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LOW COMPLEX FORWARD ADAPTIVE LOSS COMPRESSION ALGORITHM AND ITS APPLICATION IN SPEECH CODING

Jelena Nikolić — Zoran Perić — Dragan Antić
Aleksandra Jovanović — Dragan Denić *

This paper proposes a low complex forward adaptive loss compression algorithm that works on the frame by frame basis. Particularly, the algorithm we propose performs frame by frame analysis of the input speech signal, estimates and quantizes the gain within the frames in order to enable the quantization by the forward adaptive piecewise linear optimal compandor. In comparison to the solution designed according to the G.711 standard, our algorithm provides not only higher level of the average signal to quantization noise ratio, but also performs a reduction of the PCM bit rate for about 1 bits/sample. Moreover, the algorithm we propose completely satisfies the G.712 standard, since it provides overreaching the curve defined by the G.712 standard in the whole of variance range. Accordingly, we can reasonably believe that our algorithm will find its practical implementation in the high quality coding of signals, represented with less than 8 bits/sample, which as well as speech signals follow Laplacian distribution and have the time varying variances.

K e y w o r d s: forward adaptive technique, loss compression algorithm, piecewise linear optimal compandor

1 INTRODUCTION

Numerous research has been conducted during the recent years with the goal to develop a coding algorithm that minimizes the bit rate in the digital representation of a speech signal without a significant loss of the signal quality in the process. Although a great number of speech coding algorithms has been developed [1–5], there is still an indication of the reasonable need for continuation of the research in this field [6]. We decided to find the solution to the formulated problem by means of waveform coders, since they provide the highest level of speech quality [1 ,6–8].

The simplest and the most commonly used waveform coding algorithm, defined by the G.711 standard [9], provides high quality speech at 64 kb/s. Particularly, it provides the conversion of 12 bits samples to 8-bit code by using companded 8 bits/sample Pulse Code Modulation (PCM) [1, 6–9]. Up to now, so much work has been done in the field of loss compression algorithms in order to provide not only an additional reduction over PCM bit rate, but also to provide the highest possible quality of the digitized speech signal (measured by $SQNR$ — Signal to Quantization Noise Ratio). The most significant result of such a research is the development of the adaptive predictive speech coders, which has brought the bit rate for high quality speech coding to 16 kb/s providing a reduction by a factor of four over the PCM bit rate [1, 4, 6, 10]. However, since it has already been shown that the complexity of the adaptive predictive speech coders can be considerably high [1, 4, 10], we were motivated to find the manner to reduce the PCM bit rate by means of low complex non-predictive waveform coding algorithm. Take a notice of the fact that an additional reduction over the PCM bit rate can be achieved by applying some of the lossless compression algorithms, as it has been shown in [11], where the Ramalho G.711 lossless compression algorithm has been applied. The lossless compression algorithms are usually applied after the loss compression of the signal has been performed. Therefore, it is important to research the suitable loss compression algorithm that provides the highest possible signal quality for the given bit rate. After the suitable choice of the loss compression algorithm has been performed, any of the lossless compression algorithms can be applied to provide a further bit rate reduction. Accordingly, in this paper we have decided to focus on the field of loss compression algorithms.

Regarding the time varying characteristics of speech signals, we directed our research towards the field of adaptive coders [1, 2, 7, 8] that attempt to make the encoder-decoder (coder or quantizer) designs adaptable to the varying characteristics of the input signals. Furthermore, since backward adaptation provides the $SQNR$ within 1 dB of forward adaptation, as well as since forward adaptation is less sensitive to transmission errors when compared to backward adaptation [8, 11], we have destined to perform our research in the field of forward adaptive waveform coding algorithms. Regarding the smallest average complexity of scalar compandor model [13], our research has started with the forward adaptive scalar compandor. Particularly, we decided to provide an additional complexity reduction by performing the linearization on the nonlinear forward adaptive scalar compandor. Accordingly, in this paper we propose the forward adaptive piecewise linear optimal companding coding algorithm that presents the information about the input signal with approximately 7 bits/sample.

We will discuss the performance ($SQNR$ and the bit rate) of the proposed algorithm, which has been determined in a wide variance range of the input speech sig-

* Faculty of Electronic Engineering Nis, Aleksandra Medvedeva 14, 18000 Nis, Serbia, jelena.nikolic@elfak.ni.ac.rs

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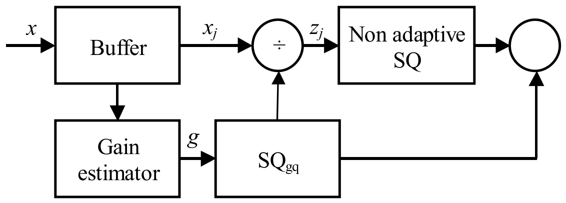

Fig. 1. Forward adaptive coding scheme with non adaptive scalar quantizer

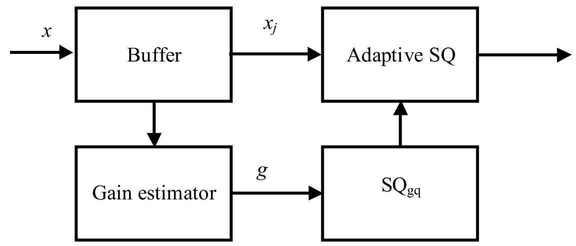

Fig. 2. Forward adaptive coding scheme with an adaptive scalar quantizer

nals. Additionally, experimental results of the real speech signal processing will be analyzed in order to practically test the performance of the proposed coding algorithm. In order to point out the benefits of the proposed algorithm, the achieved performance will be compared to the appropriate one corresponding to the speech coding algorithm defined by G.711 standard [9]. Finally, in order to point out that the proposed speech coding algorithm provides a high quality speech coding, G.712 standard [14] that defines the smallest $SQNR$, which has to be achieved for a high quality transmission, will be used.

2 NOVEL FORWARD ADAPTIVE CODING ALGORITHM

Adaptive coding schemes or algorithms work on frame-by-frame basis, where a frame consists of a certain number of samples $[1, 5, 7, 8, 12]$. The conceptual difference between the adaptive coding algorithms is based on the manner in which the adaptation is performed — whether it is performed forward, *ie*, from the input sequence or backward, *ie*, from the coded output signal $[5, 7, 8, 12]$. Due to the reduced sensitivity of the forward adaptive technique to transmission errors when compared to the backward adaptive technique, as well as since it has been demonstrated that the backward adaptive technique provides $SQNR$ within 1 dB of the forward adaptive technique $[8, 12]$, we have decided to focus our research on the forward adaptive coding algorithms. Forward adaptation can be performed by normalizing the input sequence, further coding with non adaptive quantizer (coder or encoder-decoder) and finally by performing the denormalization procedure with the same quantized value of the calculated gain that was used for normalizing (see Figure 1) $[1, 8]$. The same effect can be achieved by using an adaptive quantizer with a code book obtained by multiplying the code book of the aforementioned non adaptive quantizer with the quantized value of the estimated gain [7 ,8]. Here we have destined to base our algorithm on such a coding scheme (see Figure 2) consisting of a buffer, an adaptive N-level piecewise linear optimal compandor (PLOC), the gain estimator and the N_g-level scalar quantizer for gain quantizing (SQ_{gq}). Particularly, we have decided to implement the log-uniform scalar quantizer (for gain quantizing), since we have recently demonstrated that it could provide higher $SQNR$ than the uniform scalar quantizer [15]. Moreover, we have destined to consider the piecewise linear optimal compandor, since it has already been pointed out that although the smooth and differentiable nonlinear compressor characteristics are convenient for mathematical manipulations, there are problems of accurately implementing analog nonlinearities [7]. Particularly, it has been ascertained that the solution to these problems provides today's technology which allows the implementation of uniform quantizers with a piecewise linear compressor characteristics that can approximate the smooth compressor curve.

The design procedure of the proposed coding scheme consists of the following steps:

Step 1. Design of non adaptive (fixed) PLOC for the reference variance ($\sigma_{\mathrm{ref}}^2 = 1$) is based on finding the segment thresholds denoted by t_j, $j = -L, -L+1, \dots, L$. Namely, a piecewise uniform quantizer is a quantizer whose amplitude range consists of several segments, each of which contains several quantization cells and output levels corresponding to a uniform quantizer [7]. It is well known that by dividing the output region of the N-level compandor to $2L$ equidistant regions, where L is the number of segments (regions) in the first quadrant, the region of input signal is, according to the obtained piecewise linear compressor characteristic, divided in $2L$ non equal segments each of which contains $n = N/2L$ quantization cells and output levels that correspond to a uniform quantizer [7].

Step 2. Buffering of the input signal and the gain estimation — Buffering frame after frame enables an estimation of the gain, defined as $\sigma/\sigma_{\mathrm{ref}}$, *ie* as a ratio of the squared root of the frame variance and squared root of the reference variance [7]

$$g = \frac{\sqrt{\frac{1}{M}\sum_{i=0}^{M-1} x_{j+1}^2}}{\sqrt{\sigma_{\mathrm{ref}}^2}}\,, \tag{1}$$

where a frame consists of a certain number of samples x_{j+i}, $i = 0, 1, \dots, M-1$.

Step 3. Quantization of the estimated gain by using the N_g-level log-uniform scalar quantizer

$$\begin{gathered} 20\log(\hat{g} = \hat{g}_k) = 20\log\sigma_{min} + (2k-1)\frac{\Delta^{lu}}{2}\,, \\ k = 1, \dots, N_g\,, \Delta^{lu} = \frac{20\log\frac{\sigma_{\max}}{\sigma_{\min}}}{N_g}\,, \end{gathered} \tag{2}$$

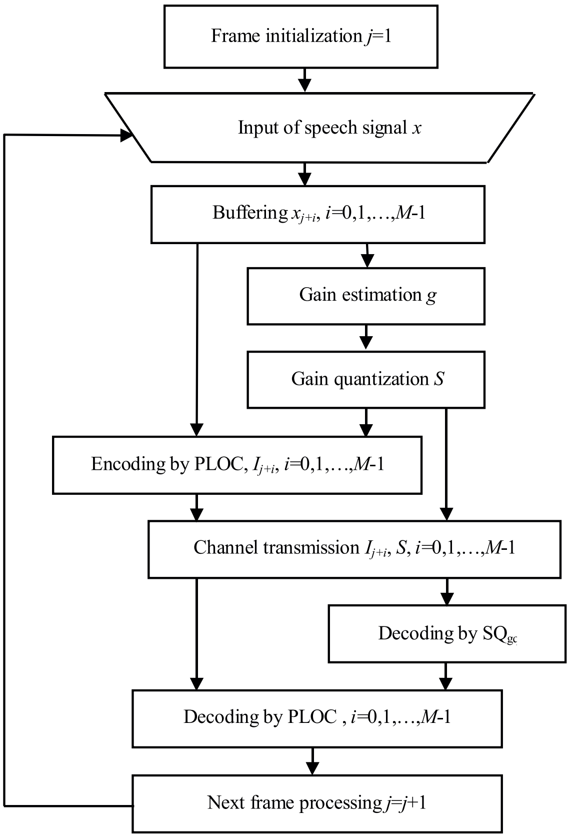

Fig. 3. Forward adaptive PLOC coding algorithm

where the variance range of the input signal in decibels $(20\log\sigma_{\min}, 20\log\sigma_{\max})$ is divided into N_g cells having equal lengths Δ^{lu}.

Step 4. Design of adaptive PLOC — The finding of decision thresholds of the adaptive PLOC, denoted by t_j^a, is enabled by multiplying the segment thresholds of the non adaptive PLOC (obtained in the step 1) with the quantized gain $\hat{g}$

$$t_j^a = t_j^a\left(\hat{g}, \sigma_{\mathrm{ref}}\right) = \hat{g}t_j\left(\sigma_{\mathrm{ref}}\right), \ j = -L, -L+1, \dots, L. \quad (3)$$

According to the described design procedure and the block diagram shown in Fig. 3, we can summarise that the novel coding algorithm firstly performs buffering of the current frame, after which it performs an estimation of the gain for the considered frame. The estimated gain is further quantized by the log-uniform quantizer having N_g levels, and then, the quantized gain obtained in such a manner is used for adjusting the codebook of the forward adaptive PLOC. Next, the encoding procedure is performed for the current frame and the code word index I is obtained as the result of the encoding process. Since the encoding procedure at the transmitting end of the transmission system is normally followed by the decoding procedure at the receiving end of the transmission system, the information about the quantized gain is necessary for the decoder. Therefore, the side information S is sent along with the codeword index I. After the side information is decoded, the decoding procedure that results in the output of the forward adaptive PLOC quantizer is provided. We believe that the proposed algorithm is very simple and suitable for implementation and therefore easy for practical use. In order to test the proposed algorithm, we will consider its application in speech coding.

3 APPLICATION IN SPEECH CODING

The quality of a quantizer (coder) is usually measured by the distortion of the resulting reproduction $\hat{x}$ in comparison to the original signal x. Observe that the distortion introduced by the proposed coding solution [7]

$$D = D(\sigma, \hat{g}, \sigma_{\mathrm{ref}}) = 2\sum_{i=1}^{L} \frac{\Delta_j^{a\,2}}{12} P_j^a, \quad (4)$$

depends on the PLOC quantization step sizes Δ_j^a

$$\Delta_j^a = \Delta_j^a(\hat{g}, \sigma_{\mathrm{ref}}) = \frac{t_j^a(\hat{g}, \sigma_{\mathrm{ref}}) - t_{j-1}^a(\hat{g}, \sigma_{\mathrm{ref}})}{n}, \quad j = -L, -L+1, \dots, -1, 1, 2, \dots, L, \quad (5)$$

as well as on the probability that the input lies in the j-th segment

$$P_j^a = P_j^a(\sigma, \hat{g}, \sigma_{\mathrm{ref}}) = \int_{t_{j-1}^a(g, \sigma_{\mathrm{ref}})}^{t_j^a(\hat{g}, \sigma_{\mathrm{ref}})} p(x, \sigma)\mathrm{d}x, \quad j = -L, -L+1, \dots, -1, 1, 2, \dots, L. \quad (6)$$

Let us now define the signal to quantization noise ratio dependence on the signal variance in order to ascertain whether G.712 standard can be satisfied with the proposed coding algorithm [7, 8]

$$SQNR = 10\log\frac{\sigma_i^2}{D(\sigma_i)}. \quad (7)$$

Additionally, let us define both, the theoretical and the experimental average signal to quantization noise ratio $SQNR_a$, $SQNR_a^{\mathrm{ex}}$ [8] in order to provide an instant comparison of the theoretical and the experimental results

$$SQNR_a = \frac{1}{k}\sum_{i=1}^{k} 10\log\frac{\sigma_i^2}{D(\sigma_i)}, \quad (8)$$

$$SQNR_a^{\mathrm{ex}} = 10\log\frac{\sum_{p=1}^{F}\sum_{q=1}^{M} x_{pq}^2}{\sum_{p=1}^{F}\sum_{q=1}^{M}(x_{pq} - y_{pq}^a)^2}, \quad (9)$$

where k denotes the number of the particular variances that are considered, while x_{pq} and $\hat{x}_{pq} = y_{pq}^a$ denote the input samples and the outputs of the adaptive PLOC quantizer, respectively. In order to provide a more detailed analysis of the performance achieved by performing an experiment on the real speech signal, we have decided

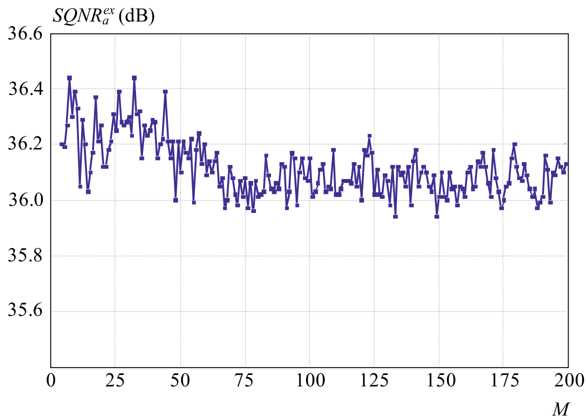

Fig. 4. Experimental results: Dependence of $SQNR_a^{\mathrm{ex}}$ on the frame length M ($W = 10200$, $N = 128$, $N_g = 32$)

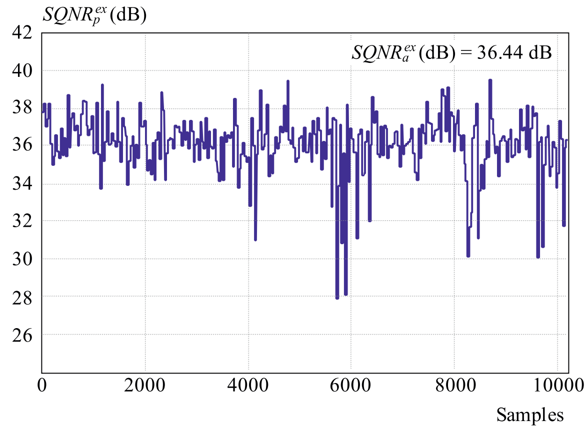

Fig. 5. Experimental results: Illustration of the $SQNR_p^{\mathrm{ex}}$ vicissitude through the frames having length $M = 33$ ($W = 10200$, $N = 128$, $N_g = 32$, $R = 7.15$ bits/sample)

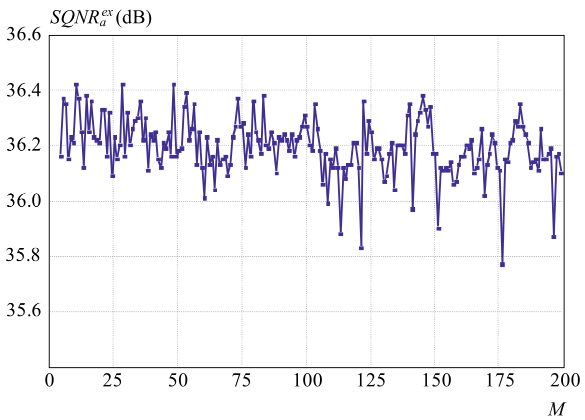

Fig. 6. Experimental results: Dependence of $SQNR_a^{\mathrm{ex}}$ on the frame length M ($W = 10200$, $N = 128$, $N_g = 64$)

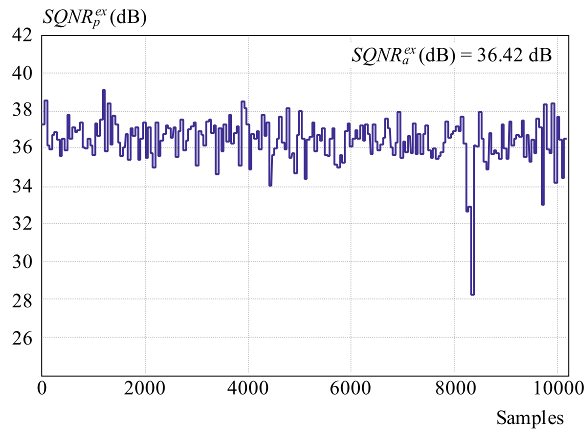

Fig. 7. Experimental results: Illustration of the $SQNR_p^{\mathrm{ex}}$ vicissitude through the frames having length $M = 49$ ($W = 10200$, $N = 128$, $N_g = 64$, $R = 7.12$ bits/sample)

to define the average signal to quantization noise ratio within the each of F frames (each having M samples)

$$SQNR_p^{\mathrm{ex}} = 10 \log \frac{\frac{1}{M}\sum_{q=1}^{M} x_{pq}^2}{\frac{1}{M}\sum_{q=1}^{M}(x_{pq} - y_{pq}^a)^2}\,, \; p = 1, \ldots, F\,. \tag{10}$$

Assuminig Laplacian distribution of the input speech signals [7]

$$p(x,\sigma) = \frac{\sqrt{2}}{2\sigma} e^{-\frac{|x|\sqrt{2}}{\sigma}}\,, \tag{11}$$

and taking into account the expression for the optimal compressor function that was derived for such a distribution [7]

$$c(x)_{\mathrm{opt}} = x_{\max} \frac{1 - \exp\left(-\gamma_c \frac{x}{x_{\max}}\right)}{1 - \exp\left(-\gamma_c\right)}\,, \tag{12}$$

$\gamma_c = C/3$, $C = x_{\max}/\sigma_{\mathrm{ref}}$, we have derived the following experession for the positive segment thresholds of the nonadaptive PLOC

$$t_j = \frac{x_{\max}}{\gamma_c} \ln \frac{L}{L - j(1 - \exp(-\gamma_c))}\,, \; j = 0, 1, \ldots, L\,. \tag{13}$$

Note that the segment thresholds are symetric to the corresponding counterparts. Finally, combining the set of equations (1), (2), (3), (13) with the result from [16], where the maximum input signal dependence on the number of scalar compandor quantization levels N was ascertained for the input speech signal modeled by the Laplacian distribution

$$x_{\max} = \frac{3}{\sqrt{2}} \ln(N - 2)\,, \tag{14}$$

one can complete the design procedure, described in the previous section.

Let us now define the bit rate R corresponding to the proposed coding algorithm

$$R = R_N + \frac{R_{N_g}}{M}\,. \tag{15}$$

Take a notice of the fact that by R_N, we have denoted the number of bits per sample required for quantizing by N-level PLOC. Further, by R_{N_g}, we have denoted the number of bits per frame having length M that is required for the quantizing by N_g-level log-uniform quantizer. From the last equation one can deduce that the decrease of the frame length M results in an unwanted increase of the bit rate R.

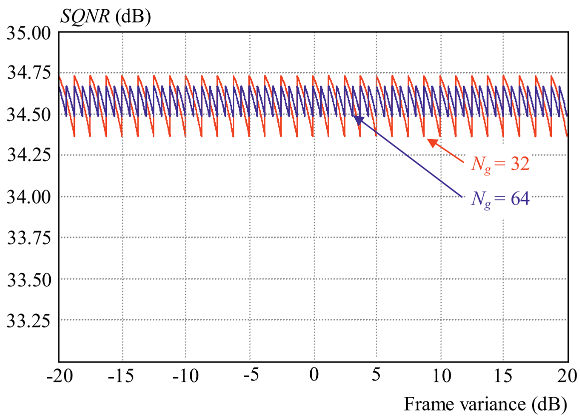

Fig. 8. Theoretical results: Signal to quantization noise ratio as a function of the frame variance

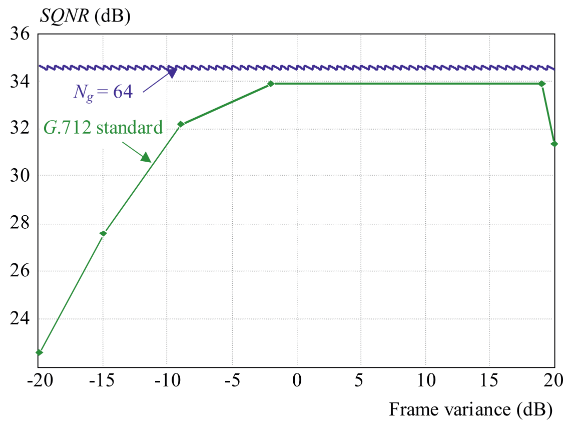

Fig. 9. Comparison of the theoretical results with the G.712 standard

4 NUMERICAL RESULTS

What is presented and discussed in this section are the theoretical and the experimental results that we have achieved with the proposed coding algorithm, as well as the benefits over the performance achievable by the coding solution designed according to G.711 standard. Namely, we have considered forward adaptive PLOC quantizer having $L = 8$ segments in the first quadrant and $N = 128$ quantization levels. Moreover, we have considered the log-uniform quantizer for gain quantizing having $N_g = 32$ and $N_g = 64$ quantization levels, respectively. Note that we have assumed 40 dB range of the frame variances [7] while, according to Eqs. (7) and (8), we were obtaining the theoretical results. In addition, we have been disposed of $W = 10200$ real speech samples (8 kHz speech) while, according to Eqs. (9) and (10), we were obtaining the experimental results. It is important to point out that, as it has been demonstrated in [1, 7, 8, 17, 18], the choice of the suitable frame length M is not a simple task and it usually results from the rate – quality compromise (or rate – $SQNR$ compromise). Regarding the dependance of the $SQNR_a^{\rm ex}$ on the frame length M (see Figure 4), obtained in the case of implementation of the log-uniform quantizer having $N_g = 32$ quantization levels in the proposed algorithm, one can notice two prominent picks corresponding to the frame lengths $M = 8$ and $M = 33$, respectively. Since for the both frame lengths the same $SQNR$ has been achieved ($SQNR_a^{\rm ex} = 36.44$ dB), we have assumed the frame length $M = 33$ as a better rate – quality solution for coding of the considered speech signal. For the assumed frame length $M = 33$, we have ascertained the $SQNR_p^{\rm ex}$ vicissitude through the frames (see Figure 5). Similarly, for the log-uniform quantizer having $N_g = 64$ quantization levels we have noticed three prominent picks at the frame lengths $M = 11$, $M = 29$ and $M = 49$ (see Figure 6), which correspond to the highest $SQNR$ ($SQNR_a^{\rm ex} = 36.42$ dB) that can be achieved in the considered case of the proposed algorithm. Again, assuming the highest frame length in order to provide rate – quality compromise, we have provided the $SQNR_p^{\rm ex}$ vicissitude through the frames having length $M = 49$ (see Figure 7). Additionally, assuming the appropriate compromise frame lengths ($M = 33$ in case of $N_g = 32$ and $M = 49$ in case of $N_g = 64$) we have provided the theoretical results (see Figures 8 and 9). From these figures, one can conclude that the proposed algorithm at the considered bit rates completely satisfies the G.712 standard, since it provides overreaching the curve defined by the G.712 standard in the whole of variance range. From the experimental results (see Figures 5 an 7), which have been obtained for the appropriate compromise frame lengths, one can ascertain that the proposed algorithm provides the average $SQNR$ which is about 3.5 dB greater than the average $SQNR$ achievable by the G.711 coding solution $SQNR_a^{G.711} = 32.94$ dB [9]. Additionally, one can highlight the fact that, in comparison to the solution designed according to the G.711 standard [9], our algorithm provides not only higher level of the average signal to quantization noise ratio, but also performs the reduction of the PCM bit rate for about 1 bits/sample. Therefore, we can reasonable believe that our algorithm provides imposing benefits over the G.711 solution. Finally, since the complexity of the quantizer implemented in the proposed algorithm is the least possible we believe that the proposed low complex algorithm is suitable for practical use.

5 CONCLUSION

This paper has demonstrated the significance of the novel coding algorithm development via the gains in the compression and the performance that have been ascertained by applying the proposed algorithm in speech coding over the G.711 standard [9]. Particularly, by comparing the theoretical results obtained for the assumed Laplacian source having a wide variance range with the $SQNR$ characteristic defined by the G.711 standard in the same variance range, it has been revealed that the proposed algorithm, along with about 1 bits/sample compression, provides a higher level of $SQNR$ which can be considered via the gain in the average $SQNR$ of about 1.62 dB. As with the theoretical results, the experimental results obtained by processing the real speech signal have revealed the average $SQNR$ gain of about 3.5 dB over the G.711 standard. Since it has been ascertained that the proposed coding algorithm satisfies the G.712 standard [14] in the whole of the considered variance range, one can believe that the proposed algorithm will find its practical implementation in the high quality coding of signals, represented with less than 8 bits/sample, which as well as speech signals follow Laplacian distribution and have the time varying variances [17, 18].

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Jelena Nikolić was born in Prokuplje, Serbia, in 1978. She received the BSc degree from the Faculty of Electronic Engineering, Nis, in 2003, and MSc degree from the University of Nis, in 2006. She is currently a teaching assistant at Faculty of Electronic Engineering at the Department of Telecommunications. Her current research interests include the information theory, source coding and signal processing. She is author and coauthor in over 40 papers.

Zoran Perić was born in Nis, Serbia, in 1964. He received the BSc degree from the Faculty of Electronic Engineering, Nis, in 1989, and MSc degree from the University of Nis, in 1994. He received the PhD degree from the University of Nis, in 1999. He is currently a full professor at the Department of Telecommunications and vicedean of the Faculty of Electronic Engineering, University of Nis, Serbia. His current research interests include the information theory, source and channel coding and signal processing. He is particularly working on scalar and vector quantization techniques in speech and image coding. He is author and coauthor in over 150 papers in digital communications. Dr Peric has been a reviewer for IEEE Transactions on Information Theory, The International Journal for Computation and Mathematics in Electrical and Electronic Engineering (COMPEL) and Informatica. He is member of the Editorial Board of the Journal Electronics and Electrical Engineering.

Dragan Antić was born in Vranje, Serbia, in 1963. He received the BSc degree from the Faculty of Electronic Engineering, Nis, in 1987, and MSc degree from the University of Nis, in 1991. He received the PhD degree from the University of Nis, in 1994. He is currently a full professor at the Department of Automation and also dean of the Faculty of Electronic Engineering, University of Nis, Serbia. His current research interests include the sliding mode control, nonlinear systems, modeling and simulation of dynamic systems, bond graphs. He is author and coauthor in over 150 papers.

Aleksandra Jovanović was born in Nis, Serbia, in 1971. She received her BSc, MSc and PhD degrees in Electrical Engineering from the Faculty of Electronic Engineering (Department of Telecommunications), University of Nis, Serbia, in 1995, 1999 and 2007, respectively. Her field of interest includes digital telecommunication, information theory and source coding. She has authored and coauthored over 50 scientific papers.

Dragan Denić was born in Vranje, Serbia, in 1963. He received the BSc degree from the Faculty of Electronic Engineering, Nis, in 1988, and MSc degree from the University of Nis, in 1992. He received the Ph.D. degree from the University of Ni, in 1996. He is currently a full professor at the Department of Measurement, University of Ni, Serbia. He is the author or coauthor of more than 50 papers. As expert he was involved in period of more than 3 years as independent consultant of GPI (Gurley Precision Instruments), leading company in his field of research. His main fields of interest are the measurement science, position measurement, telemetry, computer based measurement systems.