A Critical Assessment of Advanced Coding Standards for Lossless Audio Compression

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Abstract - Image data, text, video, and audio data all require compression for storage issues and real-time access via computer networks. Audio data cannot use compression technique for generic data. The use of algorithms leads to poor sound quality, small compression ratios and algorithms are not designed for real-time access. Lossless audio compression has achieved observation as a research topic and business field of the importance of the need to store data with excellent condition and larger storage charges. This article will discuss and analyze the various lossless and standardized audio coding algorithms that concern about LPC definitely due to its reputation and resistance to compression that is audio. However, another expectation plans are likewise broke down for relative materials. Comprehension of LPC improvements, for example, LSP deterioration procedures is additionally examined in this paper.

Keywords - component; Audio; Lossless; Compression; coding.

I. INTRODUCTION

Compression is to shrink / compress the size. Data compression is a technique to minimize the data so that files can be obtained with a size smaller than the original file size.Compression is needed to minimize the data storage (because the data size is smaller than the original), accelerate information transmission, and limit bandwidth prerequisites. There are two sorts of sight and sound information pressure i.e. Lossy Compression and Lossless Compression [1]. Whereas Lossy compression produces smaller compression files compared to existing lossless methods. Lossless compression is used if the accuracy of sanitary data is important, while Lossy compression usually removes parts of the data that are not very useful and not perceived so forethought argues that the data can still be used even in compression. The second illustration of the compression approach is shown in Figure 1 and Figure 2 [2].

Lossy compression is a method to compress data and decompress it. The data obtained may be different from the original data, but the difference is quite similar. This method is most commonly used to compress multimedia data (audio files and images). Lossy compression format encounter generation loss i.e. experiencing prose compression-decompression repeatedly will cause progressive loss of the quality.

Lossless compression is a data compression method that allows the original data to be rearranged from compressed data so that the compression ratio cannot be too large to ensure all data can be restored to its original form. The lossless method produces data that is identical to the original data. Lossless compression is primarily used for archiving and editing. For archiving purposes such as bank records, text articles, medical, assets and all interests with data that is unchanged or different from the original.

The theory of Source Encoding is one of the three fundamental theorems of information theory introduced by Shannon (1948). The theory of Source Encoding declared a fundamental limit of a size where the output of sources of
information can be compressed without causing a huge error probability. We already know that the entropy of a source of information is a measurement of the information content of a resource. Thus, it can be concluded that, from the theory, the entropy of a source is very important\[3\][4][5].

\[
H(X) = - \sum_{i=1}^{m} P_i \log_2 P_i
\]  

(1)

The source is determined by the efficiency of \( H(X)/H(X) \) max, where \( P_i \) is the probability of symbol to-I, and \( H(X) \) maximum when sources have the same probability of symbol \[6\][7][8][9].

Where \( P_i \) is the probability of occurrence of the i-th symbol of the alphabet while redundancy is the difference between the entropy of the maximum data theory and has the proper entropy. then the calculation as follows:

\[
R = \left[ - \sum_{i=1}^{n} P_i \log_2 P_i \right] - \left[ - \sum_{i=1}^{n} P_i \log_2 P_i \right]
= \log_2 n + \sum_{i=1}^{n} P_i \log_2 P_i
\]  

(2)

Where \( P \) represents the most elevated entropy symbol ratio (\( P = 1/n \)).

There are many models of mechanisms in lossless audio compression preceded and improved; there are many popular standards and non-standard compressors that use such as Linear Prediction Coding essentially because of the various conditions that will be described in the next section. The fundamental process of most prominent lossless compression algorithms calculations (Encoder) is uncovered Figure 3.

The intra-channel decorrelation stage includes the Prediction model, covering the quantization regulation for each channel of the audio signal [10]. This prediction design is the missing frame of the whole operation because there is constantly the possibility of prediction errors. In any case, by encoding the Predictor Model residual error, Entropy Coding Block remunerates a segment with Lossy properties to be lossless over. There are several terms to say what this mechanism is like, in which the Entropy Coding Block set in add a stored error then encoded, it is an advanced effect of Predictor Model to establish a sound compression scheme with lossless concept.

The Lossy decoder in the case of lossless compressed audio is as an indicator configuration that aims to recreate the original signal as identically and accurately as possible so as to reduce the error of the residual errors. Whenever the result gets a little error, then the extra optimal the entropy coding and the shorter the compressed binary data, the effect of the short of an error residue [11]. The basic indicator arrangement with entropy coding is explained in Figure 4.

In terms of introducing MPEG-4 ALS, a proceeding with expectation coding plan was purposed as its principle prescient model. This is the different frequently-utilized standard for lossless audio compression. The possibility of direct encoding expectations can be depicted that the discourse illustration, \( x(n) \) can be approximated as a continuing compound of the previous example, \( x(n - k) \) [11]. Equation (3) represents the sample prediction, \( \hat{x}(n) \).

\[
\hat{x}(n) = \sum_{k=1}^{K} a_k x(n - k)
\]  

(3)

The limitation of the design is that it needs the predictor coefficients to be encoded along including the outflow flow of the bits, costs in the pressure proportion of the sound information data files, the accuracy of the indicator coefficient relies upon the amount. On the off chance, it can be seen that the bigger the request, the more the costs will happen.

\[
e(n) = x(n) - \hat{x}(n)
\]  

(4)

For sound analysis there is an LPC (Linear Predictive Coding) representation which is difficult to do like Log Area Ratios (LAR), this is separated from the coefficients of one part of LPC (ak). Relatively, LSP disintegration ensures on indicator strength, and spectral error is placed nearby to minimize the coefficient deviations. LSP decomposition will be located to prescient channels are complete of symmetric and non-symmetric polynomials as connected as a result of the term IEEE 1857.2 and IEEE 1857.3. It is shown in Eq. (5), whither? \( f_1(z) \) is a symmetric polynomial and \( f_2(z) \) is the anti-symmetric polynomial of the quantized vector [12][13].

\[
A(z) = \frac{f_1(z) + f_2(z)}{2}
\]  

(5)

The discussed review in this paper provides a methodical coverage of fundamentals from lossless audio compression. The next point will be explained regarding lossless audio
compression, and then proceed with discussions of different utilization of each standard and algorithm. The next section will be specially divided into subsections, the first outline is the various lossless MPEG-4 rules, both on the Free Lossless Audio Codec (FLAC) and all three will use the standard IEEE 1857.2 and IEEE 1857.3 which be released in 2013 and 2014. In the fourth section will discuss the various methods enhanced as suggested by various researchers based on models and algorithms.

II. STANDARDS OF LOSSLESS CODING FOR AUDIO

A. MPEG-4 STANDARD

One of the lossy compressors is AAC (Advanced Audio Coding), from this concept is termed Lossy to lossless after the MPEG-4 compressor as a tool for the compressor sound. To improve the standard of MPEG-4 Audio Lossless has done some research, which is ALS LPC (Audio Lossless Coding Linear Predictive Coding, RLS-LMS (Audio Recursive Least Mean Square-Least Mean Square); SLS (Scalable Lossless Coding) [14] which will be explained as follows.

A1. MPEG-4 ALS

Beginning in 2003, MPEG-4 Audio Lossless Coding (ALS) has established a coding structure and continues to be developed until 2009, and the achievable result that MPEG-4 ALS with compression ratio and process speed can be assumed higher than Monkey's Audio when its declaration in 2003 [15]. There are five parts of the Encoder process of MPEG-4 ALS, as illustrated in Fig. 5. Elements contained from the depiction described previously:

b. The coefficient of Estimate and Quantization, and Predictor blocks: Intrachannel Decorrelation.
d. Multiplexer: Generates a compressed signal beside with the required header.

For reestablish the bit stream result from the Encoder, the relating Decoder bearings comprise of a demultiplexer, which will disintegrate the decoded flag and re-encode the remaining and the files code, as adequately as the quantized coefficients.

In the last standardized variation in 2009, the MPEG-4 ALS compression proportion is superior to FLAC yet slower in disentangling speed[11]. In the finishing up adaptation, the indicators are subdivided into two blocks, short-term predictor blocks, long-term predictor blocks and adaptive predictive request instruments to find the ideal indicator technique. Long-term indicator block and versatile prescient summons will additionally decrease lingering mistakes by watching connections among different edge remaining error, but compensating for encoding speed, due to the addition of complexity. Equation (6) presents the check to decide the long-term residual fault, where \( y \) is the additional and \( v \) is the slack.

\[
\varepsilon(n) = e(n) - \sum_{j=-2}^{2} y_{r+j} \cdot e(n - r + j)
\]  

A2. RLS (Recursive Least Mean Square)-LMS (Least Mean Square) mode

RLS-LMS is a developed thought of MPEG-4 ALS. As a choice to the LPC display, different coefficients are multiplexed in the encoded signal. The LMS is considered for its low complexity. In any case, LMS has feeble conjunction traits without anyone else's input that causes bad gain predictions, so the RLS display is incorporated to counterbalance property. RLS is not feasible due to its great complexity. To achieve a scales for both traits, it is possible to generate a better predictor of this scheme [16].

The concept of LPC's idea that the RLS-LMS technique does not require an indicator coefficient to be encoded on a similar progression appears in Figure 6 and Figure 7 in light of the fact that the decoder contains a similar indicator channel as an encoder. In any case, this creates a high estimate of RLS calculations. The LMS channel length indicates that it is possible that the RLS-LMS is quieter than LPC ALS mode [18]. Despite the fact that the pressure proportion is roughly 1% higher [16] [17] for customary music documents (tests at 44.100 Hz). Be that as it may, RLS-LMS mirrors a superior pressure proportion for more ideal determination sound (examples at 9.600 Hz)[18].
Although the results of this compression ratio are impressive, the variation of the deviation raises for correlations of a signal to signal with the wide variation that leads to debased execution of the RLS calculation for clean signal or signal with shorter variety [16]. And additionally the greater plan LMS it makes the decoder be included complex and slower concerning unraveling and encoding speeds [11].

\[ \hat{x}(n) = \sum_{k=1}^{K} c_k y_k(n) \]  

(7)

\[ y_k(n) = \sum_{m=1}^{M_k} a_{k,m} e_{k-1}(n-m) \]  

(8)

\[ e_k(n) = e_{k-1}(n) - y_k(n) \]  

(9)

From Equation (7)-(9), the RLS-LMS filter is different in calculations because the residual fault is assumed from the variance in residual prediction and the output of the previous k-th step. The concluding prediction, \( \hat{x}(n) \), is a well-regarded result prediction of a linearly added level [18].

A3. MPEG-4 Scalable Lossless Coding (SLS)

Not comparative with the LPC idea, this outline may want to utilize a Lossy MPEG-4 AAC (Advanced Audio Coder) development, which gives excellent perceptual coding utilizing Modified Discrete Cosine Transform (MDCT) [14]. The continuation of a lossless SLS, investigating sound signals within IntMDCT blocks (Integer Modified Discrete Cosine Transform) [19]. Essentially, are evacuated with a control the Adaptive LPC outline. Residual errors can likewise be avoided from the SLS outline, and the yield quantized signal is decoded by an arithmetic coding plot, Bit-Plane Golomb Coding (BPGC) or Context Arithmetic Coding (CBAC) [14].

In view of its versatility of MPEG-4 AAC, this approach is adjusted into the MPEG-4 Standard Audio device [20]. Considerably unique in relation to the LPC display, this strategy changes over the sound signal to the recurrence zone by IntMDCT, which is encoded with a two-layer building, as given in Figure 8. AAC core layer and lossless layers that also develop spectral forms of quantization of AAC bit core noise at intermediate rates. This increases noise-to-mask ratio, enabling optimum quality regarding perceptual quality at this bit-rate operation.

1. FLAC

FLAC endures for Free Lossless Coding Audio, which started in early 2000. Codecs are still standing immensely by developers and researchers across the times. To compress audio, FLAC applies the four-stage process given in Figure 9 [21].

The Blocking step isolates the sound signal into portions or segments of the detailed size. This right away changes the compression ratio, if the chunk is too short, the quantity of segments will increment, disposing of the bits in the encoding header, and the other way around. The inter-channel decorrelation (mid-side transformation) arrangement is accomplished by evacuating repetition in left and right channels (stereo). The expected bits to gather the flag can be lessened, with encoding the stereo into the center (left and right normal) and the side channel (left-right short). In the event that the stereo fluctuate broadly, they can be skipped with no embellishment. The prediction level is very important to provide good and changeable compression of chunks to chunk audio samples [21].

The improvement of this plan is the exchangeable prescient plan of a few of the sound section and sub-pieces to orchestrate the involution of the arithmetic computations as indicated by the structure of the present signal which enables faster-decoding speed. Be that as it may, additional overhead is required where the weight extent will be lower.

Estimation of FLAC with MPEG-ALS demonstrates that its versatile properties where pressure proportion, encoding and translating speed increment WAV document measure similarly. By deactivating MD5, it can accomplish said that FLAC is astounding in regards to compression ratio and time process which compared to all different lossless audio codecs also it endures up to 25% of improvement in speed due to features calculation of MD5 is checked [22].
III. STANDARD OF IEEE 1857.2 AND IEEE 1857.3

In 2013, final parameters for the IEEE 1857.2 Advanced standard Audio Coding implied proposed. In 2014, IEEE 1857.3 indexed to audio and video systems. The standard is a combine of straight prediction and the innovative entropy algorithm [12][13]. This Segment describes some of the fundamental elements of this parameter rule.

A. General Block Diagram

Figure 10 declares to the standard IEEE 1857.2 lossless sound encoder or Decoder plot. Not at all like the standard LPC display, the parameter IEEE 1857.2 uses Integer Wavelet Transform to establish the relating highlight parts and remake this component, and the pre-process level in which the signal straightened and thusly encoded by the bury lossless bit stream entropy coder. The de-flattened (post-processor) turned around process is excessively itemized, making it impossible to reestablish the decoded motion in Lossless route as a correct replication of its unique shape. There is something essential to recall, and it is important that the system uses fixed points rather of floating purposes that contribute to fast speed decoding.

B. Predictor IEEE 1857.2

Figure 11 depicts the different components of the indicator plot. From the Integer Lifting, wavelet Transform segment, only the high-recurrence sound information portrayal outline is embedded inside the indicator module, where Linear Predictive Coding (LPC) is then directed on each edge, by means of LSP deterioration [3]. The halfway affiliation coefficient (PARCOR) is ascertained by the calculation Levinson-Durbin algorithm [24]. The PARCOR coefficients are quantized and transmitted in a lossless bitstream. Likewise, the quantized PARCOR coefficients are excessively connected for bounded de-quantized and converted to LPC coefficients that assemble predictive returns for every sample in the block (c'd [n]), thus authorizing the computation of residual errors (e [n]) by reducing the sample remaining predicted with input signal (cd [n]).

\[
\text{quantizedPARCOR} = \left\{ \begin{array}{ll}
\frac{1}{2} & k = 1 \\
\frac{1}{2} + \frac{\text{PARCORcoef}_{k}}{2} & k = 2, \ldots, \text{LPC order}
\end{array} \right.
\]

While still in the encoder process, it is necessary to de-quantize the PARCOR coefficients before they are set as decoders in the encoder process to generate signal predictions. The quantized PARCOR coefficients are excessively connected for bounded de-quantized and converted to LPC coefficients that assemble predictive returns for every sample in the block (c'd [n]).

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\end{array} \right.
\]

Equation (12) is formulated to produce a matrix prediction replaced by an audio frame to obtain residual predictions for the coding process [24].

\[
\gamma(i) = \left\{ \begin{array}{ll}
l_{\text{PARCOR order}} & 1 \leq i \leq \text{PARCOR order} - 1 \\
l_{\text{PARCOR order}}(1 - \gamma(i)) & i = \text{PARCOR order}
\end{array} \right.
\]

frames(i), i = 0, ..., number of frames
residues(i) = frame(i) - \gamma(i), i = 1, ..., number of frames

Figure 12. Basic Block Structure of IEEE 1857.2 [23]
IV. ENHANCED MECHANISMS

This section will evoke research on predefined standards or algorithms then generate innovative purposes for the design of predictors.

A. Sparse Linear Predictor

The smallest classification approach with long range and approached by greedy algorithm, this will help overcome the problems in the smallest partial squares used in the LPC model. The main purpose of this rule is to increase the decoding process time and compression ratio when compared with non-sparse design consisting of audio encoder and lossless audio decoder [25].

Audio Compression Equipment is a development and improvement of the popular lossless audio computing available. OptimFrog was created by Florin Ghodi from one of the article writers in 1996 [26]. the specified audio channel or second channel will be set to reduce errors in the main channel. This design equation is explained following, where $x_i$ and $y_i$ are the main and second channels, $\psi^W$, whereas $a_i$ and $b_i$ describe the regression coefficients, $W$.

$$x = \sum_{j=0}^{\psi-1} a_j x_{i+j} + \sum_{j=0}^{\psi-1} b_j y_{i+j} = \psi^W$$  \hspace{1cm} (13)

MPEG-4 ALS (RLS-LMS) utilizes LPC as a compression setting, producing process speed in decoding to excellent [25], but the result of the structure rule of the scheme shows that LPC can raise the compression ratio more optimally than the MPEG-4 ALS (RLS-LMS) algorithm, but lower than OptimFrog.

B. Gradual OLS-NLMS with rules from MPEG-4 (RLS-LMS)

Based on the problem of numerical establishment of the RLS algorithm in MPEG-4 RLS-LMS, the replacement of the Ordinary Least Square (OLS) algorithm can reduce the difficulty and utilize higher computational complexity over improving the LMS filter level, and it uses the Normal LMS to withdraw quiet concentration [27]. Although previous analyses have shown that previous research has shown higher LMS order can degrade the achievement of decoders [11].

Figure 13 shows a filter of OLS-NLMS, the objective being to eliminate the remainder of $u(n)$ from OLS filters and RLS filters, as well as the Cmix Cascading filter coefficients which are parameters for stochastic transition for three stages NLMS Filter [27].

C. Improved MPEG-4 SLS

Formerly, the standard from MPEG-4 applies Laplacian allocated input data for BPGC calculations. This improved scheme for estimating input data on general Gaussian distributions as estimates from BPGC. the result to be achieved from this improvement is a more optimal compression ratio with computational complexity that is still related to the original method, gap from an increase of at least 0.01% of the original method [28].

D. Lossless Audio Coding used Code Excited Linear Predictor (CELP)

To analyze another predictive rule besides the LPC, CELP was introduced, without applying a lossless arithmetic de-correlator to decrease the multifaceted of arithmetic calculations [29]. Code Excited Linear Predictor (CELP) uses a wide assembly of stochastic codebooks to define 'linear prediction coefficients and equals LPC predictors if any residual faults are coded with coefficients. In spite of the fact that compression ratio is commonly high and cannot be assumed whether the decoding and encoding rates are better than the algorithm of MPEG-4 ALS (RLS-LMS).

V. DISCUSSION

Comparisons for all lossless compression sounds with their respective strengths and weaknesses in Table 1, can be compared that:

1. FLAC has good performance against large WAV file data and many people know it to be famous because it is open source.

2. Another case with MPEG-4, although not popular, but has faster performance on the encoding process and decoding speed.

3. However, IEEE 1857.2 and IEEE 1857.3 are good comparisons as for the development has good performance and can compete with other compression algorithm or compression schemes.

It can be accepted from previous tests as in table 1 that lossless audio compression takes into consideration high compression proportion comes about, yet additionally quick handling time coding and decoding speed. Despite the fact
that the speed of the encoding procedure does not significantly affect the client, but rather the decoding speed related with the playback show for capacity and transmission.

A. Compressor Lossless Audio.

The topic of lossless audio compression is one of the areas of rapidly evolving research in today's big data era. At first there were just some lossless audio compressors in the mid 90's. At this time already there are more than 30 kinds of utilities that can be downloaded from the internet. Growing and expanding rapidly the number of lossless compressors is due to the real need that lossless audio compression has become a common need and interest for practitioners and industry as the effect of high broadband development, strong CPUs, and low storage costs. Lossless audio analysis and computation has gone through many developments from both techniques and algorithms and continues to be developed to produce a new standard for audio encoding. In Table II, it provides comparison of various lossless audio codes with condition compression ratios, encoding and decoding speeds and coding categories.

B. Utilization of Coding Compression Lossless Audio

Along with the development and the need for data compression, the use of all things about lossless audio compression can also be implemented in various purposes and areas of science, not only used for music but also in medical or biological field since the basic mode scheme relates to the study of data processing machines, thus requiring extra space for data storage. These conditions and circumstances are not only specific to audio data but may also be used in other respects in the compression field such as video compression. In addition, further research is being developed to improve the concept of lossless audio compression based on LPC to optimize the speed of encoding and decoding in general for multimedia, because there is a high demand for FLAC codecs in the business world and technology as software continues to be released continuously until now [21].

Table 3 gives the different utilization of each algorithm according to every standard and specification of the algorithm as well as the present method using every algorithm. In common, every standard and algorithm in the review in this paper is presented with the purpose of data storage, its application to medical and transmission is of critical interest to the authors and researchers in global. For example, MPEG-4 ALS has introduced this app and has been verified to apply to medical practices, for example, ECG and additionally to stream [31]. FLAC has likewise settled that it requests to restorative frameworks, which utilize bio-signal information [32]. In spite of the fact that others are not connected in medicinal frameworks or transmissions, it thinks about conceivable purposes in this area, for example, in the case of standard IEEE 1857.2 and IEEE 1857.3, the standard presents two header file formats, one for data storage schemes and one for streaming, and transmission [12] [23].

VI. CONCLUSION

Had been estimated since 2000, lossless data compression has been known to evolve with variant algorithms for lossless compression. Most of which are popular algorithms such as MPEG4-ALS, MPEG4-SLS, FLAC. This paper presents research that was conducted on various aspects of standard compression lossless audio and algorithm impacts on the world of mobile communications, digital music, online music and video playback (streaming). This is a topic that has become a discussion since people are becoming more interested in reducing online file sizes and with higher quality and multi-channel audio files.

1. Over the years, many lossless codecs have been released and refined to the point of providing comparable performance and capability. The only distinguishing factor is the adoption rate and feature set that may appeal to different sets of audiences.  

2. From various studies, the IEEE 1857.2 and IEEE 1857.3 standard can provide the best level of compression by adding pre-processing blocks to reduce the dynamic range of predictive residues rather than encoding directly. Because of this the increased complexity of the algorithm affects the encoding time, but can greatly increase the compression ratio in the process. The IEEE 1857.3 audio compression tool not only promises a higher compression ratio but also better performance in terms of coding and decoding speed compared to other existing lossless codecs.
with LPC models like MPEG-4 ALS and FLAC.

3. With fixed-point format that encoder and decoder modules, although comparisons have explained that audio synthesis knowledge scheme such as MPEG-4 SLS have superior speed with the fact that the compression ratio is slightly lower.

4. This one of the good things that can be used as a reference is that although the standard compression IEEE 1857.3 audio application can provide more compression ratio, the FLAC codec is adaptive therefore compression ratio, encoding and decoding speed are higher with larger file sizes despite MD5 overhead checksum.

REFERENCES


<table>
<thead>
<tr>
<th>Method</th>
<th>MPEG-4 ALS (LPC)</th>
<th>MPEG-4 ALS (RLS-LMS)</th>
<th>MPEG-4 SLS</th>
<th>FLAC</th>
<th>Enhanced CELP</th>
<th>Enhanced SLS</th>
<th>IEEE 1857.3, IEEE 1857.2</th>
<th>Sparse Linear Predictor</th>
<th>Cascaded OLS-NLMS</th>
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</thead>
<tbody>
<tr>
<td><strong>Description</strong></td>
<td>the scheme in this standard indicates that the predictor coefficients and residual error are encoded with the bitstream to reconstruct the original audio data.</td>
<td>in this Concept removes the coefficients from the predictor results derived from the encoded source then replaces or alters the LPC scheme with the predictor scheme of the RLS-LMS.</td>
<td>The development of Lossy compression MPEG-4 AAC is the concept of this model with the addition of the concept of computational lossless layer.</td>
<td>FLAC can be used freely and extensively on the public license of BSD</td>
<td>in this standard scheme performs sample-by-sample adaptive encoding using outgoing code such as deletion between samples</td>
<td>Data Replacement of Laplacian Distribution Input to Gaussian for Entropy Block BPCG</td>
<td>The audio and video standards in this scheme use a new pre- and post-processor for flatten and de-flatten signals and a new approach to entropy coding.</td>
<td>Replaces LPC predictors with Sparse predictors.</td>
<td>Replace the RLS algorithm with OLS and LMS with Normalization LMS Filter.</td>
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<td><strong>Software Reference</strong></td>
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<td><strong>Advantage</strong></td>
<td>Lossless audio compression can be achieved.</td>
<td>Fast Decoding Speed</td>
<td>Scalability to improve the current MPEG-4 AAC Lossy algorithm</td>
<td>Adaptive computational complexity due to the interchangeable prediction model based on the audio signal block and the authentication data check by the MD5 code</td>
<td>Speed Encoding / Decoding is fast with MPEG-4 RLS-LMS, due to fixed excitation codebook</td>
<td>High compression ratio.</td>
<td>High compression ratio.</td>
<td>High compression ratio.</td>
<td>Low computational complexity (Faster encoding/decoding speed compared to MPEG-4 RLS-LMS)</td>
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<td><strong>Disadvantage</strong></td>
<td>Overhead coefficient of predictors during decoding demultiplexing.</td>
<td>Numerical instability with white or small variance signals.</td>
<td>High computation on arithmetic coding (Low Encoding / Decoding Speed).</td>
<td>Low compression ratio due to MD5 overhead and interchangeability model prediction with small wave file.</td>
<td>Comes from a collection of several of real audio wave signals in summarizing a set of stored and stored large stochastic codes</td>
<td>Average computation complexity of arithmetic coding (Average encoding/Decoding speed).</td>
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<td>Average decoding speed.</td>
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