Vector Quantization Using LBG Algorithm

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Abstract- Speech Compression is the field concerned with obtaining compact digital representations of voice signals for the purpose of efficient transmission or storage. In this context, Linde-Buzo-Gray (LBG) algorithm has been widely adopted for vector quantization-based speech compression. It is used for designing of codebook efficiently which has minimum distortion and error. This paper presents an optimized speech codebook efficiently which has minimum distortion and error. This paper presents an optimized speech compression algorithm using LBG.

Keywords-Speech compression, vector quantization, LindeBuzo-Gray (LBG) algorithm, Codebook Generation.

I.INTRODUCTION

Speech coding or compression is a process of obtaining a compact representation for the speech signals. It is used for the purpose of efficient transmission over band-limited wired or wireless channels and for efficient storage. The goal of speech coding is to represent the samples of a speech signal with a minimum number of bits. As bandwidth is the parameter which affects the cost of processing, speech signals are subjected to compression before transmission One of the most popular methods for speech compression is vector quantization.

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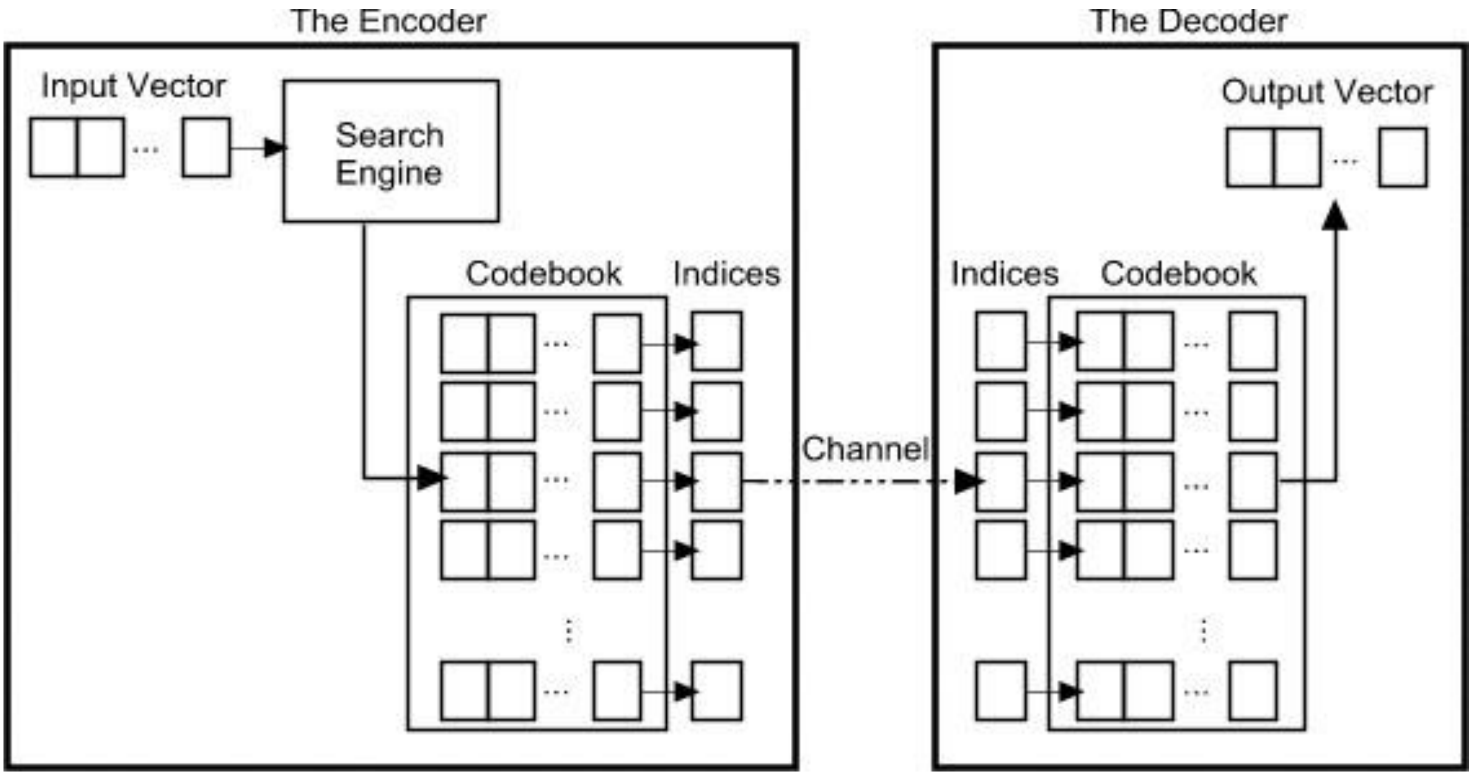
II. VECTOR QUANTIZATION

Quantization, the process of approximating continuousamplitude signals by digital (discrete amplitude) signals, is an important aspect of data compression or coding, the field concerned with the reduction of the number of bits necessary to transmit or store analog data, subject to a distortion or fidelity criterion. The joint quantization of a block of parameters is termed block or vector quantization. Compression is achieved by transmitting the index of the codeword instead of the vector itself. The key to Vector Quantization is to construct a good codebook of representative vectors. A vector quantizer is composed of two operations. The first is the encoder, and the second is the decoder. The encoder takes an input vector and outputs the index of the codeword that offers the lowest distortion. In this case the lowest distortion is found by evaluating the Euclidean distance between the input vector and each codeword in the codebook. Once the closest codeword is found, the index of that codeword is sent through a channel (the channel could be a computer storage, communications channel, and so on). When the encoder receives the index of the codeword, it replaces the index with the associated codeword. The most popular method for designing a codebook was proposed by Linde, Buzo and Gray .



II.CODEBOOK GENERATION

3. Obtain an initial codebook from the training sequence, which is the centroid or mean of the training sequence and let the initial codebook be

'C'.

4. Split the initial codebook C into a set of codewords

$$C_n^+ = C(1+\varepsilon), \quad \varepsilon = 0.01$$

is the minimum error to be obtained between old and new codewords

5 . Compute the difference between the training sequence and each of the codeword's C_n^+ and C_n^- and let the difference be 'D'.

6 . Split the training sequence into two regions R_1 and R_2 depending on the difference 'D' between the training sequence and the codeword's C_n^+ and C_n^-. The training vectors closer to C_n^+ falls in the region R_1 and the training vectors closer to C_n^- falls in the region R_2.

$$C_n^- = C(1-\varepsilon)$$

7. Let the training vectors falling in the region R 1 be TV 1 and the training sequence vectors falling in the region R 2 be TV 2 .

8. Obtain the new centroid or mean for TV 1 and TV 2 . Let the new centroids be C R1 and C R2 .

9 . Replace the old centroids C_n^+ and C_n^- by the new centroids C_{R1} and C_{R2}.

10 . Compute the difference between the training sequence and the new centroids C_{R1} and C_{R2} and let the difference be D^1.

$$\frac{D^1 - D}{D} < \varepsilon$$

11. Repeat steps 5 to 10 until

12 .Repeat steps 4 to 11 till the required number of codewords in the codebook are obtained.where represents the number of codewords in the codebook and '' represents the number of bits used for codebook generation, '' represents the difference between the training sequence and the old codewords and represents the difference between the training sequence and the new codewords.

Vector quantization of speech signals requires the generation of codebooks. The codebooks are designed using an iterative algorithm called Linde, Buzo and Gray (LBG) algorithm. The input to the LBG algorithm is a training sequence. The training sequence is the concatenation of a set LSF vectors obtained from people of different groups and of different ages. The speech signals used to obtain training sequence must be free of background noise. The speech signals can be recorded in sound proof booths, computer rooms and open

environments. In this work the speech signals are recorded in computer rooms. In practice speech data base like TIMIT database is available for use in speech coding and speech recognition.

The codebook generation using LBG algorithm requires the generation of an initial codebook, which is the centroid or mean obtained from the training sequence. The centroid obtained is then split into two centroids or codewords using the splitting method. The iterative LBG algorithm splits these two codeword's into four, four into eight and the process continues till the required numbers of codewords in the codebook are obtained.

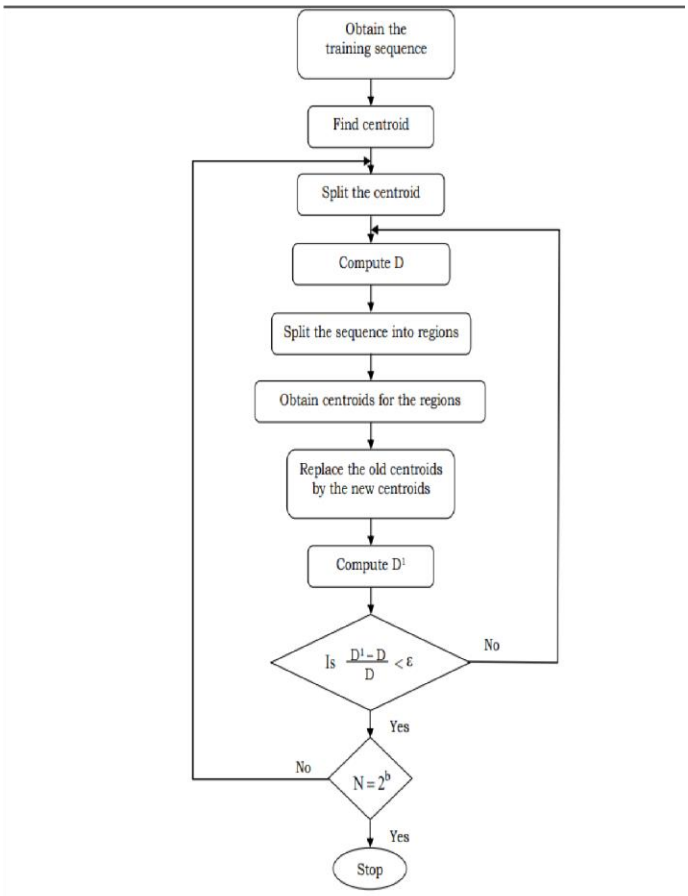
IV. LBG ALGORITHM

The LBG algorithm, also known as the Linde-Buzo-Gray algorithm, is a popular clustering algorithm used in signal processing and data compression.The flow chart of LBG

algorithm is shown in Fig 2. The LBG algorithm is properly implemented by a recursive procedure given below:

1. Initially the codebook generation requires a training sequence of LSF parameters which is the input to LBG algorithm. The train sequence is obtained from a set of speech samples recorded from different groups of people in a computer room

2. .Let 'R' be the region of the training sequence



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V.SPECTRAL DISTORTION MEASURE

The quality of the speech signal is an important parameter in speech coders and is measured in terms of spectral distortion measured in decibels (dB). The spectral distortion is measured between the LPC power spectra of the quantized and unquantized speech signals. The spectral distortion is measured frame wise and the average or mean of the spectral distortion calculated over all frames is taken as the final value of the spectral distorti on.

For an i th frame the spectral distortion (in dB), SD_i

is given by equation (1).

$$SD_i = \sqrt{\frac{1}{(f_2 - f_1)} \int_{f_1}^{f_2} \left[10\log_{10} s_i(f) - 10\log_{10} \hat{s}_i(f) \right]^2 df} \, (dB) \quad (1)$$

Where $s_i(f)$ and $\hat{s}_i(f)$ are the LPC power spectra of the un quantized and quantized i^{th} frame respectively. The frequency 'f' is in Hz and the frequency range is given by f_1 and f_2. The frequency range used in practice for narrowband speech coding is 0 to 4000 Hz

The average or mean of the spectral distortion SD is given by equation

$$SD = \frac{1}{N} \sum_{i=1}^{N} SD_i$$

The conditions for transparent speech coding are :

1. The average or mean of the spectral distortion (SD) must be less than or equal to 1dB.

2. There must be no outlier frames having a spectral distortion greater than 4dB.

3. The number of outlier frames between 2 to 4dB must be less than 2 %.

VI . RESULTS AND DISCUSSION

Frames having average spectral distortion greater than 1dB are considered transparent coding the average spectral distortion must be less than 1 dB.The expected results are as following

3.

Table : speech compression using LBG algorithm

Bits/Frames	**Spectral Distortion**	**Percentage of Outliers**	
		2-4 db	>4 db
24(8+8+8)	1.41088	0.2289	0.0301
23(8+8+7)	1.4107	0.2349	0.0241
22(8+7+7)	1.4349	0.2470	0.0904
21(7+7+7)	1.9156	0.2771	0.1807
20(7+7+6)	1.9186	0.2892	0.1024

VII. CONCLUSION

Hence we conclude that,Speech Compression focuses on reducing bit-rate of speech signals to enhance transmission speed and storage requirement of fast developing multimedia.

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