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# DIGITAL NETWORK DESIGN PROGRAM SYSTEM: DINETS

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## Introduction

This paper describes the design program system of digital filters which is called as DINETS in NEC. DINETS has been developed as the CAD system for the design of digital filters. The program system is organized by three phases of approximation, synthesis and analysis. In this program, some optimum design methods for finite word length, are introduced [1].

## I. Approximation with Infinite Word Length

### (i) IIR filter

The weighted Chebyshev approximation with an arbitrary stop band and pass band characteristics, is carried out in analog frequency domain through pre-warping. The transfer function, can be converted into the Z-plane by one of the three commonly used method: standard, matched or bilinear transformations.

### (ii) FIR filter

Linear phase FIR filters are designed by the Remez exchange method [2]. Minimum phase FIR filters are designed by using the transfer function having quasi-equiripple characteristics in the stop band [3].

An iterative Chebyshev approximation program [4] is also prepared for the filter shaping in the pass band and some other applications.

## II. Approximation with Finite Word Length

Filter coefficients with finite word length to meet arbitrary prescribed characteristics in the frequency domain, are obtained through the following optimization procedure. The initial coefficients which are designed with infinite word length, are rounded and varied by random search method suggested by Suk and Mitra [5]. In this program, the initial coefficients are varied randomly within the upper and lower bounds through next two steps. In the first step, the uniform distribution function is used for the probability distribution function of random number generater. The filter coefficients set which shows the nearest prescribed characteristics, is chosen as the initial coefficients in the second step. In the second step, the normal distribution function is used. In the view point of computing time, that is, the number of transfer function evaluation, the optimization procedure through above two steps, is superior to the conventional one which used the first step or the second step only. 50% reduction of the number of evaluation can be achieved. Fig. 1 shows the loss characteristics optimized through the proposed method and obtained by rounding the coefficients.

## III. Synthesis

Noise powers are calculated for three commonly used filter structures: direct, parallel, and cascade form. In the cascade realization, it is desirable to find a good pairing and ordering by which the minimum round off noise powers can be obtained. One possible approach is a heuristic optimization procedure for finding a "near optimal" solution suggested by Liu and Peled [6]. In this program, a heuristic optimization procedure with some constraints for a pairing and ordering, is used. For the constraint on the pairing, some poles and zeros in the near position are chosen in pairs according to the order of high Q poles. The following two constraints on the ordering, are used for each different configuration in the cascade form.

i) A group of denominaters with low Q poles precedes a group of denominaters with high Q poles, called "high Q last" constraint.

ii) When the individual denominaters are numbered according to the order of low Q poles, a group of odd numbered denominaters precedes a group of even numbered denominaters or vice versa, called as "mixed Q" constraint.

The choice of "high Q last" or "mixed Q" constraint depends on 1D or 2D form and the position of scaling in the cascade form. A heuristic optimization is carried out under these constraints on the pairing and ordering. Therefore, through less number of procedure, near optimal solution can be obtained as shown in Table 1. In Fig. 2, the number of occurrences of various noise gains for different assignments of pairing and ordering is plotted in percentage for two cases, without constraints and with constraints.

In this stage, the one of the structures and the position of scailing are chosen, in which minimum word length of signal and coefficients can be given.

## IV. Analysis

This digital system analysis program handles digital filters with arbitrary topology which is constructed of delays, multipliers and adders. From the topology informations, network equations by which the output of each operation elements are calculated, and the precedence relations can be automatically formulated [7]. The precedence relations contain the output of delay elements, so it is named "expanded precedence relations". The output of each elements are calculated by network equations and expanded precedence relations in infinite word length and in finite word length, and the latter case corresponds to the analysis of the operation of the actual hardware, such as rounding of data, over flow and over flow correction. The transfer function of digital filter can be easily obtained from the impulse response with infinite word length under the condition that the degree of the transfer function is given. The outputs of this program give the time response, frequency response, transfer function, zeros, poles, several kinds of roundoff noise analysis and bit patterns of time response which are used in the inspection process. An example of the analysis for over flow is shown in Fig. 3.

## V. Conclusions

In this paper, the design program system of digital filter is described. This program system includes some optimal design method of digital filter with finite word length. The attention is strongly paid to the saving of computing time through the program. Three important sub-programs, (i) optimization of filter with minimum coefficients word length, (ii) heuristic optimization procedure with some constraints on the ordering and pairing for getting near optimum solution for round-off noise and (iii) analysis program of digital filter with arbitrary topology, are described in detail.

The effectiveness of this program system has been already recognized through the design and manufacturing of a digital signal processing system such as TDM-FDM transmultiplexer [8].

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Table 1

| Form | Position of Scailing | Constraints on Ordering (Note 1) | Number of Procedure (Note 2) LPF 8th order | LPF 12th order |
|---|---|---|---|---|
| 1D | At the Input of Biquad | High Q Last | 25% | 26% |
| | Forward Path | Mixed Q | —— | 7% |
| 2D | At the Input of Biquad | | 31% | —— |
| | Forward Path | | 37% | 16% |

Note 1 The constraint on pairing are held.
Note 2 The number of procedure is expressed in percentage comparing the approach without constraints.

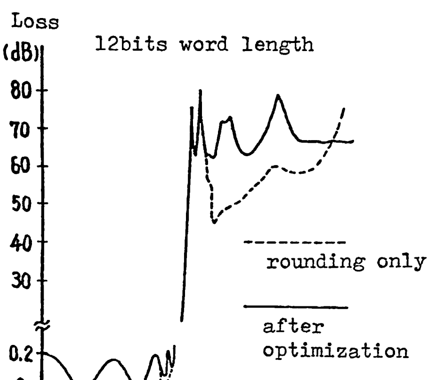


Fig. 1 The comparison between loss characteristics before optimization and after optimization.

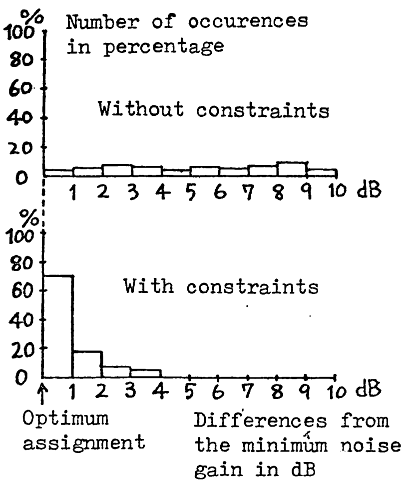


Fig. 2 Number of occurences in percentage.

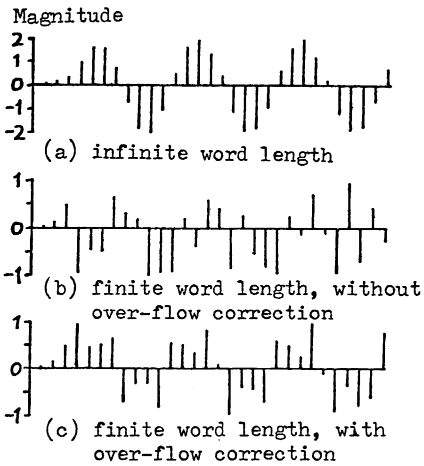


Fig. 3 Output signal for sine wave input signals.