Speech Data Compression using Vector Quantization

H. B. Kekre, Tanuja K. Sarode

***Abstract*—**Mostly transforms are used for speech data compressions which are lossy algorithms. Such algorithms are tolerable for speech data compression since the loss in quality is not perceived by the human ear. However the vector quantization (VQ) has a potential to give more data compression maintaining the same quality. In this paper we propose speech data compression algorithm using vector quantization technique. We have used VQ algorithms LBG, KPE and FCG. The results table shows computational complexity of these three algorithms. Here we have introduced a new performance parameter Average Fractional Change in Speech Sample (AFCSS). Our FCG algorithm gives far better performance considering mean absolute error, AFCSS and complexity as compared to others.

***Keywords*—**Vector Quantization, Data Compression, Encoding,, Speech coding.

I. INTRODUCTION

THANKS for the rapid technological growth and the usage of the internet today has made possible for the transmission of multimedia applications over the web. The multimedia applications consist of mainly speech, images, and videos. These applications requires large amount of data resulting in consumption of huge bandwidth and storage resources. Vector quantization (VQ) [1]-[3] is an efficient technique for data compression and has been successfully used in various applications involving VQ-based encoding and VQ-based recognition. The response time is very important factor for real time application [1]. Many type of VQ, such as classified VQ [9], [10], address VQ[9], [11], finite state VQ[9], [12], side match VQ[9], [13], mean-removed classified VQ[9], [14], and predictive classified VQ[9], [15], have been used for various purpose. VQ has been applied to some other applications, such as index compression [9], [16], and inverse half toning [9], [17], [18]. VQ has been very popular in a variety of research fields such as speech recognition and face detection [5], [19], pattern recognition [22]. VQ is also used in real time applications

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such as real time video-based event detection [5], [20] and anomaly intrusion detection systems [5], [21].

VQ can be defined as a mapping function that maps k-dimensional vector space to a finite set $CB = \{C_1, C_2, C_3, ..., C_N\}$. The set CB is called codebook consisting of N number of codevectors and each codevector $C_i = \{c_{i1}, c_{i2}, c_{i3}, ..., c_{ik}\}$ is of dimension k. The key to VQ is the good codebook. Codebook can be generated in spatial domain by clustering algorithms or using transform domain techniques [6]-[8]. The method most commonly used to generate codebook is the Linde-Buzo-Gray (LBG) algorithm [3], [4] which is also called as Generalized Lloyd Algorithm (GLA).

Speech coding techniques are mostly based on lossy algorithms. Lossy algorithms are considered acceptable when encoding speech because the loss in quality is undetectable by the human ear. Uncompressed speech is usually transmitted at 64 kb/s, using 8 bits/sample and at a rate of 8 KHZ for sampling. Any bit rate below 64 kb/s is considered as compression. There are many lossy compression technique present in literature called Linear predictive coding [24]-[26], Multi stage vector quantization technique (MSVQ) [27]-[29], Switched split vector quantization technique (SSVQ) [27], Multi switched split vector quantization (MSSVQ) [23]. In this paper we propose speech coding techniques using VQ algorithms LBG, Kekre's Proportionate Error KPE[30]-[32] and Kekre's Fast codebook Generation FCG[31]. Here we have generated codebooks of size 256 for six different speech samples considering vector dimension 16 and 32.

In the next section we present VQ algorithms LBG, KPE, and FCG. In section III consist of results and conclusions in section IV.

II. CODEBOOK GENERATION ALGORITHMS

A. LBG Algorithm [3], [4]

In this algorithm centroid is computed as the first codevector for the training set. In Fig. 1 two vectors v_1 & v_2 are generated by adding constant error to the codevector. Euclidean distances of all the training vectors are computed with vectors v_1 & v_2 and two clusters are formed based on nearest of v_1 or v_2. This procedure is repeated for every cluster. The drawback of this algorithm is that the cluster elongation is $+135^\circ$ to horizontal axis in two dimensional cases. This results in inefficient clustering.
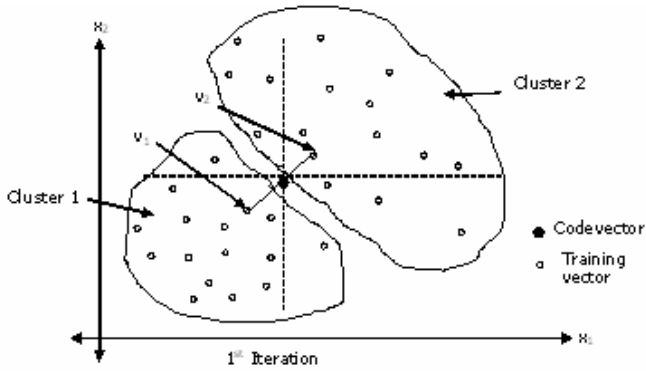
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Fig. 1. LBG for 2 dimensional case

B. Proportionate Error Algorithm (KPE) [30]- [32]

Here proportionate error is added to the centroid to generate two vectors v_1 & v_2. Magnitude of members of the centroid decides the error ratio. Hereafter the procedure is same as that of LBG. While adding proportionate error a safe guard is also introduced so that neither v_1 nor v_2 go beyond the training vector space. This removes the disadvantage of the LBG.

Both LBG and KPE requires 2M number of Euclidean distance computations and 2M number of comparisons where M is the total number of training vectors in every iteration to generate clusters.

C. Fast Codebook Generation Algorithm (FCG)

We have proposed FCG for image data compression [32]. In this paper we are proposing this algorithm for speech data compression along with LBG and KPE and comparative performance of these algorithms is given.

Let **T= {X₁, X₂, ……, X_M}** be the training sequence consisting of **M** source vector. Assume that source vector is of length **K**, **X_m={x_{m,1}, x_{m,2}, …… x_{m,K}}** for **m=1,2, …., M.** Let **N** be the number of codevectors and let C = {c₁, c₂ ,………….., c_N},represents the codebook. Each codevector is k dimensional,
e.g., **c_n = (c_{n,1} , c_{n,2} ,………….., c_{n,k}), n= 1, 2 ,….., N.**

In this algorithm codebook is generated based on comparison technique and hence is faster as compared to LBG and KPE. Initially we have only one cluster, centroid of this cluster is computed and then this cluster is split into two parts by comparing the first element of all the vectors present in the cluster with the first element of the centriod.

For all $x_{i,1}$ where i = 1, 2, ….. , M if $x_{i,1} < c_{1,1}$ then $x_{i,1}$ is grouped into cluster 1 or else $x_{i,1}$ is grouped into cluster 2. Centroid of cluster 1 (i.e. c₁) and cluster 2 (i.e. c₂) is computed and cluster 1 is split into two by comparing the second element of all the vectors present in the cluster 1 with the second element of the centroid c₁. Similarly cluster 2 is split into two by comparing the second element of all the vectors present in the cluster 2 with the second element of the centroid c₂. Now four clusters are formed centroids of all these clusters are computed and each of these cluster is split further by comparing the third element of all vectors with the third element of the centroid. The process is repeated further till the codebook of desire size is obtained. The above process is depicted in Fig. 2 for two dimensional case.
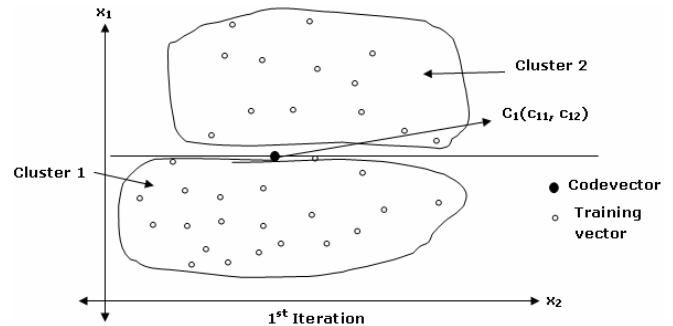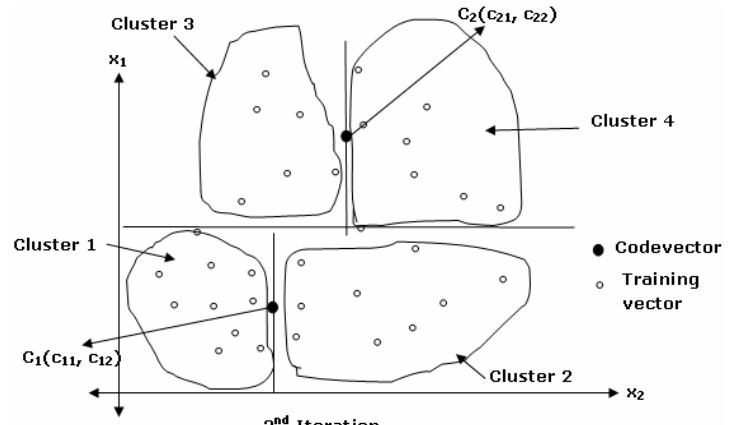

Fig. 2a

Fig. 2b

Fig. 2 FCG algorithm for 2 dimensional case

It is observed from the results that this algorithm gives minimum error and also least time to generate codebook as compared to all other clustering algorithm i.e LBG and KPE. This algorithm requires only comparisons for generating clusters. For M training vectors we require M comparisons in each iteration.

III. RESULTS

VQ has many applications in this paper we have chosen speech compression as an application. The algorithms are implemented using Pentium IV 1.7 GHz 512 MB RAM, using Matlab 6.

Here we have introduced a new performance parameter which is named as Average Fractional Change in Speech Sample (AFCSS) and it is computed as follows:

$$\frac{1}{P} \sum_P \frac{\left| f(x) - \hat{f}(x) \right|}{f(x)} \qquad (1)$$

Where $f(x)$ is original speech signal of size P and $\hat{f}(x)$ is the reconstructed speech signal

Table 1 shows the comparison of LBG, KPE, FCG with respect to Average fractional change in speech sample (AFCSS) and Mean Absolute Error for six different speech signals.

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Table 2 shows the comparison of LBG, KPE and FCG algorithm with respect to total number of Comparisons, total number of ED computations and total CPU units required.

Using formulae given in table 2, table 3 gives the total CPU units required for all six speech samples.Table 3 shows Total CPU units required for all speech samples.

TABLE I COMPARISON OF LBG, KPE, AND FCG ALGORITHMS WITH RESPECT TO AVERAGE FRACTIONAL CHANGE IN SPEECH SAMPLE AFCSS AND MEAN ABSOLUTE ERROR FOR SIX DIFFERENT SPEECH SIGNALS

Speakers	Performance Criteria	Codebook Size 256								
		Vector Dimension 8			Vector Dimension 16			Vector Dimension 32		
		FCG	LBG	KPE	FCG	LBG	KPE	FCG	LBG	KPE
HBK	AFCSS	0.6052	0.7294	0.7291	0.7585	0.8919	0.8912	0.9148	1.1378	1.0196
	MAE	0.0012	0.0018	0.0018	0.0017	0.0024	0.0024	0.0021	0.0027	0.0027
Tanuja	AFCSS	0.7657	0.9916	0.9924	1.0602	1.1280	1.1120	1.3207	1.5167	1.4275
	MAE	0.0028	0.0041	0.0041	0.0041	0.0052	0.0051	0.005	0.0064	0.0058
Sudeep	AFCSS	0.9070	1.0434	1.0247	1.0991	1.2329	1.2335	1.1534	1.2022	1.2069
	MAE	0.0039	0.0055	0.0054	0.0047	0.0066	0.0065	0.0045	0.0066	0.0062
Chetan	AFCSS	0.3902	0.6048	0.5906	0.6161	0.8087	0.8063	0.8734	0.9158	0.9107
	MAE	0.0017	0.0026	0.0025	0.0026	0.0033	0.0033	0.0033	0.004	0.0039
Saylee	AFCSS	0.3234	0.4629	0.4579	0.4296	0.5421	0.5429	0.4182	0.5416	0.5679
	MAE	0.0012	0.0020	0.0019	0.0016	0.0023	0.0023	0.0015	0.0026	0.0026
Karishma	AFCSS	0.3577	0.5425	0.5434	0.5540	0.6837	0.6673	0.7600	0.8737	0.7778
	MAE	0.0027	0.0044	0.0043	0.0042	0.0058	0.0055	0.0055	0.0071	0.0068

TABLE II COMPARISON OF LBG, KPE AND FCG ALGORITHM WITH RESPECT TO TOTAL NUMBER OF COMPARISONS, TOTAL NUMBER OF ED COMPUTATIONS AND TOTAL CPU UNITS REQUIRED

Complexity Parameters	FCG	LBG	KPE
Total Comparisons	NM	2MN	2MN
Total No. of ED	0	2MN	2MN
Total CPU units	NM	4MN(10k-1)	4MN(10k-1)

TABLE III TOTAL CPU UNITS REQUIRED FOR ALL SPEECH SAMPLES

Speakers	Codebook Size 256								
	Vector Dimension 8			Vector Dimension 16			Vector Dimension 32		
	No. of Training Vectors	FCG	LBG / KPE	No. of Training Vectors	FCG	LBG / KPE	No. of Training Vectors	FCG	LBG / KPE
HBK	6179	1581824	499856384	3089	790784	502938624	1544	395264	504356864
Tanuja	4940	1264640	399626240	2470	632320	402155520	1235	316160	403420160
Sudeep	3200	819200	258867200	1600	409600	260505600	800	204800	261324800
Chetan	5840	1495040	472432640	2920	747520	475422720	1460	373760	476917760
Saylee	3480	890880	281518080	1740	445440	283299840	870	222720	284190720
Karishma	8820	2257920	713502720	4410	1128960	718018560	2205	564480	720276480

IV. CONCLUSION

In this paper we have used 8 KHZ, 16 bit speech signal which results in very fine quantization levels (i.e 2^{16}). We have considered here three cases with different vector dimensions 8, 16 and 32 for codebook size 256 resulting in compression ratios 16:1, 32:1 and 64:1 respectively. We have used three VQ algorithms LBG, KPE, and FCG for speech coding. Among these three VQ algorithms FCG is faster as compared to LBG and KPE and further FCG algorithm gives 65% less MAE and approximately 20% less AFCSS as compared to LBG and KPE. Regarding complexity of

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algorithms LBG and KPE are more complex rather than FCG. Among LBG and KPE it is observed that KPE gives less MAE and AFCSS than LBG. The requirement of CPU units for LBG and KPE algorithms is 300 times more than FCG algorithm. At the same time the CPU units required for FCG algorithm reduce two and four times for vector dimension 16 and 32 respectively as compared to vector dimension 8.

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