nition," in *Proc. 6th Workshop Comput. Syst. Tech.*, Sun Moon Lake, Taiwan, Aug. 12-15, 1987, pp. 745-762.

[14] C. H. Wu, C. Y. Tseng, and L. S. Lee, "New recognition techniques for unvoiced Mandarin consonants based upon hidden Markov models," in *Proc. Nat. Comput. Symp.*, Taipei, Taiwan, Dec. 17-18, 1987, pp. 973-979.

[15] P. Y. Tin, C. Y. Tseng, and L. S. Lee, "A Mandarin consonant recognition system based upon time, frequency domain features and finite state vector quantization technique," in *Proc. Nat. Comput. Symp.*, Taipei, Taiwan, Dec. 17-18, 1987, pp. 980-987.

[16] C. C. Lee and P. Y. Huang, "Speaker independent connected Chinese spoken word recognition," in *Proc. 6th Workshop Comput. Syst. Tech.*, Sun Moon Lake, Taiwan, Aug. 12-15, 1987, pp. 721-743.

[17] X. D. Haung, L. H. Cai, D. T. Fang, B. J. Ci, L. Zhou, and L. Jian, "A large vocabulary Chinese speech recognition system," in *Proc. 1987 ICASSP*, Dallas, TX, pp. 1167-1170.

[18] L. Y. Wei, "Spectral energy ratio coding (SERC) of speech," *Taiwan Telecom. Tech. J.*, vol. 4, pp. 265-270, 1984.

[19] E. Zwicker, "Subdivision of audio frequency range in top critical band," *J. Acoust. Soc. Amer.*, vol. 23, p. 248, 1961.

[20] G. Fant, *Speech Sounds and Features*. Cambridge, MA: M.I.T. Press, 1973.

[21] A. Komatsu, A. Ichikawa, K. Nakata, Y. Asakawa, and H. Matsuzaka, "Phoneme recognition in continuous speech," in *Proc. 1982 ICASSP*, pp. 883-886.

[22] Y. Matsuda, S. Tesuka, M. Kanoh, M. Nishimura, and T. Kanedo, "A method for recognizing Japanese monosyllables by using intermediate cumulative distance," presented at 1984 ICASSP, Paper 9.3.1-4.

[23] L. Y. Wei, "Theoretical basis and strategy for machine understanding of Chinese language," *Taiwan Telecom. Tech. J.*, vol. 6, pp. 383-390, 1987.

Comments on "A Two-Stage Representation of DFT and Its Applications"

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***Abstract*—This correspondence contains comments on and several corrections to a recently published TRANSACTIONS paper.**

In the above paper,[1] Ersoy developed a two-stage representation in terms of preprocessing and postprocessing of DFT by vector transformation of sines and cosines into new basis functions using Mobius inversion of number theory. This comment points out first that the inversion Mobius transform pair, (A.3) and (A.4), used[1] are valid only when f is a positive rational number [1, p. 208]. Thus, (A.6) should read

$$X_c(f) = \frac{1}{4f} \sum_{\substack{m=1 \\ \text{odd}}}^{\infty} \frac{\mu_m}{m} \left(\sum_{n=-\infty}^{\infty} \left(2x\left(\frac{n}{mf}\right) - x\left(\frac{n}{2mf}\right) \right) \right) \tag{A.6}$$

and $f > 0$. Second, (2.11) should read

$$n' = 0, 1, \cdots, M_1 - 1. \tag{2.11}$$

This range is very important because it determines the size of the circular correlation in the postprocessing matrix equation. The correctness of this new range has been verified by computer simulation.

Third, the two-stage representation form can be applied directly to the computation of discrete Hartley transform (DHT) [3]. Interestingly, since the elements in the preprocessing matrix of DHT are 0, 1, −1, 2, and −2, only shift, addition, and subtraction operations are involved for the preprocessing stage of DHT. It can be shown that the postprocessing matrix of DHT is also in a block-diagonal form, with each block being a circular correlation matrix. For example, with $N = 8$ and using the same notation defined,[1] the preprocessing and postprocessing matrix equations can be obtained respectively as follows:

$$\begin{pmatrix} h(0) \\ h(4) \\ h(2) \\ h(1) \\ h(5) \\ h(6) \\ h(7) \\ h(3) \end{pmatrix} = \begin{pmatrix} 1 & 1 & 1 & 1 & 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 & 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 & 1 & 1 & -1 & -1 \\ 1 & 2 & 1 & 0 & -1 & -2 & -1 & 0 \\ 1 & -2 & 1 & 0 & -1 & 2 & -1 & 0 \\ 1 & -1 & -1 & 1 & 1 & -1 & -1 & 1 \\ 1 & 0 & -1 & -2 & -1 & 0 & 1 & 2 \\ 1 & 0 & -1 & 2 & -1 & 0 & 1 & -2 \end{pmatrix} \begin{pmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \\ x(4) \\ x(5) \\ x(6) \\ x(7) \end{pmatrix}$$

$$\begin{pmatrix} X(0) \\ X(4) \\ X(2) \\ X(1) \\ X(5) \\ X(6) \\ X(7) \\ X(3) \end{pmatrix} = \begin{pmatrix} 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & b(1, 8) & b(5, 8) & 0 & 0 & 0 \\ 0 & 0 & 0 & b(5, 8) & b(1, 8) & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 1 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & b(1, 8) & b(5, 8) \\ 0 & 0 & 0 & 0 & 0 & 0 & b(5, 8) & b(1, 8) \end{pmatrix} \cdot \begin{pmatrix} h(0) \\ h(4) \\ h(2) \\ h(1) \\ h(5) \\ h(6) \\ h(7) \\ h(3) \end{pmatrix}.$$

Finally, some typing errors[1] are listed below.

1) With the substituting of $l = m_1 n$ modulo N, (2.3) should read

$$h(l) = \sum_{k=0}^{N-1} x(k) \left[\mu\left(\frac{lk}{N} + \frac{1}{4}\right) - j\mu\left(\frac{lk}{N}\right) \right]. \tag{2.3}$$

2) The index i used in (2.7) and (2.12) should be replaced by n'.

3) The term $b(4, 16)$ used in the lowest block of (2.13) should be replaced by $b(9, 16)$.

4) Equation (A.16) should read

$$b(m_1, N) = P(m_1(l), N) - P(m_2(N - l), N). \tag{A.16}$$

REFERENCES

[1] M. R. Shroeder, *Number Theory in Science and Communication*. New York: Springer-Verlag, 1983.

[2] R. N. Bracewell, "The fast Hartley transform," *Proc. IEEE*, vol. 72, pp. 1010-1018, Aug. 1984.

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[1]O. K. Ersoy, *IEEE Trans. Acoust., Speech, Signal Processing*, vol. ASSP-35, pp. 825-831, June 1987.

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