ISSN 2303-4521
Original Research
Periodicals of Engineering and Natural Sciences
Vol. 10, No. 2, April 2022, pp.199-206

Adaptive speech compression based on AMBTC

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ABSTRACT

Most of the AMBTC-based RDH (absolute instantaneous block truncation) schemes Cannot be decrypted because AMBTC, which is unknown to most device. Also, some of the RDH methods based on AMBTC. But the load capacity obtained is low. For this purpose, in this work, a scalable RDH scheme based on AMBTC was introduced from the AMBTC zip code. In contrast to the decoder-based AMBTC-based RDH methods that are only able to achieve a constant payload for adjust the audio. Due to its advantages, sound pressure has attracted a great deal of attention in the last 20 years. The main developments concern transmission requirements and storage capacity. The need for high-quality audio data has been increased due to sudden improvements in computer manufacturers and technologies. Therefore, the developments include speech compression technologies, in which the two compression classes are lossless. This paper aims to review techniques for specification compression methods using AMBTC (a momentary absolute block truncation notation based) method, and to summarize their importance and uses.

Keywords: Compression of speech, Types of stress, compression of sound, AMBTC.

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1. Introduction

Speech considers a human's essential means to deliver information. Due to the rapid development in communication technology, speech data compression is a required field and challenging [1-3]. Simply, speech compression can be defined as transferring signals of human speech into compressed ones. Then, the output must be decoded to be as similar as possible to the original signal [3-5]. Reducing the data needed to present the original data is the main goal of any compression process. Consequently, increasing (decreasing) in data transfer (data storage). Additionally, reducing the number of bits used in representing the signal. In order for the quality to be acceptable, the data must be specified. Compression techniques are generally divided into two types of data, the first is non-lossy, and the second is lossy. The result of the non-lost classes is the same as the signal entered into the encoder [2, 6, 7]. Nevertheless, the type of loss occurs when the input signal is not matched with the retrieved signal [8]. In a lossless pattern, images and text doe satellite type and scientific purposes. In contrast, some kinds of images, video, and audio can be encrypted using data loss technology.

Generally speaking, in the losing type, the following are achieved: the retrieved signal is undoubtedly better, and the gain in compression ratio is more. In the input signal, the core of any compression process is De-repetition. The type of repetition in speech signals: temporal redundancy, statistical redundancy, cognitive redundancy, and perceptual irrelevance. To evaluate the lost audio, fidelity criteria are used for the subjective and objective types. The subjective criterion includes hearing audio before and after encoding. On the other hand, the objective criterion includes human monitoring of the quality by playing the signal and checking the quality.

The checking can be performed when the loss of the original signal is represented by a mathematical expression. With the sound loss, the most common categories of compression methods are illustrated in Figure 1 [4, 9]. In this work, we have used the instantaneous absolute mass notation method based on absolute mass truncation to obtain speech compression and using this technique to obtain the best speech quality.

2. Literature review

DWT technology is used to compress the signal, and by DCT, this compressed signal is compressed again. Then, DWT technology is used to decompress this signal. Using different wavelets filters, the performance of the speech signal is evaluated according to: root mean square error (NRMSE) and maximum signal-to-noise ratio (SNR) [10, 11, 7]. Jalal Karam and Rawat Saad discuss the effects of different pressure constraints and regimes in order to establish high compression ratio and good noise ratio to signal., the compression ratio, the level of atomization and parameters measured was used. Consequently, we can achieve high compression ratios with acceptable SNR. The resulting signal comparison is done with the SNR, the NRMSE, and the signal-to-noise ratio (PSNR) [12, 13]. Smita et al., have applied various speech compression techniques, where the signal is converted to a compressed or compressed form. Thus, the signal can be stored with less bandwidth. On the other hand, the reconstructed signals are compared using factors such as SNR, PSNR and NRMSE [14, 15].
Hatem Al-Alaidi and others. Introducing new signal-by-signal compression algorithm using DWT techniques. The demand for digital information has increased because of the growth of multimedia technology. The only way to overcome this situation is to compress the information signal by removing its excess. The pressure ratio can be easily changed with waves while other methods have fixed pressure ratios [16, 17].
Othman or Khalifa, Syring Habib Harding and Aisha Hassan description in order of importance of sound pressure. Recently, sound compression has been considered one of the most basic technologies [18-22].

3. Preliminaries

3.1. Sound compression methods with sound loss

This section highlights the most essential and common methods used in compressing audio data. As illustrated in Figure (1), the missing compression methods can be classified as follows: prediction-based, modular compression, frequency-based and quantization-based [23, 18, 24].

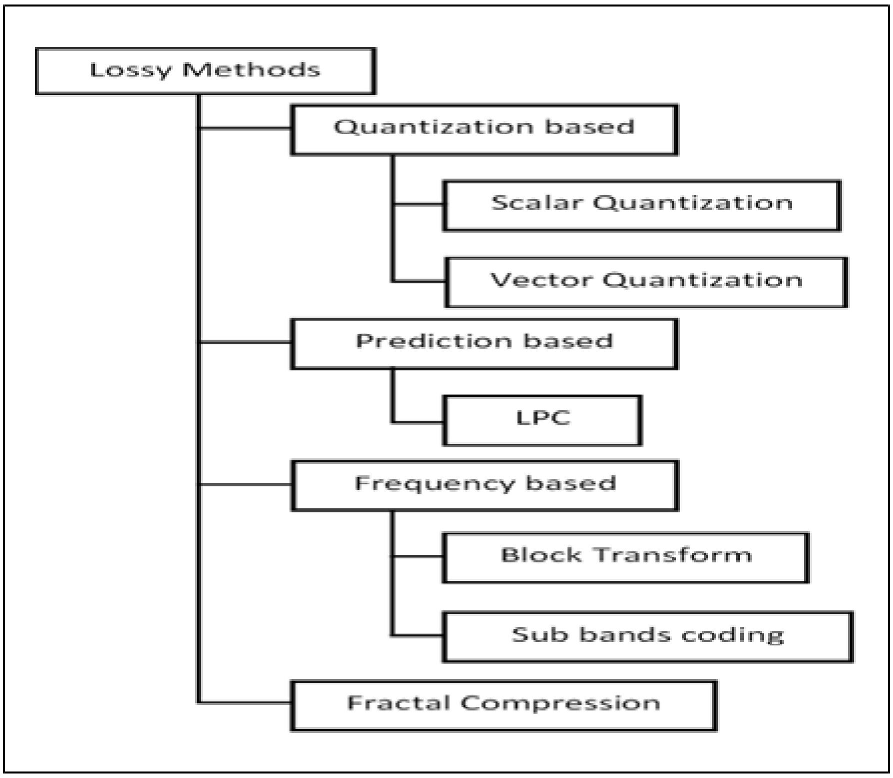

Figure 1. Most common type of speech compression with lossy sound

3.2. Speech compression techniques: Speech stress is categorized into the following: -

3.2.1. Coding of Waveform

An attempt was made to reproduce the signal sent as input at the output, which will be the same as the original signal [25-27].

3.2.2. Border coding

Signals are small parameters that describe the signals accurately. Whereas some features are extracted using a preprocessor that shall be used later to get the original signal [28].

3.2.3. Transcoding

In the transformation method, discrete wave transduction technology and discrete cosine transformation technique are used. Here, wavelet expansion is used to represent the original signal. Likewise, the DCT can be represented in terms of the DCT coefficients in this case of speech transformation. The signal is not compressed by switching technologies because they provide information about the signal and the use of different coding techniques.

The following steps are used to carry out the speech compression [2, 24].

1- Transfer Technology,
2- Threshold Transactions Transferred,
3- Quantification,
4- Encode.

3.3. Technique of AMBTC pressure

The AMBTC, is a lossless image aggregation processing method, proposed by W. Hong, in 2017 [29]. In image compression, this method is widely implemented for two reasons: Good image quality and simplicity. In the AMBTC, the cover image I of size $\psi\Gamma$ is divided non-overlapping pixel blocks of size $\{\mathrm{I_k}\}_{k=1}^{N}$, where ψ is the ambit for $\mathrm{I_k}$ and Γ is rising for $\mathrm{I_k}$, and N is set of all blocks. The average value of ai is computed in one block I_k, and so Γ_k is calculated as the mean pixel lifetime value to a_k. On the other hand, the average pixel value (the least quantization L_k) smaller than a_k. As the same size of I_k, B_k is formed by assigning '1' to the pixel that satisfies $\mathrm{I_{k,m}} \geq \mathrm{a_k}$ and set '0' to others in $\mathrm{I_k}$, where $\mathrm{I_{k,m}}$ stands for pixel j for $\mathrm{I_k}$ and $\mathrm{j} \in \{1,2,\ldots,m \times n\}$. This way, a block of size $m \times n$ can be pressure using AMBTC to a triple. Where (H_k, L_k, B_k); constructed by sequencing of H_k , L_k , and B_k. Likewise, using the sequence of binary representation of all $\{H_k, L_k, B_k\}_{k=1}^{N}$, the binary bit stream of audio I is formed.

Now, the compressed information flow in the decoding side, all 32 bits are retrieved as (H_k, L_k, B_k). The AMBTC R_k in B_k is compressed audio block is reconstructed by changing the pixels corresponding to '0' with L_k. Whereas the pixels corresponding to '1' is replaced with H_k.

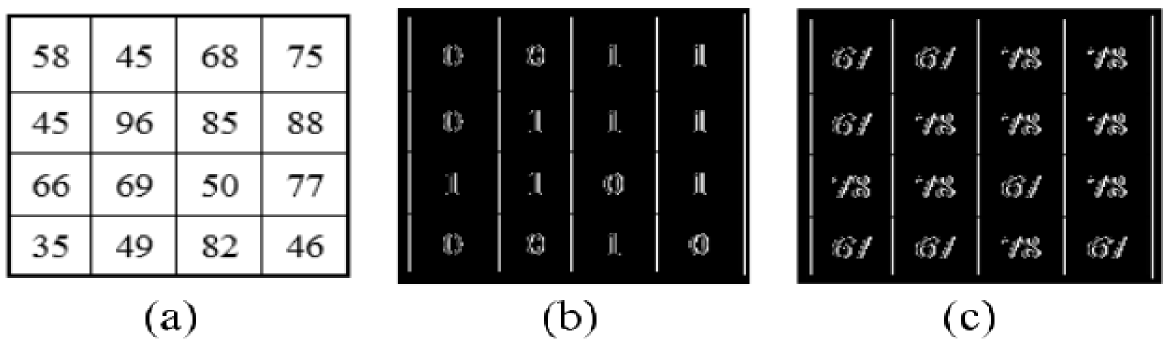

Figure 2. Example of AMBTC compression and decompression. (a) the original mass; (b) Raster audio; (c) A loose block.

First of all, in this block, a_k is determined as 64.625. Additionally, H_k is calculated as 78 according to the rule of AMBTC. Likewise, L_k is 67. As shown in Fig. 2 (b), for pixel $\mathrm{L_{k,m}}$, if $\mathrm{L_{k,m}} \geq \mathrm{H_k}$; $\mathrm{B_{k,m}}$ has been marked with 1; otherwise, $\mathrm{B_{k,m}}$; j is marked with 0. In the $\mathrm{B_k}$ bitmap, $\mathrm{B_{k,m}}$ denotes the kth bit. Therefore, we represent the mas as (78, 67, [0 0 1 1; 0 1 1 1; 1 1 0 1; 0 0 1 0]). When the data flow is being received at the receiver (on the decoding side), $\mathrm{R_k}$ in Fig. 1(c) is formed by setup $\mathrm{I_{k,m}}$ =67 when $\mathrm{B_{k,m}} = 0$ and $\mathrm{L_{k,m}} = 78$ as $\mathrm{B_{k,m}} = 1$.

3.4. LIN ET AL Schedule

Based on RDH in combination with (7, 4) Hamming's code, Lin et al. recently developed an AMBTC method [30]. The method depends on Lin et al. Same embedding method as "Chen et al." Where data were included on

the foundation of two groups-level scales. Besides, Hamming's symbol (7, 4) is used to achieve data merging of a single bitmap that corresponds to $\mathrm{H_k} \neq \mathrm{L_k}$. Specifically, the cover image, after AMBTC encoding, is split into two non-overlapping blocks of size 7 in $\{\mathrm{I_k}\}_{k=1}^{N_L}$ and each block is represented as a trinomial(H_k, L_k, B_k). N_L represents the number of 2 blocks with a size of 7. Like "Lin et al.'s" method. Lin et al's method incorporates. 16 bits in every B_k block with $\mathrm{H_k} = \mathrm{L_k}$, and 1 data in each B_k block with $\mathrm{H_k} \neq \mathrm{L_k}$ by switching $\mathrm{H_k}$ and $\mathrm{L_k}$ when the bit to include is "1" or by keeping $\mathrm{H_k}$ and $\mathrm{L_k}$ unchanged when the desired bit is Included "0". In contrast to "Lin et al." Use the Hamming -(7,4) symbol to establish a maximum of 4 bits to a single audio using $\mathrm{H_k} \neq \mathrm{L_k}$. For a single block of size 2×7 satisfying $\mathrm{H_k} \neq \mathrm{L_k}$, for protection reversibility, only one size 7 sub-block can be used in the single row of B_k for data embedding - percussion. Specifically, the sitemap is created to record and determine whether two embedding sub-blocks are utilized.
(for example, $\{\mathrm{B_{k,m}}\}_{k=8}^{14}$) is in line with 3-bit.

4. Proposed scheme

From above, despite Lin et al. higher payload is achieved by the method than Chen et al., it is not substantially better. The payload is not significantly improved. Additionally, to use Hamming's -(7, 4) code to embed data based on reversal assurance, Lin et al. demand use complex prediction mode and the judging mechanization via operations data merging and removal. We suggested a type of high-capacity type 1 of scheme AMBTC RDH.
In this work, we construct an individual simple map between unused and used bitmaps in the speech by analyzing the properties of bitmaps. Using this one-to-many map, each audio corresponding to $\mathrm{H_k} \neq \mathrm{L_k}$ able to carry multiple bits, for example, $\log_2 54$ bits of a single audio corresponding to $\mathrm{H_k} \neq \mathrm{L_k}$ in Lena. Additionally, a pre-threshold is introduced in the proposed method to promote the root execution more.
The techniques used in the proposed method are as follows:

4.1. The map of $(2K + 1)$

The audio I with $\mathrm{Q} \times \mathrm{Z}$ is broken into $\{\mathrm{I_k}\}_{k=1}^{N}$ blocks of $m \times n$, and $N = [\,\mathrm{Q} \times \mathrm{Z}\,/(m \times n)]$. So AMBTC is encrypted, every $\mathrm{I_k}$ block is pressure to a triple (H_k, L_k, B_k), where B_k is the same size as $\mathrm{I_k}$, and therefore there is a total number of N audio. Effectively, $\{\mathrm{B_k}\}_{k=1}^{N}$ audio can present rich abundance. Finally, we aim to apply redundancy between the same audio to hide confidential information. We design the RDH method by utilizing iterations on the basis of achieving high load capacity to maintain compression rate. However, for AMBTC ciphers, the sum of a zero-array of size $m \times n$, is not was born. Hence, we obtained B_k by compacting AMBTC has only $2^{m \times n} - 1$ states. All audio interview for the same state are categorized in a category, also called the used category. Assume all categories is p. The category named $\mathrm{T_r}$. taking in to account that $\mathrm{p} \leq \mathrm{K}$ and $K \ll 2^{m \times n} - 1$, so $p \ll 2^{m \times n} - 1 - p$ of $2^{m \times n} - 1 - p$, not equals to $\{\mathrm{B_k}\}_{k=1}^{N}$audio, and the unused state of t-th is named $\mathrm{z_r}$, where $t \in \{1,2, \ldots, p\}$. The data is included by constructing a one-to-many map between the used p layers and the unused $2^{m \times n} - 1 - p$ due to $2^{m \times n} - 1 - p$ layers. Also, to ensure inversion is possible, the first sound for it is not possible to modify any category when merging data. Thus, the p-classes used, if it only contains audio, it must be excluded from including the data as it cannot provide any burden on it. For the special case, it corresponds to the set of every "1", that is, all $m \times n$ elements in $\mathrm{B_k}$ be "1", and every $\mathrm{B_k}$ can supply a 32-bit, since it is also excluded from to several maps. Except for this special case and all q layers, the residual $p - |q| - 1$ layers are chosen as a p_{-q} group, and $|.|$ is used to indicate the size of the group. Eq (1) it is used to specify the number of unused instances to which one user class is assigned. That is, every class of $p - |q| - 1$ can hold $\log_2$ 2k bits.

$$\gamma = \left\lfloor \frac{2^{m \times n} - 1 - p}{p - |q| - 1} \right\rfloor \qquad (1)$$

So, z_1 to z_k are mapped to T_1, the unused states from z_{2k+1} to z_{2k} are mapped to T_2, and so on, until each p_{-q} layer compatible. Thus every layer of $p - |q| - 1$ hold $\log_2$ (2k + 1) / bit, rather than $\log_2$ 2k.

4.2. Example of an individual map $(2\mathrm{k} + 1)$

Taking the size of Lena 256×256 for instance, if $m \times n$ is set 3×3, then exist 65636 sounds after AMBTC encode, and all these sounds can be categorized into 32818 used categories. In the experiments, we know that $q = 30267$, which means that 6083 out of the 32818 classes used only contain audio. Correspondingly, each of the 2550 classes (i.e. $p - |q| - 1 = 32818 - 30267 - 1$) It can contain more than one sound. In the above, the audio has a total of $2^{16}-1$ states, 32816 of which are not used cases. In our work, the blocks containing $H_k = L_k$ need they are treated separately. Block number with $H_k = L_k$ it is $\mathrm{N_e}$ and $\mathrm{N_e} = 14$. The sound to every layer in p_{-q} not utilized to include the data cause of the possibility of inversion and every layer of q is not found from the data inclusion, hence the total of p audio is not included in the inclusion data. At last, $N - \mathrm{x} - \mathrm{p} + 1 = 65636 - 14 - 32818 + 1 = 32805$ sound is used to map a single to $(2\mathrm{k}+1)$.

4.3. Embedding stage

The process of integrating data it has two main section. The first section is preprocessing, and the second section is data collection: as follows:

1- Pre-process: The main objective of the introductory process is to be prepared to include the following data. It is to create a sitemap in order to ensure that it is possible to roll back. The second is to get the exchange of $\mathrm{H_k}$ and $\mathrm{L_k}$ when the following two conditions are satisfied with Hi and Li: $\mathrm{H_k} = \mathrm{L_k}$, and $\mathrm{B_k}$ is sound of every category in $\mathrm{q} \cup \mathrm{p_{-q}}$. We will create a sitemap of size p, named the symbol A. Especially, in sitemap A, we allow a '1' to every layer and a '0' to every audio in $\mathrm{p_{-q}}$. So we can extract the data and restore the audio. Every zip of codes $\{(H_k, L_k, B_k)\}_{i=1}^{2N}$ is studying in sequence. per block $(\mathrm{H_k}, \mathrm{L_k}, \mathrm{B_k})$, if $\mathrm{H_k} = \mathrm{L_k}$ and $\mathrm{L_k}$ are kept unchanged, for one of the first blocks of $(\mathrm{H_k}, \mathrm{L_k}, \mathrm{B_k})$, if $\mathrm{H_k} \neq \mathrm{L_k}$ and $\mathrm{B_k}$ is audio of $\mathrm{q} \cup \mathrm{p_{-q}}$, and Hi incremented by 1, $\mathrm{H_k}$ and $\mathrm{L_k}$ are swapped.
2- For one of the remaining N blocks of p, if $\mathrm{H_k} \neq \mathrm{L_k}$ and $\mathrm{B_k}$ are the first audio of every class in $\mathrm{q} \cup \mathrm{p_{-q}}$ so $\mathrm{H}_k^/ = \mathrm{L_k}$ are $\mathrm{L}_k^/ = \mathrm{H_k}$; Otherwise, $\mathrm{H_k} = \mathrm{H}_k^/$ and $\mathrm{L}_k^/ = \mathrm{L_k}$. Specifically, $S^/ = \{Q, S\}$.

3- Collection of Data
After pre-processing, for the block with $\mathrm{H}_k^/ = \mathrm{L}_k^/$, the data is 16 bits which are extracted from $S^/$ can be invisible in a $\mathrm{B_k}$. For a block with $\mathrm{H}_k^/ \neq \mathrm{L}_k^/$, the data inclusion is processed to the status: $\mathrm{H}_k^/ < \mathrm{L}_k^/$ and $\mathrm{H}_k^/ > \mathrm{L}_k^/$. To $\left(\mathrm{H}_k^/, \mathrm{L}_k^/, \mathrm{B_k}\right)$, if $\mathrm{H}_k^/ < \mathrm{L}_k^/$ i and $\mathrm{B_k}$ is audio for every class in $\mathrm{q} \cup \mathrm{p_{-q}}$ by compressing each bit of $\mathrm{B_k}$. In this method, blocks containing $\mathrm{H}_k^/ < \mathrm{L}_k^/$ can be decrypted directly at the end of the decoding. When $\mathrm{H}_k^/ > \mathrm{L}_k^/$ and $\mathrm{B_k}$ is not audio of every category in $\mathrm{q} \cup \mathrm{p_{-q}}$, so by change $\mathrm{B_k}$ with one $\left\{z_{(k-1)\times 2k+1}, z_{(k-1)\times 2k+2,\ldots,z_{k^2}}\right\}$ So as to achieve data integration.

Now, we will give the algorithm for our work as follows:
Input: AMBTC compressed $\{(H_k, L_k, B_k)\}_{i=1}^{2N}$, with data $S^/$.
Output: codes of AMBTC $\{(H_k, L_k, B_k)\}_{i=1}^{2N}$
Step 1: Read the compression code $\{(H_k, L_k, B_k)\}_{i=1}^{2N}$as a sequence.
Step 2: For $\{(H_k, L_k, B_k)\}_{i=1}^{2N}$, if $\mathrm{H}_k^/ = \mathrm{L}_k^/$,32 bits which are extracted from S' and B_k' are shaped.
Step 3: For (H_k', L_k', B_k), if $H_k' < L_k'$, then it is recorded in the one-to- $(2k+1)$ map.
Step 4: For (H_k', L_k', B_k), if $H_k' > L_k'$, Then the following three cases of embedding are carried out $\log_2{(2k+1)}$ bits in anointed trio:
Case 1: Modify S' to $(2k+1)-ary$, k is determined in Eq (1.1).
Case 2: Assume $B_k \in T_r$. So, $T_r \left\{z_{(k-1)\times 2k+1}, z_{(k-1)\times 2k+2,\ldots,z_{k^2}}\right\}$ construct assignment group of B_k.
Case 3: If B_k corresponds to itself, if the bit to include is 0; if $s = 1$, then B_k is mapped to $z_{(r-1)\times 2k+1}$, and so on; if $s = 2k+1$, then B_k is mapped to z_{k^2}. Assume that $B_k \in T_1$ and $s = 30$, and we replace B_k with z_{30}. Hence, the code of AMBTC is (H_i', L_i', z_{30}).
Step 5: Iterate steps 1 through 4 until all parts of confidential data are included.
If the bit to include is 0, B_k corresponds to itself
Step 6: After data secretion whole, Stego codes are ejected.

taking $H_k' = L_k'$ is N_e, then $C_1 = 32N_e$.If $H_k' > L_k'$ is N_v, then $C_2 = \log_2{(2k+1)} \times N_v$ represents this part's capacity. Additionally, p denotes the length of Q. Thus, $C = C_1 + C_2 - p = 32N_e + \log_2(2k+1) \times N_v - p$ It is equal to the number "1" in A.

5. Experimental results

As mentioned erlier, increasing the data transfer and reducing the required storage are the main goals of the compression process. To achieve that, compression methods have been proposed to implement [29, 30]. Despite of achieving good results by the so fare existed methods, the requirements of all types of audio files can not be met. Several factors are needed to be considered. For example, when the signal is high-link, DCT is implemented while using DWT is better because of the property of localization across the time-frequency domain [22, 31]. On the source filter model the AMBTC was built. Additionally, although AMBTC is utilized in the compression process, the outcoms are improved when implement as a material extraction tool with better outcomes. It can be determined by implementing the fractal method. Nevertheless, the delay in rebuilding the audio file is the main con of this process. As there are no same samples adopted in the existed researches, it is difficult to make a fair comparison among them. In other words, each paper use different samples and not standardize.
Here, we include the outcoms of the some existed emthods and the same sample. To demonstrate our technology used in this work, we used sound test sample in the researches [17, 26, 30]. Table 1 lists the selected sample characteristics.

Table 1. Selected sample characteristics

Audio Type	Dialog
Size (KB)	256
Sampling resolution (bps)	32
Sampling rate (KHz)	52400

In AMBTC format, the sample is a compressed wave file format. The wave pattern of the selected sample is illustrated in Figure 3.

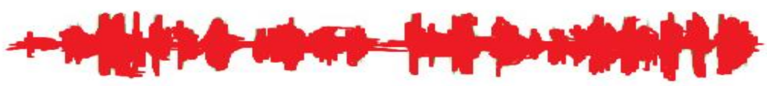

Figure 3. The waveform of the selected sample

According to Table 2, the compression technique in [26] got better outcomes for the selected sample. Compressed speech can be sent in wireless medium by modern microstrip filters and antennas reliably [31-34].

Table 2. Comparable of AMBTC, CR, and PSNR using two methods.

	First method	**Second Method**
AMBTC	11.54 1	12.79 3
CR	17.29 3	13.10 6
PSNR	23.52 7	16.85 5

6. Conclusions

In this work, general techniques for sound compression by the AMBTC method with sound loss are presented. The number of compression techniques has been increased. It is challenging to balance the quality of the reconstructed signal with compression gain. To implement a good compression system, several techniques should be combined. To achieve that, hybrid methods have been used in the literature. The method used was better in terms of quality than the CR and PSNR. The experiments showed that the file's characteristics affect the compression results. Therefore, it is hard to implement a specific method for all audio files.

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