# Vector Quantization using Speech Signal Property

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**ABSTRACT**

In this paper, we have proposed a VQ algorithm which uses a generating order to make quantize feature vector of speech signal. The proposed algorithm inspects what codeword follows after present codeword and adds new index to established codebook, when mapping speech signal. We present a variable bit rate for new codebook, and propose an efficient compressed way of information. In this way, the number of computation and the number of codewords to be searched are reduced considerably.

The performance of the proposed VQ algorithm is evaluated by spectrum distortion measure and bit rate. The obtained spectrum distortion is reduced about 0.22[dB], and the bit rate is saved over 0.21 bit/frame.

## I. Introduction

The transmission of speech signal using Vector Quantization has been researched very much, because efficiency of compression is excellent. VQ inevitably has error from original signal, but codebook size becomes larger when the error is reduced[9]. Enlarged codebook size requires larger bits and computation amount. VQ is divided into two categories, that is, memory VQ and memoryless VQ[1]. Full search VQ, tree structure VQ, classified VQ, partitioned VQ, hierachical VQ, transform VQ, multistage VQ and lattice codebook VQ belong to memory VQ. Predictive VQ, finite-state VQ and trellis VQ belong to memoryless VQ.

This paper proposes a VQ algorithm with memory using speech signal's generating order[1]. The proposed algorithm inspects what codeword follows after present codeword and adds new index to established codebook using memoryless splitting algorithm[1][4]. In this way, the number of computation and the number of codewords to be searched are reduced considerably. And we present a variable bit rate[6] for new codebook, and propose an efficient compressed way of information. The performance of the proposed VQ algorithm is evaluated by spectrum distortion measure and bit rate.

## Ⅱ. Proposed VQ algorithm

2.1 Generating range of vector

The proposed VQ algorithm makes use of the fact that feature vector of speech signal is generating with some relation in a limited range. For example, if there are 12 bits quantization, it means that has 4096 levels and it covers from index 0 to index 4095 code vector, that is 4096 levels can generate for code vector follow after each code vector. But simulated result shows that every 4096 code vector is not generated, but some part of 4096 code vector is generated, and a number of certain code vector is generated all at once.

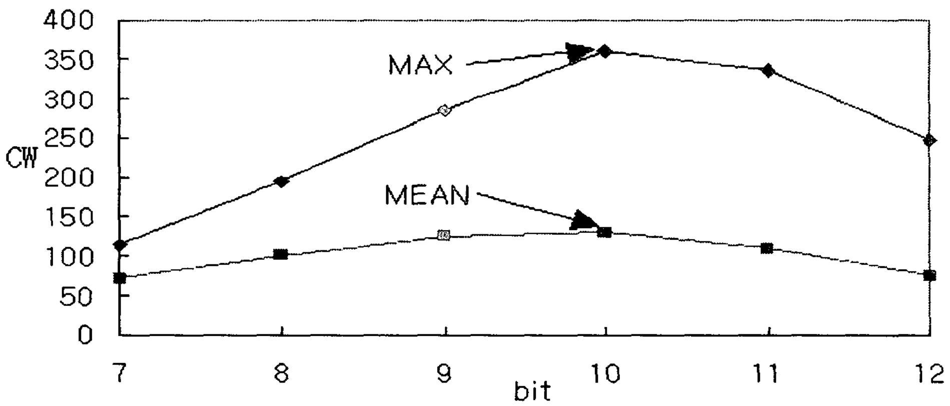


Fig.1 Maximum and mean number of generating codeword for various number of quantization bit

Fig.1 shows that x-axis represents quantization level from 7 bits to 12 bits, and y-axis represents maximum and mean generating codeword number per each quantization level.

VQ inevitably has error from original signal, but codebook size become larger when the error is reduced[9]. But codebook size become larger and then become larger required bits and computation amount. But if we use the VQ algorithm proposed in this paper, then error from original signal is reduced and also required bits & computation amount is reduced.

2.2 Proposed VQ algorithm

VQ is divided into two categories, that is memory VQ and memoryless VQ[1]. Proposed VQ belongs to memory VQ. It is a strong point that the code vector proposed in this paper could have number of bit as variable.

2.2.1 Making sub-codebook

First, we design full size codebook of planned quantization level by using splitting algorithm. And as we mentioned above section 2.1, feature vector of speech signal is generating with some relation in a limited range. Using this point, possible code vectors following after each code vector are collected to make new sub-codebook and then instead of full size codebook with splitting algorithm, this new sub-codebook is used for quantization. In this way we reduce not only required bits but also computation amount, and efficiency of quantization is as good as codebook generally used before.

2.2.2 Vector quantizer

Fig.2 shows encoding algorithm using new sub-codebook and describes this algorithm as shown below.

stpe 1 : First input vector is compared with full size codebook and we search the closest code vector index to first one and save it in memory buffer and transmit the value.

step 2 : Index value of previous input vector saved in memory buffer. Sub-codebook which consist of index of code vector expected to come out after saved each of them. Index value, sub-codebook and full size codebook are used for mapping each of input vector from second to last one. After saved sub-codebook indices resulted from mapping and transmit the index value.

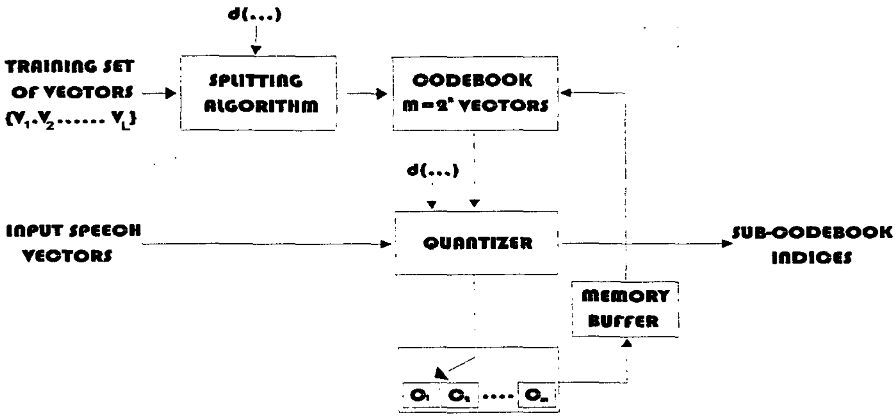


Fig.2 Vector quantizer

2.2.3Vector dequantizer

Fig.3 shows decoding algorithm using new sub-codebook and describes this algorithm as shown below.

step 1 : After first transmitted index is compared with full size codebook, we synthesize reproduction vector, and the transmitted index is saved in memory buffer.

step 2 : From second transmitted index to last transmitted index, using previous transmitted index in memory buffer and full size codebook, we synthesize reproduction vector and save transmitted index in memory buffer.

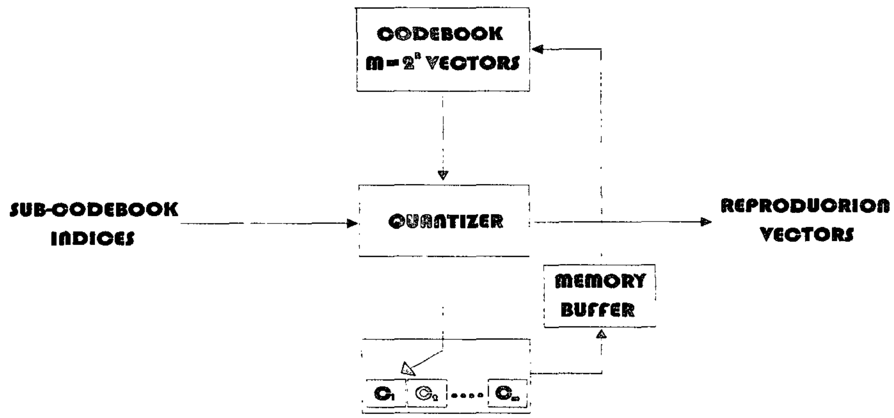


Fig.3 Vector dequantizer

## Ⅲ. Simulation and result

For the simulation, 12 male speakers and 1 female speaker pronounce a random 5,000 compositions which are selected from a book. 5,000 original analog signals convert to digital signals using ADi1848 board with 8 kHz sampling rate, 16 bits quantization level. With obtained 434,756 frames as a result of LPC analysis applied interpolation, we desigend the reference codebook which has 7 bits ~ 12 bits quantization level using full search VQ. And for evaluating reference codebook and proposed VQ algorithm, we used five compositions that 1 male speaker pronounced 20 times each.

The performance of the proposed VQ algorithm is evaluated by spectrum distortion measure and requried bits per each bit rate between full search VQ algorithm and proposed VQ algorithm. Spectrum distortion is expressed by dB. Simulation results are shown in table 1, fig.4 and fig.5. Here, VQ represents full search VQ and PVQ represents proposed VQ.

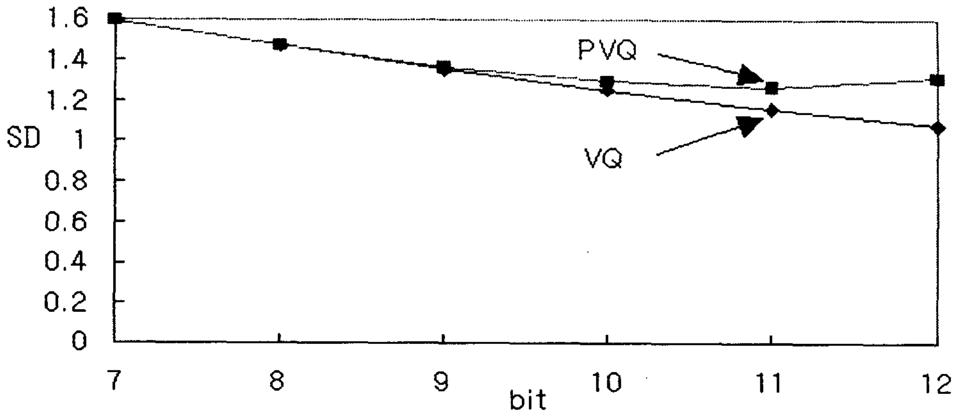


Fig.4 Spectrum distortion for various number of quantization bit

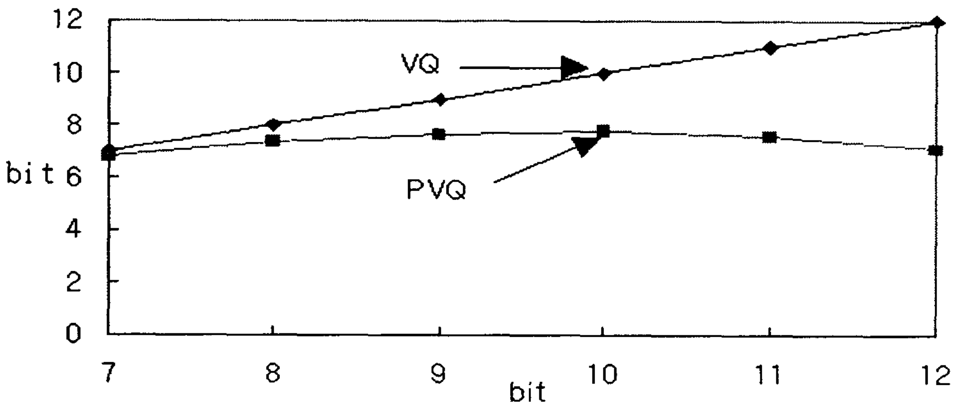


Fig.5 Requried bits for various number of quantization bit

Fig.4 shows that x-axis represents quantization level from 7 bits to 12 bits, and y-axis represents spectrum distorton per each quantization level. And fig.5 shows that x-axis also repesents quantization level from 7 bits to 12 bits, and y-axis represents required bits per each quantization level. Table 1 expresses fig.4 and fig.5 numerically.

Fig.4 shows that spectrum distortion of full search VQ is smaller than proposed VQ. And as quantization level of full search VQ becomes larger, spectrum distortion becomes smaller. But spectrum distortion of proposed VQ is minimum at 2048 (11 bits) level and increased over 4096 (12 bits) level. On the other hand, fig.5 shows that required bits of full search VQ was larger as quantization level increases, but required bits of proposed VQ is maximum at 1024(10 bits) level and decreased over 2048 (11 bits) level. And the amount of bits of increase per each quantization level are much smaller than full search VQ.

Table 1 shows not only that 256 level's spectrum distortion of full search VQ is larger than 512, 1024, 2048, 4096 level's spectrum distortion of proposed VQ, but also required bits of full search VQ is larger than proposed VQ. And, not only 512 level's spectrum distortion of full search is larger than 1024, 2048, 4096 level's spectrum distiortion of proposed VQ, but also required bits of full search VQ is larger than proposed VQ.

Table 1. The experiment results

| bits& SD LEVEL | Required bits of full search VQ | Required bits of proposed VQ | SD of full search VQ | SD of proposed VQ |
|---|---|---|---|---|
| 128 | 7 bits | 6.79 bits | 1.591 dB | 1.591 dB |
| 256 | 8 bits | 7.36 bits | 1.470 dB | 1.471 dB |
| 512 | 9 bits | 7.65 bits | 1.347 dB | 1.357 dB |
| 1024 | 10bits | 7.79 bits | 1.251 dB | 1.288 dB |
| 2048 | 11bits | 7.59 bits | 1.156 dB | 1.260 dB |
| 4096 | 12bits | 7.10 bits | 1.076 dB | 1.307 dB |

We give attention to proposed VQ that required bits per each quantization level increases until maximum at 1024 (10 bits) level and decreases over 2048 (12 bits) level. And also spectrum distortion per each quantization level decreases until minimum at 2048 (11 bits) level and increases over 4096 (12 bits) level. Ist means that the reference data space is enough till 1024 level but is not enough over 2048 level. Therefore if we have a simulation in a good enough reference data space[1], then we have a better result. Generally speaking, proposed VQ algorithm is like that specrum distortion of full search is smaller as quantization level increases. And required bits of full search VQ are larger as quantization level increases. But spectrum distortion becomes inverse case over 2048 level and required bits become inverse case over 1024 level.

## Ⅳ. Conclusion

This paper has proposed a VQ algorithm which uses generating order to make a quantize feature vector of speech signal. The proposed algorithm inspects what generating codeword follows after present codeword and adds new index to established codebook. And we have a result that shows that required bits and amount of computation for quantization are reduced.

The performance of the proposed VQ algorithm is evaluated by spectrum distortion measure and bit rate. In the bit saving aspect, we have saved more than 0.22 bit per frame, and in the spectrum distortion aspect, we have reduced maximum 0.21 dB per frame.

Therefore, we confirmed the efficiency of a proposed VQ algorithm, and if we have a simulation in a good enough reference data space in future, then we have a better result.

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