New Adaptive Compandor for LTE Signal Compression Based on Spline Approximations

Lazar Zoran Velimirovic and Svetislav Maric

With the constant increase in network traffic, wireless operators are finding it more challenging to keep network hardware costs to a minimum. At the same time, the energy cost associated with operating a network has increased proportionally. Therefore, the search for higher network capacity is simultaneously accompanied by the search for a cost-efficient network deployment. In this paper, we show that a saving in transmitted signal energy can be achieved at the signal design level by deploying very specific signal processing techniques. Using an orthogonal frequency-division multiplexing signal for Long-Term Evolution networks as an example, we utilize a novel non-uniform companding quantizer to save a transmitted signal energy. Our results show that by using non-uniform quantization it is possible to further optimize 4G wireless networks.

Keywords: LTE signal, adaptive quantization, compandor, gain quantization, C-RAN, CPRI.

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I. Introduction

With the deployment of advanced Long-Term Evolution (LTE) techniques (for example, carrier aggregation) in 4G networks, the amount of transmitted traffic has increased significantly. With this increase, network operator's hardware costs and associated energy costs have also increased. To counter these rising costs, a new Centralized Radio Access Network (C-RAN) architecture has been proposed [1]. Within such an architecture, baseband radio units are centrally located in a pool configuration and are connected to remote radio units through optical fibers. Note that this is a significant difference in comparison to previously distributed cellular networks.

In this paper, we discuss how to reduce the amount of traffic carried between nodes (that is, points where remote radio units are connected with optical fibers) by modifying the transmitted orthogonal frequency-division multiplexing (OFDM) signal, in line with the Common Public Radio Interface (CPRI) standard [2]. One way to do so is through the use of quantizers. The main goal in a quantizer design is to select a point where a quantizer will be jointly optimized in terms of bit rate, errors in the reconstructed signal, signal delay, and complexity of a practical realization [3].

Using a non-uniform companding quantizer that delivers a performance rate similar to that of an optimal companding quantizer in real time and without increasing the number of quantization levels, the aforementioned goal can be achieved.

To design and realize a non-uniform companding quantizer, it is necessary to determine the optimal inverse compressor function and the complementary characteristics of both the compressor and the expandor. Both the design and practical realization of a non-uniform companding quantizer are very complex [3]. To simplify the design procedure for a non-uniform companding quantizer and achieve an easier practical realization (in both hardware and software), linearization and approximation of the optimal compressor function are implemented in [4].

There is a new class of compandor, whose design is based on an approximation of the optimal compressor function by first-degree spline functions. The results in this paper show that our proposed novel non-uniform companding quantizer (quantizer model) represents an effective solution in achieving high-quality quantized signals. The proposed non-uniform companding quantizer, being of the adaptive type and based on first-degree spline functions, is created through an adaptation of a fixed quantizer, which is achieved by adjusting the fixed quantizer's input according to an estimated gain. The technique of forward adaptive quantization is applied to the proposed non-uniform companding quantizer (thereby creating what can be termed a "*forward-adaptive* non-uniform companding quantizer"). The simplicity of this novel forward-adaptive non-uniform companding quantizer can be seen in the fact that it consists of only $2L = 4$ segments (L is the number of segments the support region is divided into), and $N = 128$ quantization levels (N is the number of quantization levels the support region is divided into).

The proposed quantizer has significantly lower complexity compared to a symmetric piecewise uniform scalar quantizer, defined in the G.711 standard as comprising $2L = 16$ segments at 8 bits/sample. The proposed novel forward-adaptive non-uniform companding quantizer also satisfies the G.712 standard at 7 bits/sample [5], [6].

This paper is organized as follows. In the next section, we review the C-RAN architecture, and describe the motivation for signal optimization. We also describe our proposed novel forward-adaptive non-uniform companding quantizer. The results are presented in Section III, and we show that savings in bit rate and complexity of the quantizer design are achieved due to our proposed forward-adaptive non-uniform companding quantizer. The average bit rate is smaller than that required by the G.711 and G.712 standards, at around 1 bits/sample. Further, a higher signal quality compared to G.711 and G.712 quantizers is achieved using a segmental signal-to-quantization noise ratio (SQNR_{seg}), [5], [6]. We conclude the paper with a brief summary of our results.

II. Methodology

1. C-RAN and Corresponding C-RAN Interface

In 4G LTE networks, due to the standard design of the base stations, it is no longer necessary to have both a baseband unit (BBU) and a remote radio head (RRH) collocated in a single cell site. To make use of this new architecture (4G LTE), a C-RAN was introduced. A C-RAN allows for the removal of all BBUs from their respective cell sites, and their co-location at a new site (centralization). To ensure a reliable signal reception, RRHs are left unmoved from their respective cell sites; thus, the C-RAN must also accommodate the connection of the now co-located BBUs with the RRHs through a large number of fiber-optic cables. For a downlink connection, a baseband LTE signal is generated in a BBU and then transferred to an RRH in accordance with the CPRI [2].

Various techniques have been proposed to reduce the amount of network traffic, and consequently decrease the cost of fiber optics. For instance, in [7], the authors optimized a transmitted OFDM signal by removing signal redundancy, and performing uniform quantization.

In this paper, we further develop the idea of removing signal redundancy by introducing a novel forward-adaptive non-uniform companding quantizer specifically designed to be used with LTE control signals. This, in turn, reduces the traffic between the RRHs and the BBUs, thus reducing the overall cost of a network.

2. Design of Adaptive Compandor

The coding schemes shown in Figs. 1 and 2 consist of the following blocks: buffer, non-adaptive N-level scalar quantizer, N_g-level scalar quantizer (for gain quantization), and gain estimator.

The aforementioned adaptation procedure is performed as follows: First, the input sequence with a quantized value of the calculated gain is normalized. Second, coding with the fixed quantizer is performed. Finally, the same value of the gain that is used for normalizing is denormalized.

Division of an LTE input signal, $x[n]$, into M frames is carried out by a buffer [8]. A fixed quantizer is defined according to a set of decision thresholds, $\{x_1, x_2, \ldots, x_{N-1}\}$, and a set of representation levels, $\{y_1, y_2, \ldots, y_{N-2}\}$. The decision thresholds and representation levels of the fixed quantizer (by which we design our compandor model), are determined in the same way as in [9].

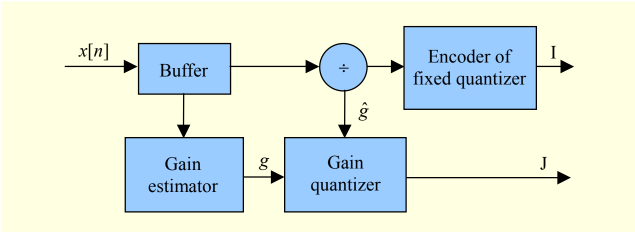

Fig. 1. Block diagram of encoder.

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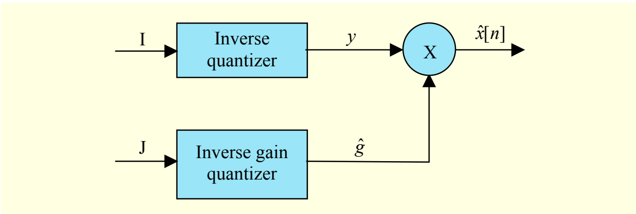

Fig. 2. Block diagram of decoder.

The signal quality analysis performed in [9] shows that decision thresholds and representation levels determined in this way achieve a signal quality that is very similar to that achieved by an optimal compandor. Therefore, in this paper, we similarly determine the decision thresholds and representation levels of a fixed quantizer in the same way as in [9].

We can determine the decision thresholds and representation levels of a forward-adaptive non-uniform companding quantizer through the use of a nonlinear compressor function, $c(x)$, which we define as follows [3]:

$$c(x) = x_{\max} \operatorname{sgn} x \frac{\int_0^{|x|} p^{\frac{1}{3}}(x)\,\mathrm{d}x}{\int_0^{x_{\max}} p^{\frac{1}{3}}(x)\,\mathrm{d}x}, \qquad (|x| \le x_{\max}). \tag{1}$$

It is well known that the inverse function of the nonlinear compressor function $c(x)$ is difficult to determine, that is, $c^{-1}(x)$. This is due to the fact that it is necessary to solve systems of nonlinear integral equations.

In (1), $p(x)$ is the Laplacian probability density function of the unit variance ($\sigma = 1$), and is written as follows [3], [10]:

$$p(x) = \frac{1}{\sqrt{2}} e^{-|x|\sqrt{2}}. \tag{2}$$

The granular region threshold, $x_{\max}$, of the companding quantizer defined in [9] is as follows:

$$x_{\max} = \frac{3}{\sqrt{2}} \ln\left(\frac{N+1}{3}\right). \tag{3}$$

The solution to determining $c^{-1}(x)$ is described in [9]. To simplify the determination of $c^{-1}(x)$, or to make its practical realization simpler, an approximation of the nonlinear compressor function $c(x)$ using a first-order spline function, $s(x)$, is performed as follows [9], [11]:

$$\begin{aligned} s_i(x) &= c(x_i) + \frac{c(x_i) - c(x_{i-1})}{x_i - x_{i-1}}(x - x_i), \\ x &\in [x_{i-1}, x_i], \quad x_i = s_i^{-1}\left(i\frac{N}{2L}\right), \end{aligned} \tag{4}$$

where $i = 0, \ldots, L-1$, and it obviously holds for $x_L = x_{\max}$.

For $s(x)$ determined in this way, we design our compandor. The decision thresholds are as follows:

$$x_{i,j} = s_i^{-1}\left(j\Delta^l\right), \qquad i = 1,\ldots,L, \quad j = 1,\ldots,\frac{N_i}{2}, \tag{5}$$

whereas the representation levels are

$$y_{i,j} = s_i^{-1}\left(\frac{2j-1}{2}\Delta^l\right), \qquad i = 1,\ldots,L, \quad j = 1,\ldots,\frac{N_i}{2}. \tag{6}$$

The total number of reproduction levels for all segments is as follows:

$$\sum_{i=1}^{L} N_i = N - 2, \tag{7}$$

where the number of reproduction levels per segment is given by $N_i/2 = N/2L$, $i = 0, \ldots, L-1$. In the last segment, the number of reproduction levels is equal to $N_L/2 = N/2L - 1$.

The step-size value in (6) is determined as follows:

$$\Delta^l = \frac{2x_{\max}}{N-2}. \tag{8}$$

A gain, g, is estimated for each particular frame of a signal, where each frame comprises exactly M samples, as follows [12], [13]:

$$g = \sqrt{\frac{1}{M}\sum_{i=1}^{M} x_i^2}. \tag{9}$$

The estimated gain can be determined in two ways. In the first of such ways, g is quantized by a uniform scalar quantizer, $\hat{g}$, with N_g quantization levels as follows [12], [13]:

$$\begin{aligned} \hat{g}_i &= \sigma_{\min} + (2i-1)\frac{\Delta}{2}, \qquad i = 1,2,\ldots,N_g, \\ \Delta &= \frac{\sigma_{\max} - \sigma_{\min}}{N_g}, \end{aligned} \tag{10}$$

where interval ($\sigma_{\min}$, $\sigma_{\max}$) denotes the dynamic range of the input signal.

In the second of such ways, g is quantized by a log-uniform scalar quantizer, $\hat{g}$, with N_g quantization levels as follows [12], [13]:

$$\begin{aligned} 20\log_{10}(\hat{g}_i) &= 20\log_{10}(\sigma_{\min}) + (2i-1)\frac{\Delta}{2}, \qquad i = 1,2,\ldots,N_g, \\ \Delta &= \frac{20\log_{10}(\sigma_{\max}/\sigma_{\min})}{N_g}. \end{aligned} \tag{11}$$

For the sake of ease, in the succeeding equations, we have not introduced special marks for the quantized gain $\hat{g}$, which is determined depending on how g is to be quantized. However, in Section III, the results shown take into consideration both of the two ways in which to determine the quantized gain $\hat{g}$.

By multiplying both the decision thresholds and the representation levels of the fixed quantizer by the quantized gain $\hat{g}$, we can determine the same such thresholds and levels of an adaptive quantizer, $x_{i,j}^a$ and $y_{i,j}^a$, respectively, as follows [12], [13]:

$$x_{i,j}^a = \hat{g} \cdot t_{i,j}, \qquad y_{i,j}^a = \hat{g} \cdot y_{i,j}. \tag{12}$$

The granular distortion, D_{g_j}, used to describe the quality of a signal, is defined by [3]

$$D_{g_j} = 2\sum_{i=1}^{L}\sum_{j=1}^{\frac{N_i}{2}} \frac{\left(\Delta_{i,j}^a\right)^2}{12} P_{i,j}^a, \tag{13}$$

where the length of an adaptive cell is given by the following equation:

$$\Delta_{i,j}^a = \hat{g} s_i^{-1}\left(j\Delta^l\right), \qquad i = 1,\dots,L, \quad j = 1,\dots,\frac{N_i}{2}. \tag{14}$$

Combining (2) and (12), we can derive the following closed-form expressions for the adaptive probablities, $P_{i,j}^a$ [3]:

$$\begin{aligned} P_{i,j}^a &= \int_{x_{i,j-1}^a}^{x_{i,j}^a} p(x)\,\mathrm{d}x \\ &= \frac{1}{2}\left[\exp\left(-\frac{\sqrt{2}x_{i,j-1}^a}{\sigma}\right) - \exp\left(-\frac{\sqrt{2}\,x_{i,j}^a}{\sigma}\right)\right]. \end{aligned} \tag{15}$$

The overload distortion, D_o, is defined by [3]

$$\begin{aligned} D_{o_j} &= 2\int_{x_{\max}^a}^{\infty} \left(x - y_N^a\right)^2 p(x)\,\mathrm{d}x \\ &= \exp\left(-\frac{\sqrt{2}x_{\max}^a}{\sigma}\right)\left(\left(x_{\max}^a - y_N^a + \frac{\sigma}{\sqrt{2}}\right)^2 + \left(\frac{\sigma}{\sqrt{2}}\right)^2\right), \end{aligned} \tag{16}$$

where the adaptive granular region threshold is represented by $x_{\max}^a$, which is as follows:

$$x_{\max}^a = \hat{g} x_{\max}. \tag{17}$$

The last adaptive representation level is represented by y_N^a, which is determined from the centroid condition [10] as $y_N^a = \hat{g}\, y_N$.

From the granular and overload distortions, as described by (13) and (16), the total distortion can be determined. The total distortion, D_i, at the ith frame, is equal to

$$D_i = \frac{1}{M}\sum_{j=1}^{M}\left(D_{g_j} + D_{o_j}\right), \qquad i = 1, 2, \dots, k. \tag{18}$$

Here, σ_i^2 is the variance of an LTE input signal at the ith frame and is given by

$$\sigma_i^2 = \frac{1}{M}\sum_{j=1}^{M} x_j^2, \qquad i = 1, 2, \dots, k, \tag{19}$$

where M is the frame length of the LTE input signal.

The quality of the LTE input signal is determined by SQNR_{seg}, where SQNR_{seg} is considered to be a better perceptual model compared to the traditional signal-to-quantization noise ratio, since it evaluates the quantization noise with respect to the energy in each underlying LTE segment [12]. We define SQNR_{seg} as follows:

$$\text{SQNR}_{\text{seg}} = \frac{1}{k}\sum_{i=1}^{k} 10\log\left(\frac{\sigma_i^2}{D_i}\right), \tag{20}$$

where k denotes the total number of frames.

For the number of levels, N, of the adaptive compandor; the number of gain quantization levels, N_g; and the frame length of the LTE input signal, M, the bit rate, R, of the proposed adaptive compandor is defined as follows [14]:

$$R = \log_2 N + \frac{\log_2 N_g}{M}. \tag{21}$$

III. Numerical Results and Discussion

The numerical results presented in this section are obtained for the case where the number of segments is equal to $2L = 4$ and the number of levels is equal to $N = 128$. The number of gain quantization levels is $N_g = 2$ (1 bits/frame), $N_g = 4$ (2 bits/frame), $N_g = 8$ (3 bits/frame), $N_g = 16$ (4 bits/frame), and $N_g = 32$ (5 bits/frame).

The results shown in Table 1 and Figs. 3 and 4 are obtained from an LTE input signal sequence 19,200 samples in length. Other LTE input signal parameters are as follows: allocated resource blocks = 6, DFT-OFDM symbols per subframe = 12, modulation = QPSK, code rate = 1/3, maximum throughput (kbps) = 568 (9). The frame length of the LTE input signal is $M = 10$.

Based on the results shown in Table 1 and Fig. 3, it can be seen that with the proposed forward-adaptive non-uniform companding quantizer, and the uniform and log-uniform gain

Table 1. Value SQNRseg for proposed forward-adaptive non-uniform companding quantizer.

N_g level quantizer	SQNR_{seg} for uniform gain quantization (dB)	SQNR_{seg} for log-uniform gain quantization (dB)
2	36.1562	36.0187
4	36.2355	36.2254
8	36.2367	36.2459
16	36.2384	36.2464
32	36.2388	36.2478

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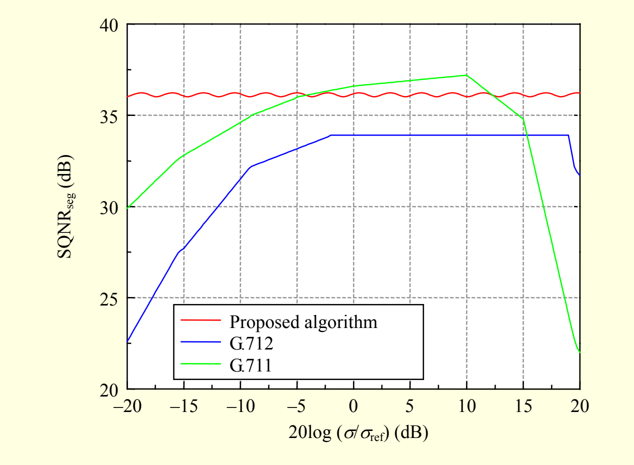

Fig. 3. SQNR$_{seg}$ of proposed forward-adaptive non-uniform companding quantizer with wide variance range compared to G.711 and G.712 standards.

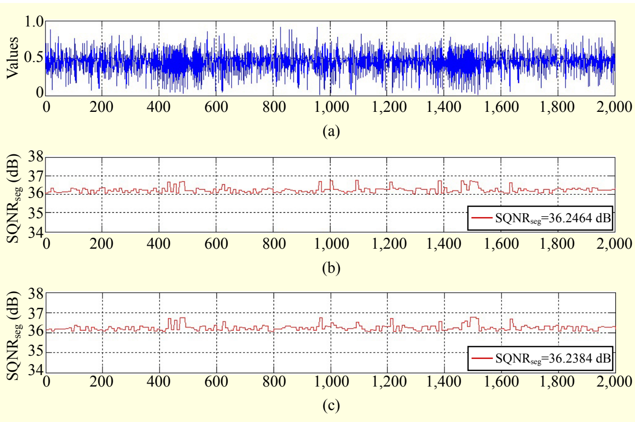

Fig. 4. (a) LTE input signal, (b) value of SQNR$_{seg}$ for $N_g = 16$ log-uniform scalar quantizer, and (c) value of SQNR$_{seg}$ for $N_g = 16$ uniform scalar quantizer.

quantizations, an SQNR$_{seg}$ of 36.24 dB is achieved. Since the SQNR$_{seg}$ value of a G.711 quantizer is 32.29 dB, it is evident that we have achieved a saving of nearly 4 dB. Similarly, with regard to the maximal value of a G.712 quantizer, the saving is around 2.25 dB. It is important to emphasize that the SQNR$_{seg}$ saving is achieved when the bit rate is lower by 1 bits/sample when compared to the G.711 and G.712 quantizers.

From the literature, it is known that an increase in the bit rate of 1 bits/sample produces a 6 dB improvement in the signal quality [15]. Considering that the bit rate for the G.711 and G.712 quantizers is defined as $R = 8$ bits/sample, with one additional bit/sample, the SQNR$_{seg}$ of the proposed forward-adaptive non-uniform companding quantizer would be increased by 6 dB [15]. This means that the total saving would be higher, at around 10 dB, with regard to the G.711 quantizer, and at around 8.2 dB with regard to the G.712 quantizer.

Comparing these results, we can see that the proposed forward-adaptive non-uniform companding quantizer is significantly superior to both the G.711 and G.712 quantizes.

Figure 4 shows the SQNR$_{seg}$ values in the cases of the gain quantization with an $N_g = 16$ *uniform* scalar quantizer and an $N_g = 16$ *log-uniform* scalar quantizer. It is evident from the figure that there is little difference in the two sets of SQNR$_{seg}$ values over the course of the entire LTE input signal.

In addition, based on the results in Table 1, it can be concluded that a high-quality SQNR$_{seg}$ is achieved in the case of an $N_g = 4$ uniform scalar quantizer. In fact, the proposed forward-adaptive non-uniform companding quantizer represents the best solution in terms of the resulting signal quality and complexity of the design. We note that an increase in the number of gain quantization levels (N_g) results in the following: (a) an exponential increase in the design complexity and (b) an insignificant increase in the signal quality. Therefore, when designing the proposed forward-adaptive non-uniform companding quantizer, there is no need to opt for a higher value of N_g.

Because the realization of the proposed forward-adaptive non-uniform companding quantizer is much simpler in the case of a uniform gain quantization than in the case of a log-uniform gain quantization, this is recommended as the best solution, with 2 bits/frame for the gain quantization.

From the standpoint of implementation complexity, it can be said that our proposed forward-adaptive non-uniform companding quantizer is much simpler to implement than the quantizer described in [12]. The quantizer described in [12] has to transfer side-information to a decoder (but not gain information) and information on the chosen quantization model (based on each frame's correlation), thus further increasing its bit rate.

The advantage of our proposed forward-adaptive non-uniform companding quantizer can be seen in comparison with the codec described in [16]. Namely, while the codec described in [16] achieves a better signal quality by 0.5 dB than that achieved by the G.711 standard, at a bit rate of 10 bits/sample, our model achieves a better signal quality by 4.2 dB at around 7 bits/sample. In addition, our proposed forward-adaptive non-uniform companding quantizer is much simpler in terms of practical realization than the codec with vector companding described in [16].

IV. Conclusion

Even with LTE networks nearing maturity, work on their optimization continues. New technologies such as C-RAN make use of the possibility to divide BBUs and RRHs, thus reducing network hardware costs. However, this comes at the expense of increased usage of CPRI resources through a signaling traffic increase.

In this paper, we proposed a novel forward-adaptive non-uniform companding quantizer to compress an LTE signal transmitted over the CPRI. The compression reduces the number of bits necessary to transmit the signal, and in turn reduces CPRI traffic. It was shown that an LTE signal processed in this way achieves significant advantages in terms of signal quality and bit compression, as well as being notably simpler to realize in practice in comparison to the G.711 and G.712 standards and the model described in [16].

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