Variable delayed auditory feedback (DAF)

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Program delay provides variable time delay between analog-to-digital converter (ADC) input and digital-to-analog converter (DAC) output using a PDP 8/L computer with 4K memory. The program is written for perceptual experiments that involve delayed auditory feedback. In essence, the program sets up a temporary storage buffer of variable size, depending upon the desired delay time, reads in data through an ADC, and starts outputting data through a DAC when the buffer is filled. Since the memory size available for the buffer is limited to about 3,900 computer words, some trading is necessary to accomplish fairly long delay between sampling rate and delay time. For example, delay time less than 150 msec can be accomplished with approximately 25 kHz sampling rate, while delay time of 500 msec necessitates sampling rate to be reduced to 7.62 kHz. The program uses nominal 12.5 kHz sampling rate, with the range of delay from 0 to 300 msec. Other ranges are available simply by changing content of one location. Through the use of interrupt facility, the program also allows change of delay time during execution as long as the desired delay time is within the given range. This DAF system has advantages over conventional delayed auditory feedback devices (Fairbanks, 1955; Fairbanks & Guttman, 1958) in its availability of much smaller delay time and practically continuous variable delay time.

LANGUAGE

The program is written in assembly language and occupies 72 core locations.

INPUT/OUTPUT

Input to the computer is an analog signal to an A/D converter, and output is an analog signal from a D/A converter with a specified delay.

AVAILABILITY

A binary paper tape and the program listing may be obtained free by request (include a self-addressed, stamped envelope) from the author.

REFERENCES

- Fairbanks, G. Selected vocal effects of delayed auditory feedback. *Journal of Speech & Hearing Disorders*, 1955, 20, 333-346.
Fairbanks, G. & Guttman, N. Effects of delayed auditory feedback upon articulation. *Journal of Speech & Hearing Research*, 1958, 1, 12-22.

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Digital sound spectrograms with simultaneous plotting of intensity and fundamental frequency for speech study

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Sound spectrographs are frequently used for acoustic analysis of speech. Program SSPEC employs the fast

Fourier analysis (Cooley & Tukey, 1965) and the Cepstrum techniques described by Noll (1964) and by Schroeder (1968) in order to generate digital sound spectrograms with simultaneous plottings of intensity level and, if voiced, fundamental frequency for connected discourse.

INPUT/OUTPUT

The user initially specifies three threshold values to be used by the program: a threshold for silence, a threshold for possible voicing, and a threshold for definite voicing. The other input to the program is speech waveform data sampled and digitized 10,000 times per second.

A set of 512 data points (which correspond to 51.2 msec with 10 kHz sampling rate) constitutes a "time segment," and is first analyzed in terms of intensity (calculated from the middle 100 data points in the time segment by taking log of the sum of squares). If the segment has intensity above the given threshold for silence, the 512 values receive the Hamming weighing (Hamming, 1962) and are submitted to fast Fourier analysis. From the returned sine-cosine magnitudes, amplitude levels of the spectral components are calculated and special printing characters are selected according to the energy levels present in the frequency regions. A total of 10 levels of character coding is available to represent various energy levels. The program then calculates a ratio of the energy present in the lower region of this time segment. If the spectral ratio is greater than the given threshold for definite voicing, the fast Fourier analysis is again applied on the logarithmically converted amplitude values (now reduced to 216 points), and a peak-picking technique on the resulting outputs determines the fundamental frequency for this segment. If the spectral ratio is between the threshold of possible voicing and definite voicing, the program provides up to three alternative values for the fundamental frequency. If the overall intensity level of the time segment is below the threshold of possible voicing, however, the second Fourier analysis is bypassed. In any case, the program outputs the results of spectral, intensity, and fundamental frequency analysis for this time segment in a printout similar to regular sound spectrograms, but with simultaneous plottings of intensity and, if voiced, fundamental frequency. The program in the next sequence uses the last 412 data points in the previous time segment, plus the following 100 data points as a new time segment—that is, the time window of 51.2 msec is moved by 10 msec to obtain the new time segment.

PROGRAM LANGUAGE AND COMPUTER

The program is written in FORTRAN IV language except for a subroutine to read digitized data, which is written in assembly language for a CDC 1700 computer.

AVAILABILITY

The program listing is available (free except for postage) on request from the author.

REFERENCES

- Cooley, J. W. & Tukey, J. W. An algorithm for the machine calculation of complex Fourier series. *Mathematics of Computation*, 1965, 19, 297-301.
Hamming, R. W. *Numerical analysis for scientists and engineers*. New York: McGraw-Hill, 1962.
Noll, M. A. Short-time spectrum and 'Cepstrum' techniques for vocal-pitch detection. *Journal of the Acoustical Society of America*, 1964, 36, 296-302.
Schroeder, M. R. Period histogram and product spectrum: New methods for fundamental frequency measurement. *Journal of the Acoustical Society of America*, 1968, 43, 829-834.

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