PERFORMANCE EVALUATION OF THE G.729, AMR AND ILBC VOICE CODECS ON LTE-ADVANCED

Jesús D. Quintero P., Martin D. Bravo O. and José J. Salgado P.
Department of Electronic Engineering, Surcolombiana University, Neiva, Colombia
E-Mail: jdavid@usco.edu.co

ABSTRACT

This article presents a referential framework given by an exhaustive analysis of the quality of the service concerning the G.729, AMR, and iLBC coding standards on LTE-A (LTE-ADVANCED) technology. The architecture that forms the LTE-A technology, making a particular focus on the characteristics that make up its OFDMA interface (Orthogonal Frequency Division Multiple Access). Simulations of the scenarios are implemented, showing data on the traffic generated by a certain number of users. On the other hand, an algorithm was performed to extract the generated packet by the voice encoders and the transmitting according to the networking traffic information; it offered by the simulator. Finally, the objective and subjective evaluations were realized, and the results of the voice service quality on IP (VoIP) are presented, taking into account the packet loss.

Keywords: voice-codecs, LTE-A, quality of service, OFDMA, VoIP.

1. INTRODUCTION

Due to the evolution of the third generation (3G) of cell phone technologies, the voice service is no limited; however, have been imposed the transformation of circuit switching to packet switching. Which improve the voice over IP (VoIP) service offer; as also, the called voice over LTE technology (VoLTE), the latter already revolves around fourth-generation communications (4G). (Labyad, *et al*., 2014)

The packet switching in wireless networks has changed the way of communication between people being a better option in terms of costs and quality. It has led to the creation of applications that provide VoIP services offering different types of qualifying conditions. These services are given in real-time using different kinds of codecs (encoder and decoder), and the users are who give their opinion about the quality of voice service (Abdelrahman *et al*., 2015).

The 4G LTE-A technology is capable of providing services with a large number of subscribers (more excellent than those offered in the 3G), which must establish a series of parameters that define the link. LTE-A uses an OFDMA radio interface (Orthogonal Frequency Division Multiple Access) and has a scalable bandwidth, its bandwidth varies from 1.4 MHz to 20 MHz, and its downlink speed reaches up to 1Gbps while for link rise reaches up to 500 Mbps. (Luna *et al*., 2015).

Despite the benefits such as high transmission rates in the uplink and downlink that LTE-A offers, the transmission of data over the internet suffers from packet losses, this means that not all the information reaches its destination. The transfer of voice when experiencing problems degrades the quality of its signal received by causes such as latency, jitter, and echo. (Opazo, 2008).

Voice codecs perform the analysis and processing of the original voice-signal. Subsequently, the analog voice signal is transformed into a digital signal formed by a sequence of received bits. Each codec uses a particular coding scheme that can reconstruct the voice with a lower loss of quality. Otherwise, redundancy is introduced into the transmission channel in which the packets contained in tiny pieces will be sent, and in an orderly manner, the voice information, this redundancy avoids the distortions introduced by the transmission medium. (Carmona, 2009)
It is vital to perform evaluations that support the quality of service; the primary method to measure the quality of a voice communication service is the subjective evaluation. Its assessment of voice quality consists of user visual perception; however, this method is costly due that requires a quantity of time to carry out their tests. For this reason, the process of objective evaluation arises; this tries to predict the subjective quality from quantitative measurements that are obtained from the signals that will be evaluated. (Carmona, 2009)

This article discusses the analysis and evaluation of the performance of G.729, AMR, and iLBC voice coders using LTE-A technology. Contributing to the construction of a reference for decision making regarding the quality of the voice service taking into account the most representative characteristics of each coder, the parameters that had the most impact at the network level were identified, and the simulations were carried out. The SU-MIMO and MU-MIMO scenarios are guaranteed to the provision of the VoIP service.

2. METHODOLOGY

The development of the research is divided into three stages:

2.1 Stage 1: MU-MIMO and SU-MIMO

The first stage is the implementation using the LTE-A in a link level of the simulator, which is provided by the Vienna Technological University. Then two scenarios are created, one of a single multi-antenna user (SU-MIMO) and another multi-user multi-antenna (MU-MIMO) in the LTE-A technology, which will provide the information to perform the necessary demonstrations.

Table-1 shows the parameters of the two scenarios that were simulated, both with the same type of channel, bandwidth, and number of resources, subcarrier spacing, and cyclic prefix.

Table-1. Simulation parameters.

PARAMETERS	SCENARIOS	
Type of simulation	**LTE-A SUMIMO**	**LTE-A MUMIMO**
Number of subframes	50000	100000
Channel	AWGN	AWGN
Time elapsed	5.641e+04	7.4484e+04
Bandwidth system	3MHz	3MHz
Number of resource blocks	15	15
Subcarrier spacing	15KHz	15KHz
Cyclic prefix	Normal	Normal

2.2 Etapa 2: Coding standards

The second stage is the adaptation of the three codecs standards that were used, G.729, AMR, and iLBC. Each one with their encoder and decoder.

Table-2 shows the characteristics and presents the bit rate that will be used for the encoders. In the case of the AMR coder, it has eight different rates from 4.75 to 12.2 kbps. The iLBC coder has two that are 13.33 and 15.2 kbps; also, it is possible to observe the value of the frame in terms of milliseconds, the bits that have each frame, and the type of compression algorithm used by each encoder.
The second stage is the adaptation of the three coding standards that were used, G.729, AMR and iLBC. Each with its respective encoder and decoder with the exception of the iLBC that its algorithm has implemented encoder and decoder in a single executable, in the case of the other two are given separately.

Table-2 shows the most important characteristics and exposes the binary rate that will be used for the AMR and iLBC encoders since both encoders have more than one. In the case of AMR has eight different rates ranging from 4.75 to 12.2 kbps. iLBC has two that are 13.33 and 15.2 kbps. You can also observe the value of the frame in terms of milliseconds, the bits that have each frame and the type of compression or algorithm used by each encoder.

Table-2. Characteristics of voice coders.

codec	Rates of bit (kbps)	Frame (ms)	Bit/frame	Compression type
G.729	8	10	80	CS-ACELP
AMR	5.15	20	103	ACELP
iLBC	15.2	20	304	LPC

2.3 Stage 3: Objective and subjective evaluations

For the third stage, objective and subjective evaluations were carried out. For this, 6 audio samples were taken, to each of the samples 3 different degradations were made, this means that the packages were extracted to the three samples. This is why the number of lost packages must be calculated in order to carry out these evaluations. To find the value of lost packets, equations 1 and 2 were used.

$$\mathrm{FER} = 1 - (1 - \mathrm{BLER})^{N_{\mathrm{bits}}^{\mathrm{Sf}}} \quad (1)$$

In equation 1, we define:
FER: error rate per frame.
BLER: error rate per resource block.
N_bits ^ Sf: Number of bits per sub frame.

Equation 1 is realized since in the simulation the value of BLER and not of FER is obtained, in this way the number of lost packages can be calculated with equation 2 and a direct connection between BLER and equation 2 is given.

$$Packetlost = \frac{NPA*FER}{100} \quad (2)$$

In equation 2, reference is made to:
Packet lost: number of lost packages.
NPA: number of packages per audio.

The number of lost packets allows us to know the exact extraction value in which a CQI value is guaranteed where the block error rate BLER (Block Error Ratio) is less than or equal to 〖10〗 ^ (- 3).

For the application of the subjective evaluation, the average opinion score MOS (Mean Opinion Score) was used, named by ITU-T in its recommendation P.800. Ten people were interviewed, which had to qualify a total

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of 72 audios each. Each interviewee had to grade each audio as shown in Table-3.

Table-3. Average view score.

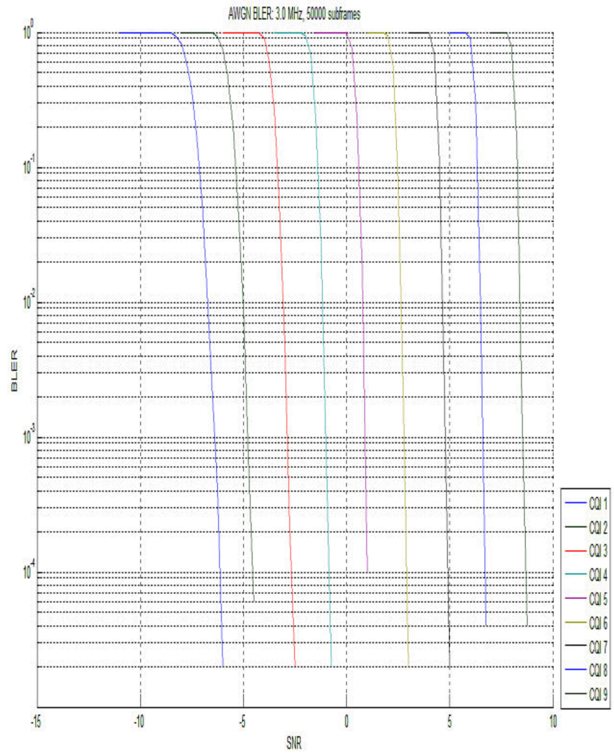
MOS	QUALITY
5	Excellent
4	Good
3	Acceptable
2	Poor
1	Bad

In this way you get a MOS average for each of the qualified audios.

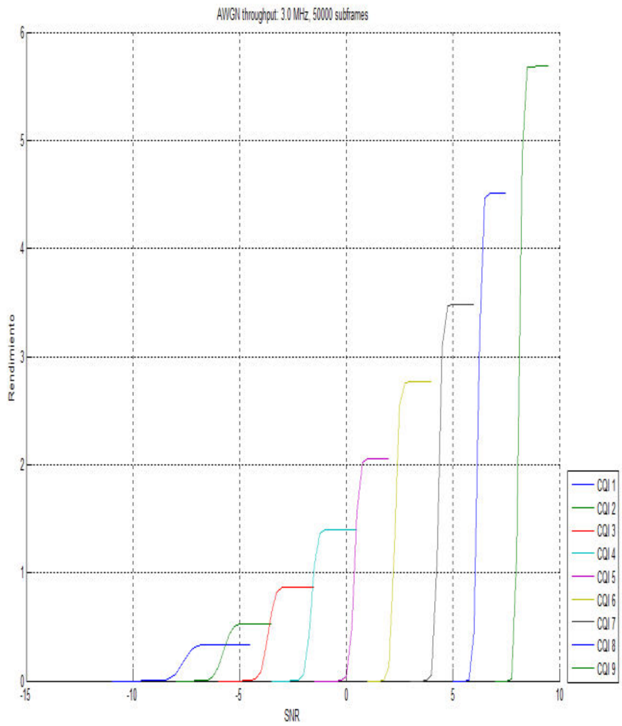
For the objective evaluation, recommendation P.862 was used to evaluate the quality of speech by perception PESQ (Perceptual Evaluation of Speech Quality) standardized by the ITU-T. This algorithm uses the original voice signal and compares it with the degraded one, later it delivers the MOS value. It should be noted that this value tries to be in agreement with the original MOS value that is why it is mediated with great accuracy

3. RESULTS AND DISCUSSIONS

To obtain the block error ratio (BLER) and performance for the modulation and coding scheme (MCS) corresponding to each CQI value, AWGN simulations were performed. The MCS determines both the modulation type and the effective code rate (ECR) of the channel encoder. Figure-1 shows the BLER results of CQIs 1-9. Each curve is spaced approximately 2 dB apart. In Figure-1, the yield curves are plotted for each CQI value. The SNR difference is around 2 dB for most CQI values. Increasing the number of CQI increases the performance of the system. Taking into account that CQIs 1-6 work with QPSK modulation and CQIs 7-9 (16-QAM).



(a)



(b)

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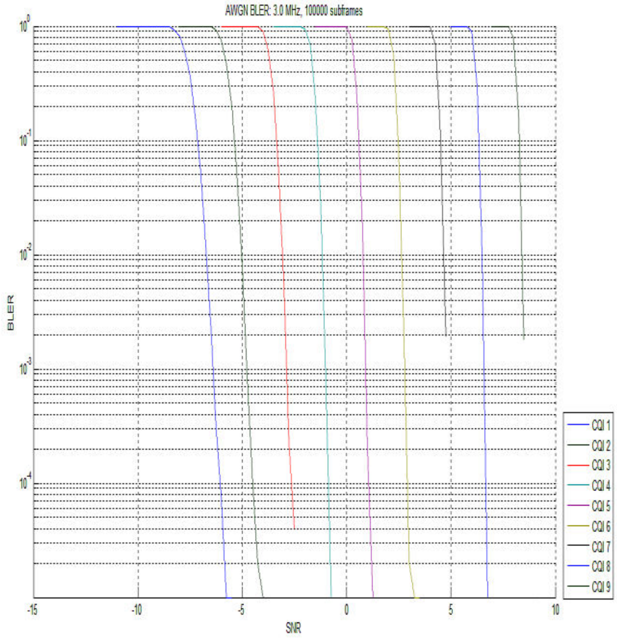
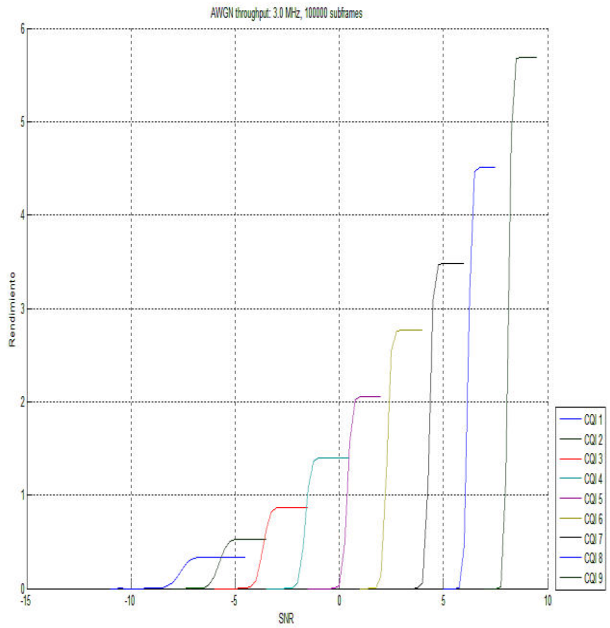


(c)



(d)

Figure-1. (a) SUMIMO performance vs SNR, (b) SUMIMO BLER vs SNR, (c) MUMIMO of performance vs SNR, (d) MUMIMO BLER vs SNR

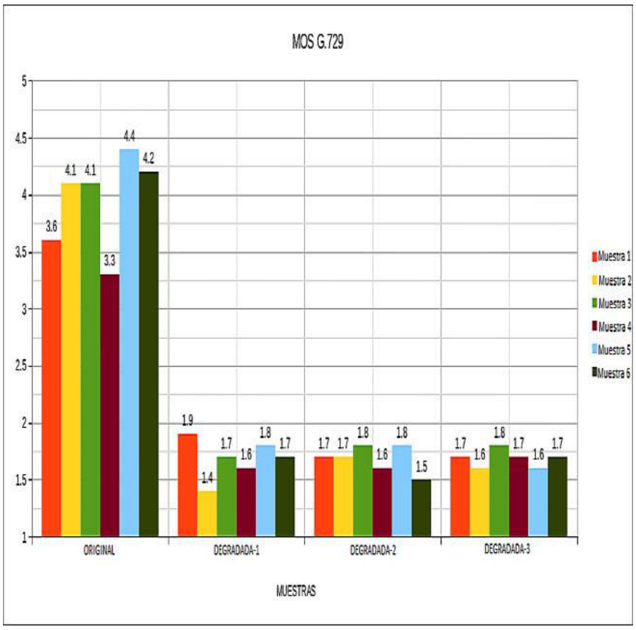
It is not possible to obtain BLER graphs from a multiuser system since the simulator has this limitation, what was done was to choose a user to be able to graph. The simulation took too long in processing, as you can see in the image to be able to guarantee the value of 〖10〗 ^ (- 3), the number of sub frames was increased to 100000.

At a high noise signal ratio the symbol, energy will be lower; therefore, we use a modulation of a lower order to not have a greater loss of information. Due to this, the QPSK and 16-QAM CQIs will be mapped.

Figure-2 shows that the percentage MOS rating of the three coders, figure a) shows the MOS of recommendation G.729. b) Shows the MOS of the AMR and c) shows the MOS of the iLBC.

The encoder G.729 its lowest value falls to 1.4 while in the AMR its lowest value is 2.2 and iLBC has the lowest value 3.3. This demonstrates that the iLBC is more robust compared to the other two encoders, and its highest rating is also highlighted with a value of 4.5 compared to the G.729 of 1.9 and the AMR of 3.5, taking into account that the value is ignored MOS of the original samples.
It also shows that the encoder with the lowest value in its ratings is G.729, although the three encoders have the same percentage of loss their robustness to the losses is very low compared to the other two encoders.

The graph (a) of Figure-2 shows that the highest average MOS value is in the number one degraded samples. On the other hand, in graph (b) the value of the MOS of the AMR encoder is observed in the degraded samples number three and for the graph (c) the iLBC this can be seen in the degraded samples number two.



(a)

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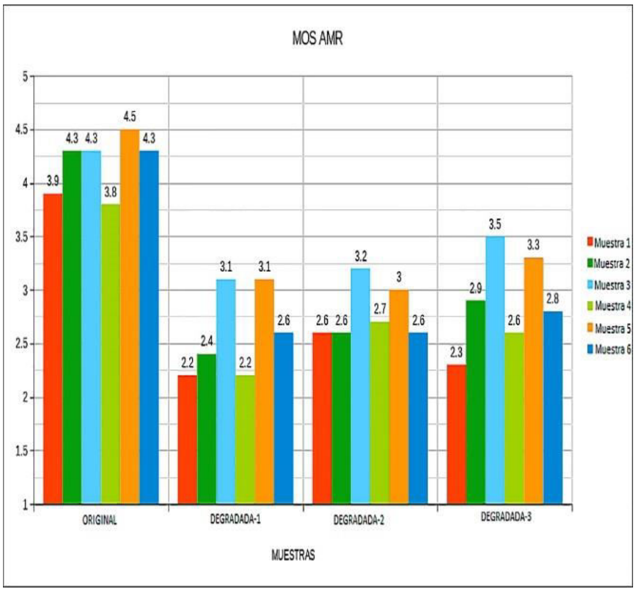
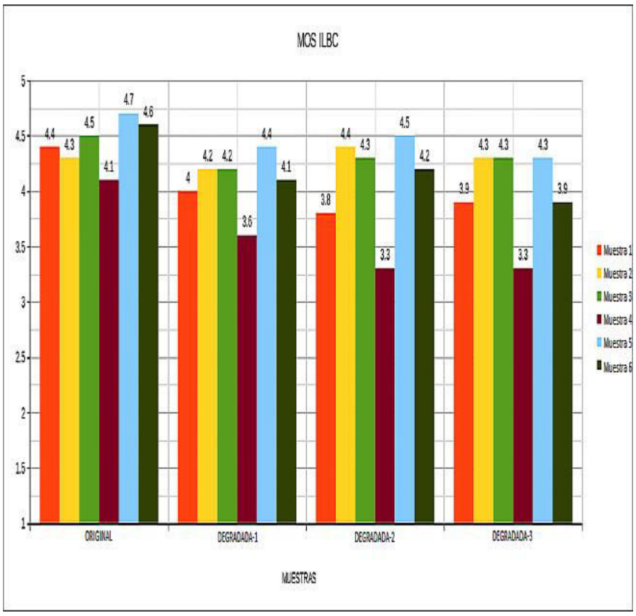


(b)



(c)

Figure-2. (a) MOS G.729, (b) MOS AMR, (c) MOS iLBC.

Figure-3 shows the averaged results of the PESQ objective evaluation. Red G.729 encoder, blue AMR and yellow iLBC. The horizontal asymptote is located from sample 1 to sample 6 and the vertical asymptote are the values from 1 to 5 which are the value of the MOS that throws the PESQ.

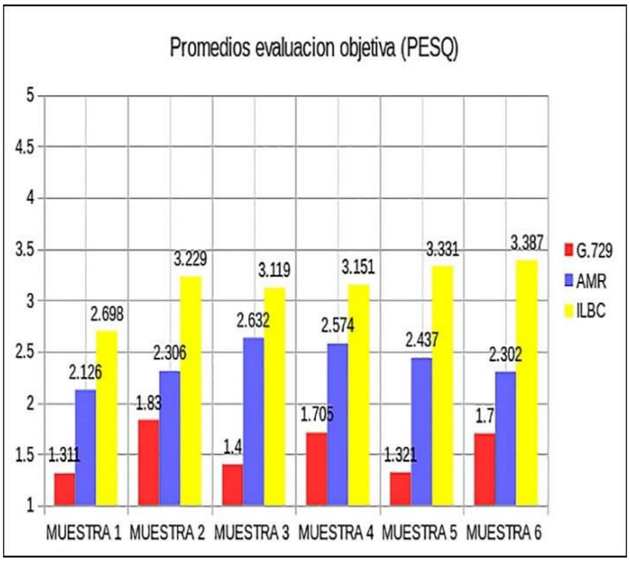


Figure-3. Average PESQ.

The percentage results show that the value that remains higher in each of the degraded samples is the iLBC (yellow), on the contrary the value that is always lower for each sample is the G.729 making it the most sensitive in the presence of losses.

Comparing the results of Figures 4 and 5 the minimum value that I give in the graph (a) the subjective test for the G.729 encoder was 1.4 and the objective test gave a minimum value of 1.311. The highest value for the subjective test was that of the iLBC coder with an average value of 4.5 and a little further away from the objective evaluation with 3.387.

4. CONCLUSIONS

The subjective and objective evaluations of voice quality show that the coder that best behaves in the presence of degradation is the iLBC, taking into account that it has a transmission rate higher than G.729 and AMR coders. It should be noted that the iLBC encoder has the same frame size as the AMR, although the iLBC does not have a lost packet mitigation module but has the ability to interpolate the previous and subsequent packet, replacing the lost packet, which makes the packet loss more robust. The results obtained show that the packet losses have little impact with the use of the iLBC encoder compared to the G.729 and AMR encoders.

The comparison of the results between the MOS and the PESQ shows a very close relationship between the values obtained for the G.729 and AMR coders, for example, the highest value of the subjective evaluation for the degraded samples in the G.729 coder was 1.9 while in the objective evaluation was 1.83. Likewise, the lowest value for the subjective evaluation was 1.4 and in the objective evaluation was 1.311, however, the values obtained for the iLBC coder differ despite the fact that both evaluations show how the encoder with the most sensitivity to loss is G.729 followed by the AMR, being the iLBC encoder the most robust of the three.

It should be considered that the people interviewed in the subjective evaluation have different auditory perceptions, which makes it necessary to carry out several evaluations in order to guarantee the reliability of the results.

The channel quality indicator (CQI) ensures an appropriate transmission, where increasing the value of the CQI improves the performance of the system, since the block error rate (BLER) in the corresponding channel does not exceed 10%.

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