퍼지 벡터 양자화기 사상화와 신경망에 의한 화자적응 음성합성

이 진 이† · 이 광 형††

요 약

본 연구에서는 퍼지사상화(fuzzy mapping)와 FLVQ(fuzzy learning vector quantization)에 의한 사상된(mapped) 코드북을 사용하는 화자적응 음성합성 알고리즘을 제안하고, 기존의 음성합성결과와 비교한다. 입력화자와 기준화자의 코드북은 FLVQ 방법으로 작성한다. 사상된 코드북은 퍼지 히스토그램을 작성하여 이들을 선형결합함으로써 얻어지는 퍼지 사상화에 의하여 작성된다. 대응 코드벡터의 퍼지 히스토그램은 동일 입력벡터에 대해 선택된 입력화자의 코드벡터와 기준화자의 코드벡터 사이의 DTW(dynamic time warping)을 행하여 대응하는 코드벡터들의 소속값(membership value)을 누적하여 얻는다. 음성합성시에는 사상된 코드북을 사용하여 입력화자의 음성을 퍼지벡터 양자화한 다음, FCM (fuzzy c means) 합성규칙을 사용하여 사상된 코드북내의 코드벡터가 아닌 새로운 하나의 합성벡터를 얻게 되어 좀 더 입력화자에 적응된 합성음을 얻게 된다. 이 기술의 성능평가는 성별이 서로 다른 화자를 입력화자및 기준화자로 선정하여 입력화자의 음성에 가까운 정도로 평가하였으며 그 결과 기존의 음성합성보다 입력화자에 더 적응된 합성음을 얻었다.

Speaker-Adaptive Speech Synthesis based on Fuzzy Vector Quantizer Mapping and Neural Networks

Jin-Yi Lee† · Kwang-Hyung Lee††

ABSTRACT

This paper is concerned with the problem of speaker-adaptive speech synthesis method using a mapped codebook designed by fuzzy mapping on FLVQ(Fuzzy Learning Vector Quantization). The FLVQ is used to design both input and reference speaker's codebook. This algorithm is incorporated fuzzy membership function into the LVQ(learning vector quantization) networks. Unlike the LVQ algorithm, this algorithm minimizes the network output errors which are the differences of class membership target and actual membership values, and results to minimize the distances between training patterns and competing neurons. Speaker Adaptation in speech synthesis is performed as follows; input speaker's codebook is mapped a reference speaker's codebook in fuzzy concepts. The Fuzzy VQ mapping replaces a codevector preserving its fuzzy membership function. The codevector correspondence histogram is obtained by accumulating the vector correspondence along the DTW optimal path. We use the Fuzzy VQ mapping to design a mapped codebook. The mapped codebook is defined as a linear combination of reference speaker's vectors using each fuzzy histogram as a weighting function with membership values. In adaptive-speech synthesis stage, input speech is fuzzy vector-quantized by the mapped

† 정 회 원:충남산업대학교 전자공학과
†† 정 회 원:숭실대학교 전자공학과
논문접수:1995년 9월 23일, 심사완료:1996년 1월 12일

codebook, and then FCM arithmetic is used to synthesize speech adapted to input speaker.

The speaker adaption experiments are carried out using speech of males in their thirties as input speaker's speech, and a female in her twenties as reference speaker's speech. Speeches used in experiments are sentences /anyoung hasim nika/ and /good morning/. As a results of experiments, we obtained a synthesized speech adapted to input speaker.

1. Introduction

Conventional speaker adaptation algorithms have been mainly useful for speaker-independent speech recognition system. We applied this method to synthesize a speech containing individuality. Speech individuality is above all important in daily communication. When conversation is made through a telephon line, it is expecially important in speaker indentification. Therefore a technique to control speech individuality plays a great role in speech processing, and offers to many applications. The purpose of this study is to adapt unknown input speaker to reference speaker in speech synthesis and is to develop technique of endowing individuality speech with synthesized speech. Speaker adaptation method using the fuzzy VQ(Vector Quantization)[1] mapping membership functions is higher in recognition rate than conventional methods in speech recognition[2]. We have improved synthesized speech quality by using FVQ in provious study[3].

In this paper, we propose a speaker-adaptive speech synthesis method which can improve the intimacy of synthesized speech quality by endowing questioner's individuality speech with answer's response, getting out of a meaning communication speech synthesis in a general way. We consider that this method can be useful in the Q/A (Query/Answer) system.

Generally speech individuality consists of two major factors. One is acoustic features and the other is prosodic features. Here to begin with, we improve synthesized speech quality by controlling acoustic features. According to as far study results, the acoustic features that contribute speech individuality are relied among various parameters, such as formant frequencies and-bandwidth, spectral slope, glottal waveforms[4]. In some studies these parameters have been synthesized in a separate way. However, we synthesize speech not to be seperated each of above parameters in order to avoid complexity. For this purpose, we use FLVQ codebook containing all individual speech information. We have shown that the VQ's performance mainly depends on codebook's quality consisted of representive vectors.

Methods of codebook design are iterative clustering algorithms known in the pattern-recognition literature as k-means or LBG algorithm. Recently, we have designed codebooks by using fuzzy clustering algorithm to consider membership values between subspaces[5]. These methods shows the danger that characteristic of input speech may be lost due to large amount of computation. Alternative training algorithm to solve this problem is a FLVQ[6] using fuzzy concepts and LVQ networks.

A fundamental benefit of formulating method of codebook design as a neural networks task is that the large body of neural networks training algorithms can be used to solve this problems. Here speaker-independent codebook is designed by FLVQ training algorithm.

The basic idea of adaptation to the other speaker's acoustic features from unknown speaker's is based on a mapping between codebooks of different speakers. The codebook mapping for speaker-adaption seeks to determine a mapping function between different speaker's parameter domain.

The mapped codebook is designed by using FLVQ, fuzzy mapping[7], fuzzy histogram, and DTW(Dynamic Time Warping). In next section, we present a FLVQ codebook design for speaker-adaptation speech synthesis. Section 3 deals with Fuzzy VQ mapping algorithm. In section 4, speaker-adaptation speech synthesis experiments are described. We present concluding remarks in section 5.

2. FLVQ-codebook design

LVQ codebook training algorithms formulating neural networks task are as such: 1) the competitive learning(CL) network that only learns "winning" neural unit, 2) the Kohonen self organizing feature map(KSFM) that, during the training process, the winning neural unit as well as the neural units in the neighbourhood of the winner is updated, and 3) the frequency sensitive competive learning(FSCL) network that results in maximum entropy used in encoding the data set because, by keeping a count of how frequently each neural unit is the winner during the course of the training process, all neural units are modified to an approximately equal number of times [8][9].

We construct the codebooks by using FLVQ network, considering that above mentioned VQ algorithm have problems of initial net state, sensitivity to rugged decision surface or overlapping data sets, and neuron under utilization.

Let input vectors $X=\{x_i \mid i=1,\dots,N\}$,
neuron's weight vectors $Y=\{y_j \mid j=1,\dots,L\}$
and membership functions $U=\{\mu_{ij} \mid i=1,\dots,N; j=1,\dots,L\}$.

The FLVQ algorithm is to minimize the objective function

$$\min Z(U, Y)=\sum_{i=1}^{N}\sum_{j=1}^{L}\{tmv^{F}-(\mu_{ij})^{F}\}\, d(x_i, y_j) \qquad (1)$$

subjected to

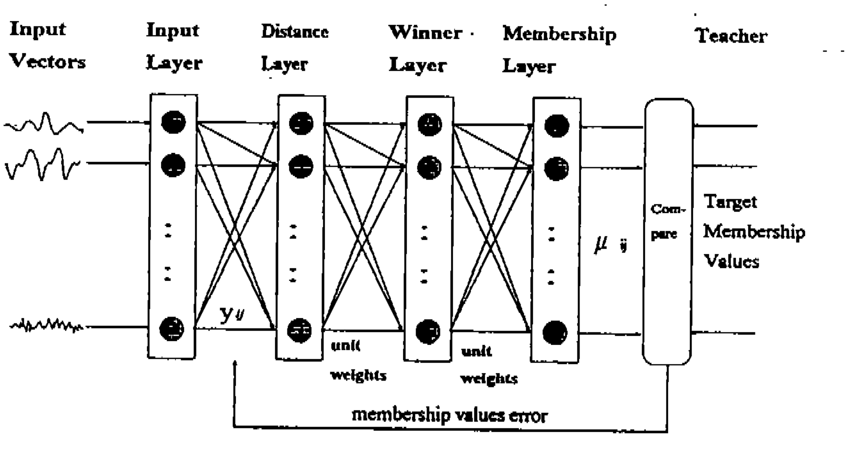
$$\sum_{j=1}^{L}\mu_{ij}=1;\ \forall i\ (i=1,\dots,N) \text{ and}$$
$$\mu_{ij}\in[1,0];\ \forall i, j\ (j=1,\dots,L) \qquad (2)$$

where $d(x_i, y_j)=|x_i-y_j|^2$ is distance between input vector x_i and its weight vector pattern y_j, and tmv is target membership value. F is fuzziness. By fixing U and applying the steepest descent optimization to $Z(U, Y)$,

FLVQ learning law is obtained.

$$y_j(n+1)=y_j(n)+\varepsilon(n)\,[(tmv^{F}-\mu_{ij}^{F})\,(x_i-y_j(n))] \qquad (3)$$

By fixing Y and applying the Lagrange multiplier



(Fig. 1) Fuzzy Learning VQ Networks

method to U, the membership updating rule is obtained.

$$\mu_{ij}=\frac{1}{\sum_{k=1}^{L}\left\{\frac{d(x_i, y_j)}{d(x_i, y_k)}\right\}^{1/(F-1)}} \tag{4}$$

In Fig. 1, the FLVQ network architure is depicted. The distance layer computes the distances $d(x_i, y_j)$ which are then fed forward to the membership layer to calculate/update the class membership outputs using equation (4). During the learning stage, the membership outputs are fed back to modify the neuron's weight vectors according to equation(3).

3. Speaker-adaptive speech synthesis by FVQ mapping on FLVQ

Speaker-adaptive speech synthesis performs through two steps. One is training step, and another is adaptive speech synthesis step. At training step, we generate a fuzzy mapped codebook. We use the fuzzy mapped codebook to synthesize speech adapted to input speaker. Fuzzy vector quantization is a powerful approach to reducing VQ synthesis distortion because it represents an input vector by using a weighted linear combination of VQ codevectors.

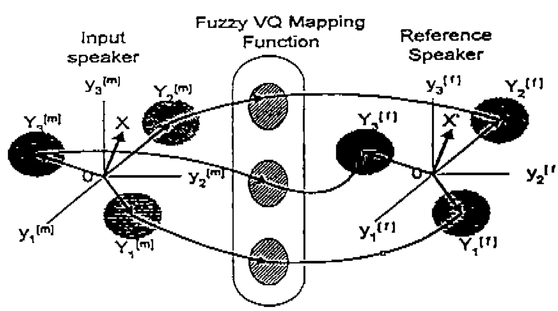
3.1 Construction of mapped codebook.

Mapped codebook(male→femal) is constructed by Fuzzy VQ mapping. Fuzzy VQ mapping replaces a codevector while preserving its fuzzy membership function. Therefore, mapped codebook is constructed by a linear combination of reference speaker's vectors using each histogram as weighting coefficients. The FVQ analysis stage generates a vector containing individuality speech information, whose components represent the degree to which input vector matches reference vector.

The FVQ-mapping between input speaker's vector space and reference speaker's performs through FVQ-mapping function F(m→f), and then the codebook mapped an input speaker's vector into a refererence speaker's is constructed. This codevectors of mapped codebook are calculated as

$$Y_i^{(m\to f)}=\frac{\sum_{j=1}^{L} h_{ij}\, Y_j^{(f)}}{\sum_{j=1}^{L} h_{ij}} \quad i=1,\dots,L \tag{5}$$

Where mapped codebook size is the same with input and reference codebook size. The i, j represent each input and reference speaker's codevector order. Fig. 2(a) depicts speaker adaptation analysis stage to find Fuzzy VQ Mapping function $F(m\to f)$ in 3 dimension domain. Input codevectors have relations of one to one correspondence with reference codevectors by Fuzzy VQ mapping function, where X, X′ are input and reference speaker's vector, respectively.



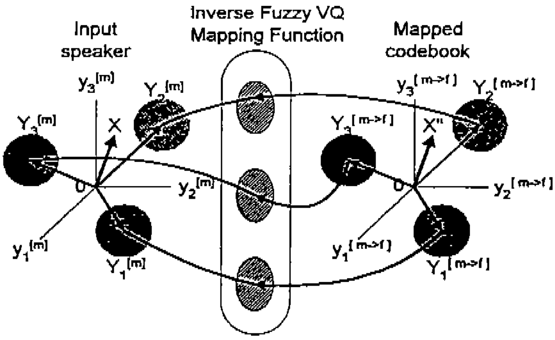
(Fig. 2(a)) Speaker adaptation analysis stage using Fuzzy VQ Mapping function $F(m\to f)$ in / anyoung hasim nika/ domain

Inverse Fuzzy VQ mapping performed by inverse Fuzzy VQ mapping function $F^{-1}(m\to f)$ represents a correspondence degree of reference speaker's individuality speech to input speaker's.

$$Y_i^{(m\to f)} \xrightarrow{\;F^{-1}(m\to f)\;} Y_i^{(m)} \tag{6}$$

Where $F^{-1}(m\to f)$ is a inverse fuzzy-VQ mapping function. Fig. 2(b) shows speaker-adaptation synthesis stage through inverse fuzzy VQ mapping function F^{-1}

$(m \to f)$. Input speaker's codevectors have relations of one to one correspondence with mapped codevectors by inverse fuzzy VQ mapping function.



(Fig. 2(b)) Speaker adaptation synthesis stage using inverse Fuzzy VQ Mapping Function($F^{-1}(m \to f)$ in /good morning/ domain

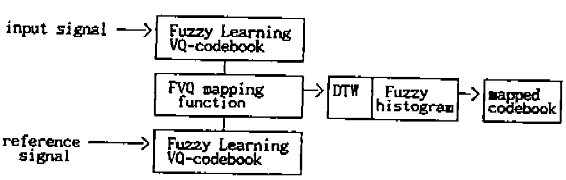
$Y_i^{(m)}$: input speaker's(male) codevectors

$Y_i^{(m \to f)}$: mapped(male-)femal) codebook's codevectors

X″: mapped codebook's codevectors corresponded to input vectors

The codevector's correspondence histogram is obtained from accumulating the vector correspondence along the DTW optimal path. By regarding the fuzzy membership function as correspondence probabilty, a correspondence histogram can be accurately approximated.

Fig.3 shows the generation of mapped codebook.



(Fig. 3) Mapped codebook generation

Training procedure for obtaining the mapped codebook is as follows:

1) Make an input speaker's and a reference's codebook using FLVQ training.
2) Fuzzy vector-quantize input vector.
3) Find the optimal codevector correspondence between input vector and reference vector using DTW.
4) Make fuzzy histogram (hij) accumulating the membership values using FVQ.

$$h_{ij} = h_{ij} + \mu_{ij} \tag{7}$$

Where i and j are the codevector order in input speaker's codebook and in reference speaker's codebook, respectively. While histogram in VQ-mapping is obtained by accumulating one whenever correspondence vector is found, histogram in FVQ-mapping is obtained by accumulating the membership values. Membership values(μ_{ij}) which represents the degree to which an input speaker's codevector(i) matches each reference speaker's vectors(j) can be determined as[10]:

$$\mu_{ij} = \frac{\left[\frac{1}{d(Y_i^{(m)}, Y_j^{(f)})}\right]^{\frac{1}{F-1}}}{\left[\frac{1}{d(Y_i^{(m)}, Y_k^{(f)})}\right]^{\frac{1}{F-1}}} \tag{8}$$

Where F: fuzziness

L: codebook size

$Y_i^{(m)}$: the ith codevector of input speaker(male)

$Y_j^{(m)}$: the jth codevector of reference speaker(female)

$Y_j^{(m \to f)}$: the jth codevector of mapped codebook

$1 \leq i \leq L,\ 1 \leq j \leq L$

5) Mapped codebook can be determined as:

$$Y_i^{(m)} \to Y_i^{(m \to f)} = \frac{\sum_{j=1}^{L} h_{ij} Y_j^{(f)}}{\sum_{j=1}^{L} h_{ij}} \tag{9}$$

Where h_{ij} is fuzzy histogram values.

6) Replace input speaker's codebook with the mapped codebook.
7) Until the average distortion has converge. goto 3).

3.2 Adaptive speech analysis-synthesis

We vector-quantize input speaker's speech by FVQ using mapped codebook, then fuzzy mapped membership output function(O_i) is obtained as the components of which is membership value.

$$O_i=[\mu_{i1}, \mu_{i2}, \mu_{i3},\ldots, \cdot\cdot\mu_{iL}]^T \qquad (10)$$

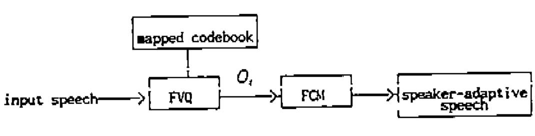
Component of the vector is positive, and sum of it's component is one. The F(Fuzziness)〉1 is a constant representing the degree of fuzziness, then determination of its value changes correspondence probability which an input vector is matched reference vector.

The O_i is obtained under the condition to minimize the fuzzy objective function of the following equation. Where n is input vector's order.

$$\min Z(U, Y^{(m\rightarrow f)})=\sum_{i=1}^{n}\sum_{j=1}^{L}(\mu_{ij})^F d(X_i, Y_j) \qquad (11)$$

Notice that as F tends to infinity, each component of O_i tends to 1/L; as F tends to one, then only the component corresponding to minimum value of $d(X_i, Y_j)$ tends to one and other components tend to zero. Thus decision of F value makes the FVQ's decision very hard(F→1) or very soft (F→∞). Fig. 4 shows the speaker-adaptive speech analysis and synthesis system.

Because we performs the FVQ synthesis by using the output vector of analysis stage and fuzzy-C-means operations, our method is obtained synthesized speech which is adapted a few more to input speech. The adaptive synthesized speech $\hat{X}_i$ is



(Fig. 4) Speaker-adaptive speech analysis-synthesis by fuzzy-VQ

$$\hat{X}_i=[\hat{x}_{i1}, \hat{x}_{i2}, \hat{x}_{i3},\ldots\ldots, \hat{x}_{iN}]^T \qquad (12)$$

The component of $\hat{X}_{ij}$, $\hat{x}_{ij}$ can be computed with the following equation.

$$\hat{x}_{ij}=\frac{\sum_{j=1}^{L}(\mu_{ij}^F\, y_{ji}^{(m\rightarrow f)})}{\sum_{j=1}^{L}\mu_{ij}^F},\ (1\le i\le n) \qquad (13)$$

4. Experiments of speaker-adaptive speech synthesis

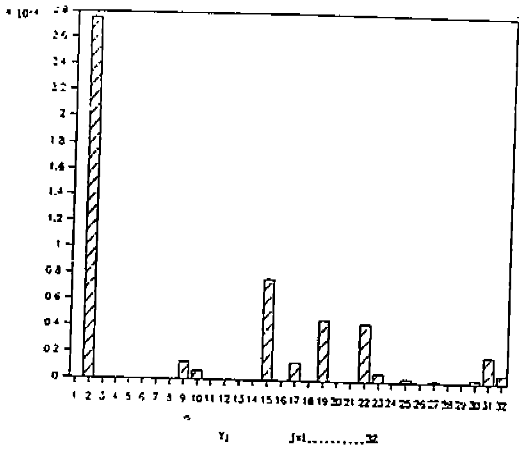
The speaker adaption experiments are carried out using speech of a male in his thirties(male 1), speech of a male in his twenties(male 2), and speech of a female in her twenties(female 1), as input speaker's speech, and speech of a other female in her twenties (female 2), as reference speaker's speech. Speech used in experiments are sentences /anyoung hasim nika/ and /good morning/. Speech is obtained by sampling rate of 8 kHz. All codebook size is the same 4 dimension 32 level.

The codevector correspondence histogram is obtained by accumulating the vector correspondence along the DTW optimal path. Then mapped codebook is constructed by a linear combination of reference speaker's vectors using each histogram as weighting coefficients.

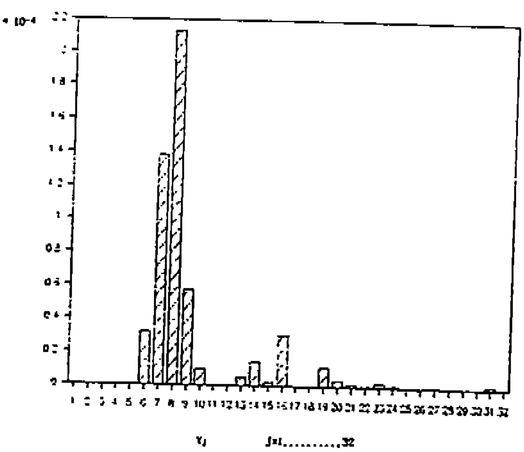
Fig. 5 shows represently the fuzzy VQ-histogram for input vector X_7 and X_{10}. This fuzzy VQ-histogram is obtained by relation input speaker's(male 1) with reference speaker's(female 2) for utterence /anyoung hasim nika/.

Input speaker's codebook and reference's is designed using FLVQ algorithm for obtaining several speaker's FVQ-mapped codebook. For design a precise mapped codebook, the mapped codebook is trained to minimize inter-speaker's distortion.

Fig.6 shows a demonstration of the fuzzy VQ-mapping relations to give a speaker adaptive speech.



(a) input vector X_7



(b) input vector X_{10}

(Fig. 5) Fuzzy VQ-histogram

In experiments, we first measure that the fuzzy-VQ mapping function mapping an input speaker's utterence /anyoung hasim nika/ (male 1, 2, femal 1) speech into reference speaker's(female 2). Then we calculate inverse FVQ-mapping function from it. Using this function, we obtain an utterence /good morning/ containing individuality speech of input speaker(male 1) from his utterence /anyoung hasim nika/ by using a reference speaker's /good morning/.

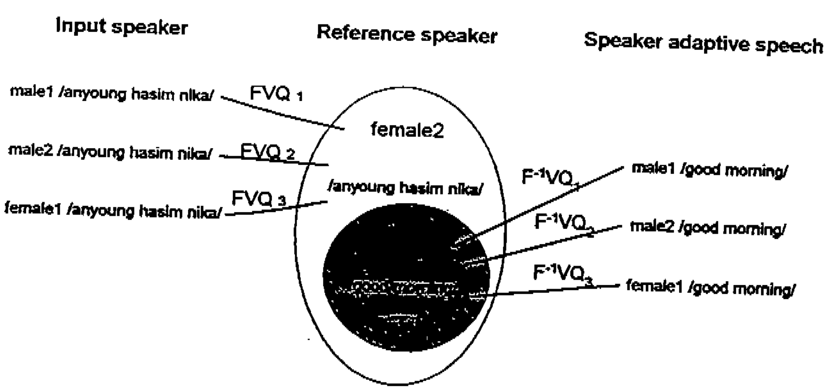
Speaker adaptive speech is synthesized using a mapped codebook and adaptive parameters(fuzzy histogram). The results of experiments of speaker adaptive speech synthesis for male 1's /good morning/ adapted to input speaker's speech (male 1)/anyoung hasim nika/, is shown.

Fig. 7(a), (b) indicate the original speech waveforms of an input speaker's (male 1) and a reference speaker's (female 2) /anyoung hasim nika/, respectively.

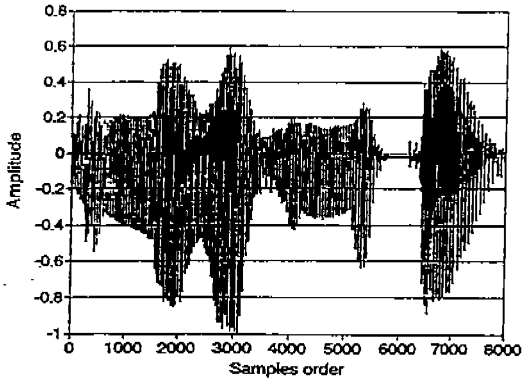
Fig. 8(a), (b) show the spectrograms of an input speaker's (male 1) and a reference speaker's (female 2) /anyoung hasim nika/, respectively.

Fig. 9(a), (b) indicate original speech waveforms of an input speaker's (male 1) and a reference speaker's (female 2) /good morning/, respectively.

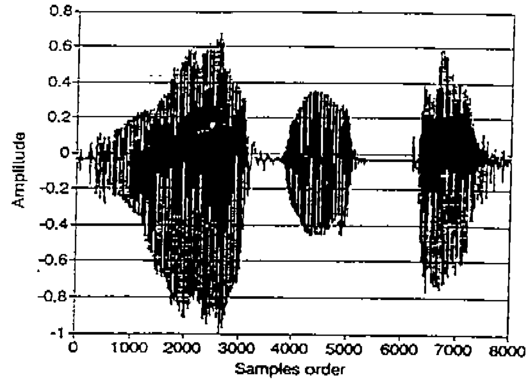
Fig. 10(a), (b) show the spectrograms of an input speaker's (male 1) and a reference speaker's (female 2) /good morning/, respectively.



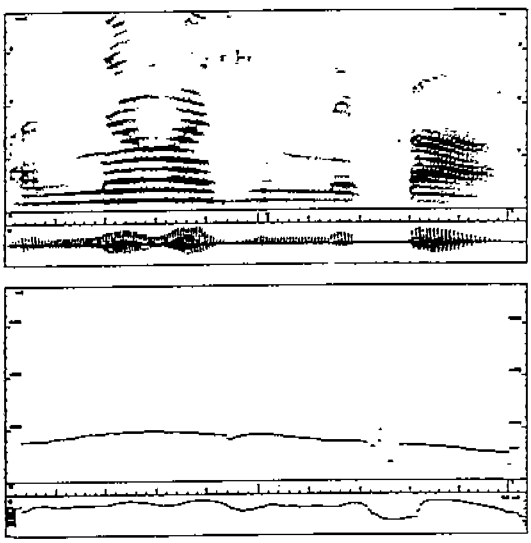
(Fig. 6) Speaker-adaptive speech synthesis using Fuzzy VQ mapping function.



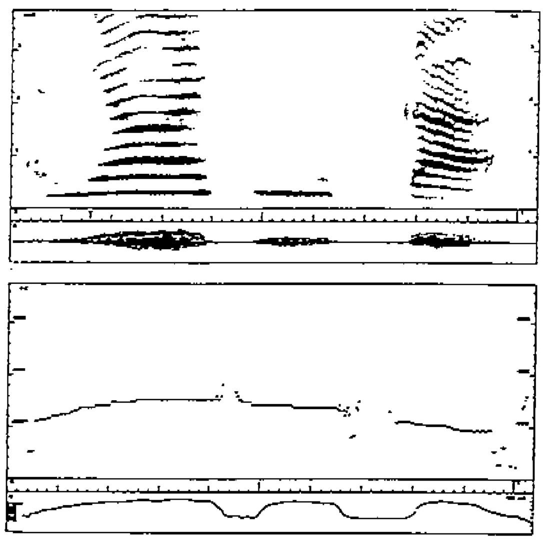
(Fig. 7(a)) Original speech /anyoung hasim nika/ of male 1



(Fig. 7(b)) Original speech /anyoung hasim nika/ of female 2



(Fig. 8(a)) Spectrogram of original speech of male 1 /anyoung hasim nika/

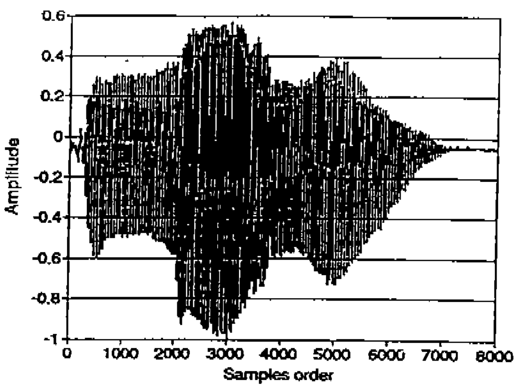


(Fig. 8(b)) Spectrogram of original speech of female 2 /anyoung hasim nika/

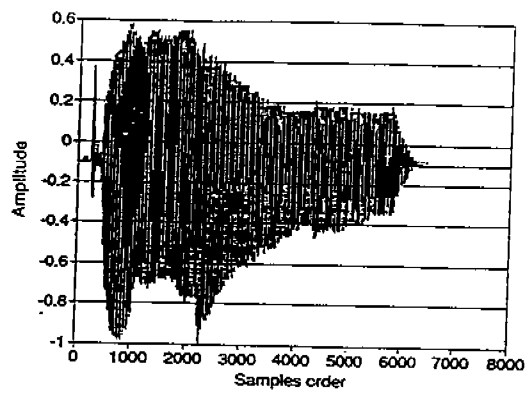
Fig. 11 (a), (b) indicate speech waveform /anyoung hasim nika/ of male 1 converted female 2 by fuzzy-VQ mapping function FVQ and its spectrogram, respectively.

Fig. 12 (a), (b) indicate a speech waveform/good morning/ of female 2 adapted to male 1 by inverse fuzzy VQ-mapping function and its spectrogram, respectively.

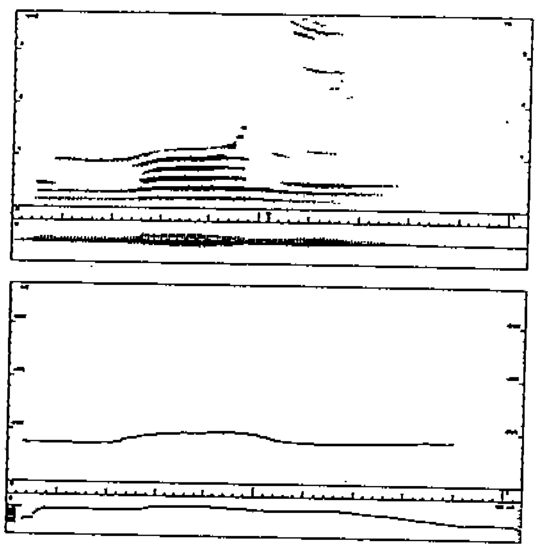
Experimental results show a /good morning/ speech adapted to male 1 /anyoung hasim nika/, so that our attempts for speaker's adaptation in speech synthesis is more or less sucessful. Fig. 11(a) and Fig. 12(b) are shown as input speaker's waveform is mixed with reference speaker's waveform. As codebook size is increased, Fig. 11(a) waveform will closely resemble a Fig. 7(b), and Fig. 12(a) waveform will closely resemble



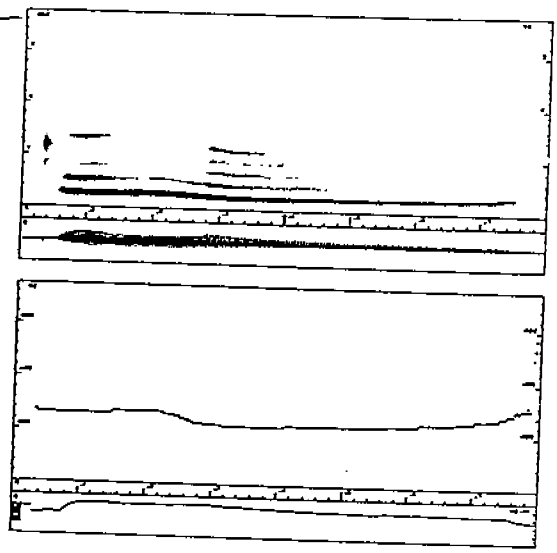
(Fig. 9(a)) Original speech /good morning/ of male 1



(Fig. 9(b)) Original speech /good morning/ of female 2



(Fig. 10(a)) Spectrogram of original speech of male 1 /good morning/



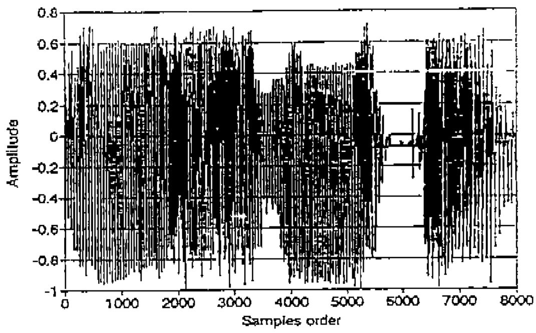
(Fig. 10(b)) Spectrogram of original speech of female 2 /good noming/

Fig 9(a).

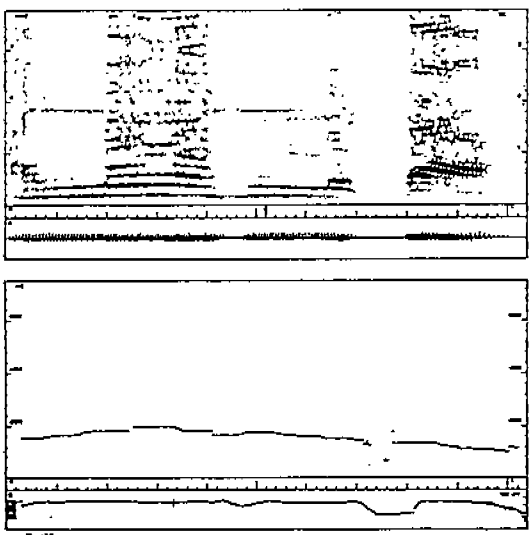
We shows that the synthesized speech by fuzzy VQ-mapping is better than that by VQ-mapping in synthesized speech waveforms. This effect is due to using fuzzy histogram as a weight function when a mapped codebook is defined as a linear combination of reference speaker's speech.

5. Conclusions

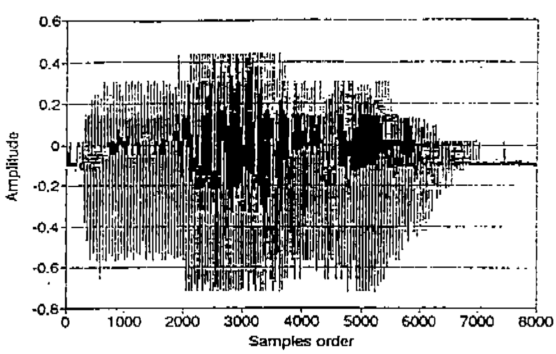
We have presented a speaker-adaptive speech synthesis technique using a mapped codebook designed by Fuzzy VQ mapping and neural net algorithm. The neural net algorithm is employed for learning the membership functions. Using a new supervised competitive learning net called fuzzy LVQ(FLVQ) in which fuzzy learning concept is corporated into the LVQ net, we have designed an input speaker's and a reference speaker's codebook. The FVQ mapping



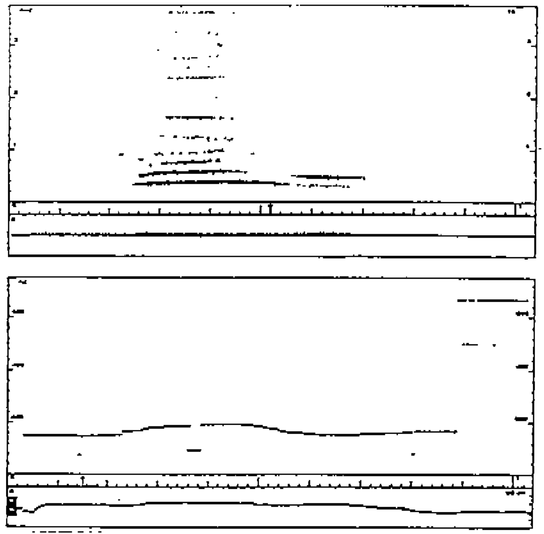
(Fig. 11(a)) Speech waveform/anyoung hasim nika/ of male 1 converted female 2 by fuzzy-VQ mapping function FVQ



(Fig. 11(b)) Spectrogram of converted speech 11(a) : /anyoung hasim nika/



(Fig. 12(a)) Female 2-to-male 1 conversion speech /good morning/ by $F^{-1}VQ$



(Fig. 12(b)) Spectrogram of converted speech 12(a) : /good morning/

method between an input speaker's and a reference speaker's vector space performs through the FVQ mapping function on FLVQ. Speaker-adaptation speech synthesis is carried out through inverse Fuzzy VQ mapping function.

The performance of the proposed method is evaluated subjectively with a degree closed to input speaker's speech. We also showed that individual speech can be controlled for a synthesized speech by using this adaptive techniques.

We will soon employ a back-propagation model to minimize of individuality error in synthesized speech, and further will study adaptive FAM that is transformations of time varying mapping from fuzzy sets to fuzzy sets for adaptive speech synthesis.

REFERENCES

[1] H. P. Tseng, M. J. Sabin, and E. A. Lee, "Fuzzy vector quantization applied to hidden Markov modeling", Proc. ICASSP 87, Paper 15.5.

[2] Lee Ki-Young, "Speaker adaptive word recognition using mapping membership function",

Ph.D. Thesis, Muyng Ji University in korea, 1992.

[3] Lee Jin-Yi, Lee Kwang-Hyung, "The Optimum Fuzzy Vector Quantizer for Speech Synthesis", Fifth IFSA Congress, 1993, pp. 1321-1325.

[4] Kuwabara, H., Takagi, T., "Quality control of speech by modifying formant frequencies and bandwith", 11th Inter. Congress of Phonetic Science, pp. 281-284, August 1987.

[5] Lee Jin-Yi, Lee Kwang-Hyung, "Performance comparison of VQ codebook designed by Fuzzy-C-means algorithm", Prodeedings of JCENANF '93Korea, pp. 413-417, september, 1993.

[6] Fu-Lai Chung and Tong Lee, "Fuzzy Learning Vector Quantization", Proceedings of 1993, International Joint Conference on Neural Networks.

[7] Nakamura, Shikano, "A comparative study of spectral mapping for speaker adaptation", ICASSP, S3.7, 1990.

[8] Stanley C, Ahalt, Ashok K, Krishnamurthy, Prakoon Chen, and Douglas E.Melton, "Competitive Learning Algorithms for Vector Quantization", Neural Networks, Vol. 3, pp. 277-290, 1990.

[9] De Sieno. D., "Adding a conscience to competitive learning". In IEEE International Conference on Neural Networks, pp. 1117-1124, 1988.

[10] Nasrabadi, N.M., & Feng, Y. "Vector quantization of images based upon the Kohonen self-organizing feature maps", IEEE International Conference on Neural Networks, pp 1101-1108, San Diego.

[11] Kiyohiro Shikano, Kai-Fu Lee, Raj Reddy, "Speaker adaption through vector quantizatization", ICASSP, pp. 2643-2646, April 1986.

[12] Kohonen, T. Self-orginization and associative meomry. Berlin: Springer-Verlag.

[13] Naylor. J,. &Li, K. P. "Analysis of a neural network algorithm for vector quantization of speech parameters," Proceedings of the First Annual INNS Meeting, p.310, New York: Pergamon Press.

[14] Matsuyama, Y,. "Vector quantization with optimized grouping and parallel distributed processing", Journal of Neural Networks, 1988.

[15] Gray.R.M. "Vector quantization", IEEE ASSP Magaine, 1(2), pp. 4-29.

[16] Grossberg. S., "Adaptive pattern classification and universal recording: I.Parallel developement and coding neural feature detectors", Biological Cybernetics.

[17] Grossberg. S., "Adaptive pattern classification and universal recording: Ii Feedback, expectation, olfaction, illusions", Biological Cybernetics.

[18] Lippmann, R.R, "An introduction to computing with neural nets", IEEEASSP Maganzine, 4(2), pp. 4-22, 1987.

[19] Winters. J. H., & Rose, "Minimum distance automata in parallel networks for optimum classification", Neural Networks, 2, pp. 127-132, 1989.

[20] Hecht-Nielsen, R., "Applications of counterpropagation networks", Neural Networks 1(2), pp. 131-141, 1988.

[21] Grossberg, S., "Competitive learning; From interactive activation to adaptive resonance", Cognitive Science, 11, pp. 23-63, 1987.

[22] H. J. Zimmermann, Fuzzy set theory and its applications, second edition,Kluwer academic publisers, 1991.

[23] L.Larsen, E.Ruspini, J.McDew, D.Walter, and W.Adey, "A test of sleep staging system in the unrestrained chimpanzee", Brain Res., vol. 40, pp. 319-343, 1972.

[24] J.C.Bezdek, "Feature selection for binary data: Medical diagnosis with fuzzy sets", in Proc. 25th Nat. Computer Conf., 1976, pp. 1057-1068.

[25] J.C.Bezdek,and W.A.Fordon, "Analysis of hyoertensive patients by the use of fuzzy ISODATA algorithm", in Proc. JACC, vol. 3, 1978, pp. 349-356.

[26] J.C.Bezdek, "Fuzzy mathematics in pattern classification", Ph.D. dissertation, Cornell Univ.,

Ithaca, NY, 1973.

[27] Lee Jin-Yi, Lee Kwang-Hyung, "Performance comparison of Speech Synthesis Using Learning VQ in neural networks", Proceedings of ANF'93 Korea, pp. 10-18, May 1993.

[28] Lee Jin-YI, Lee Kwang-Hyung, "A Study on Speaker Adaptation based on Fuzzy Vector Quantizer Mapping and Neural Networks in Speech Synthesis", Proceedings of ICONIP'94 Seoul, pp. 1970-1979.

이 진 이

1985년 숭실대학교 전자공학과 졸업.
1988년 숭실대학교 전자공학과 석사.
1994년 숭실대학교 전자공학과 박사.
1995년~현재 충남산업대학교 전자공학과 전임강사로 재직중.
관심분야:지능 신호처리, 음성 정보처리, 디지탈 통신, ATM망, 퍼지 신경망 응용임.

이 광 형

1968년 서울대학교 공과대학 전자공학과 졸업.
1972년 서울대학교 전자공학과 석사.
1987년 중앙대학교 전자공학과 박사.
1973년~1982년 일본 동경대학 전자공학과 연구원.
1993년~1995년 대한전자공학회 인공지능, 신경망, 퍼지시스템 연구회 위원장.
1994년~현재 국제 전기전자학회(IEEE Korea Council) 학술이사(Conference Committee).
1994년~1995년 대한 퍼지 및 지능 시스템 학회 국제담당이사.
1982년~현재 숭실대학교 전자공학과 교수로 재직 중.
관심분야:지능 신호처리, 퍼지 신경망 음성인식 및 합성, 웨이브렛, 프랙탈 영상 부호화, 멀티미디어 통신처리, ATM 네트워크임.