the worst case, for the scenario of ten intermediate channels the error reaches -9%. It can be seen that for fewer channels, *BurstR* of the complete path reaches higher values. For ten intermediate channels the highest value of *BurstR* slightly exceeds 2.5, while for two channels it is over 3.5. This effect can be explained by analyzing Eq. (9). The formula shows that *BurstR* value of the complete path is approximately equal to the weighted harmonic mean of all intermediate channels' *BurstR* values. As the result, the more channels are involved in the transmission, the lower probability that end-to-end burst ratio reaches high values.

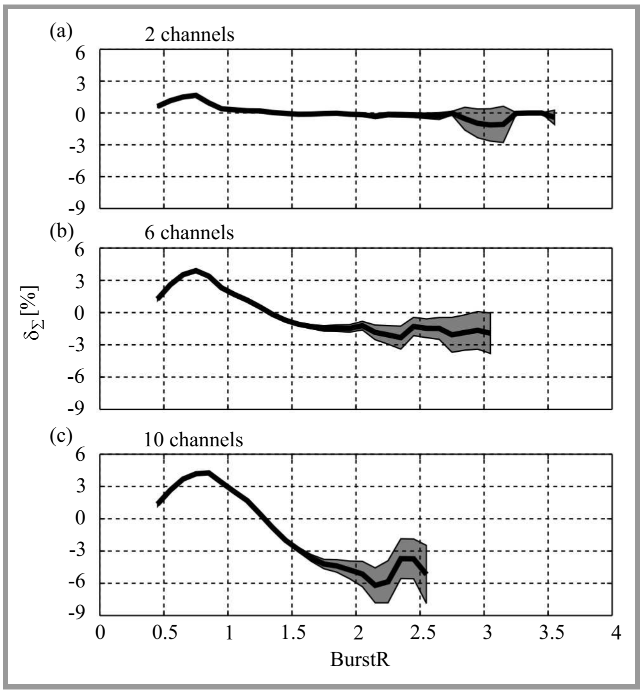

Fig. 7. Dependency of the burst ratio calculation error δ_Σ on the *BurstR* value of the complete transmission. The solid lines represent mean relative error while the gray areas present the 95% confidence intervals of the mean. The subplots presents results for simulations of two, six and ten intermediate channels.

All these results show that when Eq. (6) is used it provides reliable results and a high precision of the measurement. The accuracy of the calculation is always very high, but the most precise results are achieved in the two-channel scenario, when packet loss of the complete transmission path is limited or the burst ratio of the complete transmission path is not higher than $BurstR = 1.5$.

5. Accuracy of the Simplified Equation

As well as the regular burst ratio equation, validated above, we also show a simplified version of the equation, Eq. (9).

This simplification reveals that the burst ratio of the whole transmission path can be approximated with a weighted harmonic mean of properties of individual channels. This equation was verified numerically in [8]. The results indicate that the simplified equation's inaccuracy increases with higher values of packet loss and burst ratio of the whole transmission. However, the verification was performed with the assumption that the transmission channels can be modeled with two-state Markov chains, which is a form of simplification. This section presents the results of equation validation performed in an environment that simulates real characteristics of transmission channels.

The verification of the simplified burst ratio equation – Eq. (9) is performed by calculating the simplification error Δ_{BurstR}. It expresses the difference between the error of the simplified equation and the error of the regular burst ratio equation – Eq. (6). The values that are compared are mean relative error (in %) and the 95% confidence interval of the mean. Both were introduced in Section 4. The comparison of mean error is performed by calculating the difference between absolute values of mean error δ'_{Σ} of the simplified burst ratio equation (Eq. 9) and mean error of the regular burst ratio equation δ_{Σ}, as described in Eq. 10. The comparison is presented below:

$$\Delta_{BurstR} = \left|\delta'_{\Sigma}\right| - \left|\delta_{\Sigma}\right| . \tag{11}$$

If the calculated difference of the mean error is equal to 0, both Eq. (6) and Eq. (9) are equally accurate. When Δ_{BurstR} is positive, Eq. (6) is more accurate, while if Δ_{BurstR} is negative, the simplified equation is more accurate.

The comparison of the 95% confidence intervals of the mean is performed in a similar way, by subtracting the value of the confidence interval for the regular equation from the value of the confidence interval for the simplified equation.

The figures published in this section present the calculated differences of mean error using black lines. The gray areas in the figures correspond to the confidence interval differences of the means.

Figure 8 presents the differences of mean errors and confidence intervals in the domain of packet loss in the range of 0–10%. It can be seen that regardless how many intermediate channels are used, the difference is negligible in that it never exceeds 0.5%. However, it should be noted that there is almost no difference in the confidence interval width (marked with gray fields).

The results clearly show the validity of the simplified equation. The difference in performance, compared with the regular equation, is almost indistinguishable. However, the regular equation almost always performs slightly better than the simplified formula. Therefore, when the highest accuracy of the measurements is required, the regular equation is used. However, when the top priority is ease of calculation, the simplified equation is applied.

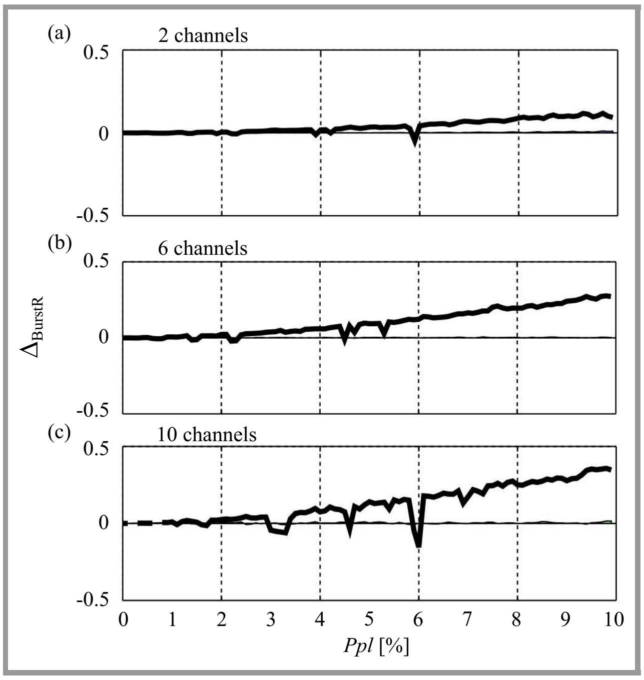

Fig. 8. Difference Δ_{BurstR} between the calculation errors of the regular equation δ_{Σ} and the simplified equation δ'_{Σ} in the domain of packet loss Ppl of the whole transmission. The solid lines represent the difference between mean relative errors while the gray areas represent the difference between the 95% confidence intervals. The subplots presents results for simulations of two, six and ten intermediate channels.

6. Applications

As mentioned above, burst ratio is one of the parameters used in the ITU-T E-model, which is used to assess the quality of VoIP. Therefore, the formula presented has a wide spectrum of potential applications, mainly facilitating the VoIP MOS level assessment.

The formulas can be used during network planning. When a network is being designed, a set of technical requirements is specified for the network. They include packet loss, round trip time and mean opinion score (MOS) of VoIP transmission. When network topology is defined, the characteristics of all the network elements are assumed. Even if the topology is complex and the network contains hundreds of elements, the VoIP transmission MOS assessment between any endpoints may be required. Without proper calculation of the burst ratio value between the endpoints, a precise assessment of application quality is not possible. Using the formulas presented and the E-model, MOS can be easily and precisely assessed between any endpoints of the designed network. Therefore, during the network design phase, corrections may be applied to the network topology to help provide the best quality of the VoIP service.

Another application of the formula is when a network is already operating and a re-design of the topology or routing is required. In this case the formula may help assess the impact of the changes on the quality of the VoIP transmission. A good example would be a network that contains multiple elements which introduce packet loss. If only one

of them could be upgraded, it would be important to select the optimal element to upgrade. By using the formula, the network administrator can easily assess how end-to-end VoIP quality would be affected, depending on which elements are upgraded.
The formulas can also be successfully used during monitoring of networks. The measurements, as described in [17], need specially configured environments. Therefore they can only be performed within a single network, owned by a single company. If a VoIP transmission path is established via several different networks, which are administered by different companies, the complete path monitoring is not possible. In this case, the formulas can be used in order to calculate the VoIP transmission MOS using monitoring logs of the individual networks.

7. Conclusions

The results clearly show that the equations presented can be successfully used to calculate the burst ratio parameter, when the complete transmission path consists of multiple concatenated channels. Although the equation has been derived theoretically using two-state Markov models, in real-life scenarios, simulated here using NS2, the equation is still valid. Its accuracy is the highest when the number of concatenated channels is limited to two, when the packet loss of the complete transmission path is low, or the burst ratio of the complete transmission path is not higher than $BurstR = 1.5$.
Moreover, the results show that the simplified version of the equation is almost as accurate as the regular equation, therefore it can be used as an engineering tool. The simplified formula reveals that the burst ratio value of the complete transmission path can be regarded as a harmonic mean of the individual channels burst ratio values, weighted with the their packet loss probabilities.
The results also demonstrate that the equation is valid and therefore can be used in QoE measurements and network performance assessment. Moreover, the formula has a wide spectrum of potential application. As such, it would be useful in improving the quality of VoIP applications.

Acknowledgments

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References

[1] “Vocabulary for Performance and Quality of Service”, ITU-T Rec. P.10/G.100 (incl. Amendment 2), Tech. Rep., 2008.

[2] A. Raake, *Speech Quality of VoIP: Assessment and Prediction*. Wiley, 2006.

[3] J. W. McGowan, “Burst ratio: a measure of bursty loss on packet-based networks”, 16 2005, US Patent 6,931,017.

[4] C. Perkins, O. Hodson, and V. Hardman, “A survey of packet loss recovery techniques for streaming audio”, *IEEE Network*, vol. 12, no. 5, pp. 40–48, 1998.

[5] A. Raake, “Short-and long-term packet loss behavior: towards speech quality prediction for arbitrary loss distributions”, *IEEE Trans. on Audio, Speech, and Lang. Process.*, vol. 14, no. 6, pp. 1957–1968, 2006.

[6] A. D. Clark, P. D. F. Iee, and M. Ieee, “Modeling the effects of burst packet loss and recency on subjective voice quality”, in *Proc. IP Telephony Worksh.*, New York, 2001.

[7] W. Jiang and H. Schulzrinne, “Perceived quality of packet audio under bursty losses”, in *Proc. 21st Ann. Joint Conf. IEEE Comp. Commun. Soc. InfoCom 2002*, New York, USA, 2002.

[8] J. Rachwalski and Z. Papir, “Burst ratio in concatenated markovbased channels”, *J. Telecommun. Inform. Technol.*, no. 1, pp. 84–90, 2014.

[9] “The Network Simulator NS-2” [Online]. Available: http://www.isi.edu/nsnam/ns/

[10] S. M. Ross, *Introduction to Probability Models*, 11th ed. Academic Press, 2014.

[11] J.-C. Bolot, “Characterizing end-to-end packet delay and loss in the interne”, *J. of High Speed Netw.*, vol. 2, no. 3, pp. 305–323, 1993.

[12] H. A. Sanneck and G. Carle, “Framework model for packet loss metrics based on loss runlengths”, *Proc. of SPIE*, vol. 3969, pp. 177–187, 1999.

[13] “The E-Model, a computational model for use in transmission planning”, “ITU-T Rec. G.107, Tech. Rep., 2014.

[14] “Transmission impairments due to speech processing”, ITU-T Rec. G.113, Tech. Rep., 2007.

[15] J. Rachwalski, “Bursty loss modelling in E-model”, Semminar in Telekom Innovation Laboratories, Berlin, 25 Apr. 2013.

[16] S. Kurkowski, T. Camp, and M. Colagrosso, “MANET simulation studies: the incredibles”, *ACM SIGMOBILE Mob. Comput. and Commun. Rev.*, vol. 9, no. 4, pp. 50–61, 2005.

[17] “Cisco IOS IP SLAs Configuration Guide”, Cisco Systems, Inc., Tech. Rep., 2008.

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