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A Microcomputer Implementation of Real Time, Continuously Programmable Digital Filters

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A MICROCOMPUTER IMPLEMENTATION OF REAL TIME, CONTINUOUSLY PROGRAMMABLE DIGITAL FILTERS

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BY

WILLIAM EDWARD STORMA B.S.E., Florida Technological University, 1978

THESIS

Submitted in partial fulfillment of the requirements for the degree of Master of Science in Engineering in the Graduate Studies Program of the College of Engineering at the University of Central Florida; Orlando, Florida

> Fall Quarter 1979

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BY

WILLIAM E. STORMA

ABSTRACT

When a filter transfer function in s is replaced with the bilinear transform in z, the resulting discrete model represents the original continous model within a second order accuracy of integration . A unique set of recently discovered minimum memory algori thms that perform the bilinear transform on a continuous transfer function are implemented on an INTEL 8080 microprocessor system. Scaling techniques are used to frequency scale all transfer functions to a standardized frequency . All data words are represented in a signed binary double precision format to maintain higher calculation speed and accuracy .

Three test case transfer functions of different order are implemented using the bilinear transform algorithms. First, the algorithms are used to generate the three discrete models. Second, the continuous time models are driven by a step input function , generating a continuous time output. Third, the step function input is discretized and used to drive the bilinear algorithm derived models . Finally, the discrete outputs are compared with the continuous time outputs to validate and evaluate the software techniques used to implement the bilinear algorithms, which imply that the techniques provide a basis for new hardware designs.

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 $\Delta_{\rm{H}}$.

I. INTRODUCTION

Analog circuits and filters designed to process analog signals often are limited in accuracy due to:

a. thermal drift

•

- **b. component tolerances**
- c. offset and bias conditions of operational amplifiers
- d. signal noise introduced by the circuit itself

The only means to build highly accurate analog circuits is through careful design and the use of high quality components. This often results in designing expensive circuits and allowing bench time to minimize circuit sensitivities due to circuit parameters.

The age of digital electronics has brought about many new methods to handle the processing of analog signals. The ability to design signal processing circuits that can handle the signals digitally over comes many of the handicaps of the analog circuits. Digital Signal Processing (D.S.P.) is a newer, more accurate and less **expensi ve means to analyze and process signals. The digital circuits have no thermal drift, no offset or bias problems, do not re**quire high quality circuit components , and do not introduce noise **into the circuits. Thus, many signal processing systems have be**come digital in nature, using analog-to-digital (A/D) and digitalto-analog (D/A) converters to interface between the analog and digi**tal systems .**

The design of digital filters, a special case of D.S.P . , has become a fairly common practice with standardized design procedures.

The use of these standard design procedures involves implementing a filter transfer function in the form of a difference equation. The result of this design 1s a digital circuit that is 'hard wired', 1.e. the characteristics of the circuit are not readily alterable. This feature is unfortunate if the exact characteristics of the fllter are unknown and several designs must be tried before a circuit is chosen.

An alternative to the above problem is the design of a computer software package that allows a real time implementation of a filter transfer function 'in circuit'. Also, giving the software package the ability to alter the filter transfer function while the digital fllter 1s processing signals allows a ' continuous programming ' feature. The result is a real time continuously programmable digital filter. By using an interface capability, the software can be implemented on a microprocessor system and run 'in circuit'. This allows the microprocessor to actually synthesize any filter function and modify the transfer function characteristics while the filter is ' in circuit'.

The basis of this thesis is the implementation of a software package as described above . The software package is designed around a new set of algorithms that perform a bilinear transform using a minimum memory approach. An INTEL 8080/8085 based microprocessor is used to process these bilinear algorithms. The program starts with a transfer function in differential equation (or s domain) form. Then, using a bilinear transform approach, the differential equation is transformed into a difference equation. The program

then exeautes the difference equation in a real time mode, allowing real time output .

The program has memory allocated to operate on transfer functions up to fifth order, using a double-precision (16 bit) data word . **The output from the program is a transient response in time, with the input presently being a step function (though easily modified for any signal input) . A transient response (or time response) is preferred over a frequency response in this case since a step function inputted** in a transfer function forces all filter characteristics to be displayed in the output. The combined features of a digital filter that **is continuously programmable, operates in real time, and can be used** 'in circuit' make this digital filter system highly useful in the **design of digital signal processing systems.**

II. BACKGROUND

Filtering is a technique whereby the frequency spectrum of a signal is specified, such that certain frequencies are passed through the filter and other frequencies are rejected by the filter. Filters are initially designed in the frequency domain (or complex s plane), **where the frequency characteristics can be used to obtain a differential equation. This characteristic filter equation is usually refered to as a transfer function (denoted by H(s)) and 1s a ratio** between the output ($Y(s)$) and the input ($X(s)$). The equation **1s written as:**

$$
\frac{\Upsilon(s)}{\chi(s)} = H(s) \tag{2.1}
$$

and 1s desert bed in the block diagram form as:

$$
X(s) \longrightarrow H(s) \longrightarrow Y(s)
$$

where

•

$$
Y(s) = H(s) X(s)
$$

Once an H(s) is specified, the equation can be transformed into **the time domain , using an inverse Laplace t r ansform:**

$$
\mathcal{L}^{-1}[\quad H(s) \quad] = h(t) \tag{2.2}
$$

The resulting $h(t)$ is an equation of the analog filter characteristics in a continuous time domain. Analog filter design, unlike digital filter design, can be run on an analog computer, which operates **in a continuous time mode . However , with the advent of high speed**

digital computers , a trend has developed to use digital equipment to implement algorithms . The digital computer requires that the algorithms be modified to work in other than a continuous time domain . This is because a digital computer does not run in a continuous time mode I like the analog computer, but in a discrete time mode. This discrete time mode is due to the fact that a digital **computer works in cycle times , and calculations require a certain** number of machine cycles to implement. The result from a digital **computer is a string of outputs at discrete intervals of time .**

It is therefore necessary to transform an H(s) into a discrete **time mode equation. The necessary discrete time mode equation is the difference equation, which 1s implemented in the z domain . The equation is written as:**

$$
\frac{\Upsilon(z)}{\Upsilon(z)} = H(z)
$$
 (2.3)

where $X(z)$ are discrete time inputs and $Y(z)$ are discrete time outputs. The transformation from the z domain to a discrete time mode, **nT , is called the inverse z t ransform , denoted by:**

$$
\mathbf{\mathcal{L}}^{-1}[\text{ H(z)}] = h(n\text{T}) \tag{2.4}
$$

where T is the time sample interval and n is the n^{th} sample period.

Ordinarilly, $H(s)$ models are not transformed directly to $H(z)$ **models . As an example of a textbook approach, the H(s) must first** be transformed into an $h(t)$, then the continuous time, t , must be changed to a sample interval time, nT , and finally the $h(nT)$ must be transformed to an $H(z)$.

Mathematically:

$$
h(t) = \frac{1}{2} \left[H(s) \right]
$$

\n
$$
h(nT) = h(t) \Big|_{t = nT}
$$

\n
$$
H(z) = \frac{1}{2} \left[h(nT) \right]
$$

\n
$$
2.5)
$$

\n
$$
2.6)
$$

\n
$$
2.7)
$$

This and other similar approaches are cumbersome and slow processes for a digital computer to perform. What would be more desireable would be an algorithm that could calculate an $H(z)$ based on an $H(s)$. **This would avoid having to transform into and out of the time domain . This calculation for an s to z conversion would be an approximation** of $H(z)$, based on $H(s)$ and sampling rates.

Although there are computer programs for transforming from the 5 to the z domain , these programs require some amount of memory for all temporary results. Some digital systems posess only a small memory and therefore cannot use the s to z transformation processes. What would be ideal for these digital systems with small memory space would be an accurate algorithm that could approximate an H(z), based on an H(s) and the sampling rate, and perform this algorithm 'in **place' , i .e . using only the memory required for coefficient storage for the algorithm process.**

The specific algorithm to be discussed is based on the bilinear **transfom:**

$$
s = \frac{2}{T} \left(\frac{z-1}{z+1} \right) \tag{2.8}
$$

which is the average of the first order forward difference equation and the first order backward difference equation. This bilinear transform is the standard algorithm used in digital filter design.

The in-place algorithms for equation 2 . 8 were discoverd in 1978 [J] and were published and later modified to handle any general bilinear transformation [2J. The general form of the algorithms are reprinted here for convenience :

the bilinear transform:
$$
s = \frac{az+b}{cz+d} = \frac{\alpha}{z+\beta} + \gamma
$$
; $c \neq 0$ 2.9)

where $\gamma = \frac{a}{c}$ $\beta = \frac{d}{c}$ and the $\frac{2}{T}$ **factor is incorporated into the a,b,c,d variables. given a polynomial in z: Now,**

$$
D(z) = \sum_{i=0}^{N} d_i z^i
$$
 (2.10)

and the bilinear transform (equation 2.9), the polynomial $D(s)$ is found by :

$$
D(s) = \sum_{i=0}^{N} d_i \left(\frac{az+b}{cz+d} \right)^i = \frac{P(z)}{(cz+d)^N}
$$

or

$$
P(z) = \sum_{i=0}^{N} p_i z^{i} = (cz+d)^{N} D(s)
$$
 (2.12)

The problem in getting an 'in place ' algorithm requires computing the p_i 's, the coefficient set of $P(z)$, from the d_i 's, the coefficient set of $D(s)$.

The four step algorithm process for this bilinear transformation is as follows:

substituting 2.9 into 2.12:

$$
P(z) = c^N (z+\beta)^N D \left(\frac{\alpha}{z+\beta} + \gamma \right)
$$
 (2.13)

 E quation 2.14 can be broken down into elementary transforms, which are:

$$
E(z) = D(z+y) \tag{2.14}
$$

$$
H(z) = G(z+\beta) \qquad \qquad 2.17)
$$

Each elementary transform consists of a shift in the z domain of the form:

$$
z = 2z \qquad \qquad 2.18a)
$$

$$
z = z + \beta \tag{2.18b}
$$

$$
z = 1/z \tag{2.18c}
$$

$$
z = z + \gamma \tag{2.18d}
$$

and each of these operations can be applied to polynomials by an 'in place ' operation . *This* **means that any bilinear transform can be applied to polynomials by performing a sequence of 'in place ' operations, such as the general equations of 2 . 18.**

To prove that $H(z) = P(z)$, substitute 2.16 into 2.17, 2.15 into 2. 16 and 2.14 into 2.15 .

$$
H(z) = G(z+\beta)
$$

\n
$$
= (z+\beta)^{N} F\left(\frac{1}{z+\beta}\right)
$$

\n
$$
= (z+\beta)^{N} c^{N} E\left(\frac{\alpha}{z+\beta}\right)
$$

\n
$$
= (z+\beta)^{N} c^{N} D\left(\frac{\alpha}{z+\beta} + \gamma\right)
$$

\n
$$
= P(z)
$$

\n(2.19)

The strategy is to compute first the coefficients of $E(z)$ from the coefficients of $D(s)$, then the f_i 's from the e_i 's, then the g_i 's from the f_i 's and finally the h_i 's from the g_i 's.

From these elementary transforms, a set of computational **equations can be obtained [2J . The final form of these equations are:**

$$
e_j = d_j + \sum_{i=j+1}^{N} \binom{i}{j} \gamma^{i-j} d_i
$$
 (2.20)

$$
f_{\mathbf{1}} = c^N \alpha^{\mathbf{1}} e_{\mathbf{1}}
$$
 (2.21)

$$
\varepsilon_{\mathbf{1}} = f_{N-1} \tag{2.22}
$$

$$
h_{j} = g_{j} + \sum_{i=j+1}^{N} \left(\frac{i}{j}\right) \beta^{i-j} g_{i}
$$
 (2.23)

where

$$
\begin{pmatrix} \mathbf{i} \\ \mathbf{j} \end{pmatrix} = \frac{\mathbf{i}!}{\mathbf{j}! (\mathbf{i} - \mathbf{j})!}
$$

An analysis of these equations will prove that all these operations can be performed I in place' . For the general case of a transfer function in $H(s)$:

$$
H(s) = \frac{\sum_{i=0}^{M} a_i s^i}{\sum_{i=0}^{N} b_i s^i} = \frac{A(s)}{B(s)}
$$
 2.24)

the four step bilinear algorithm would be applied to both the numer**ator and the denominator seperately, with the highest coefficient** order (either M or N) being the order of both the numerator and denominator in $H(z)$. The $H(z)$ would then be written as (assuming M^{th} order):

$$
H(z) = \frac{\sum_{i=0}^{M} c_i z^i}{\sum_{i=0}^{M} d_i z^i} = \frac{C(z)}{D(z)}
$$
 (2.25)

The resulting coefficients of $H(z)$, i.e. the c_i 's and d_i 's, **now occupy the memory locations originally designated for the**

ai 's and bi's, respectively . After obtaining the H(z), an inverse z transform can be applied to transform the equation to the time domain. For the general case:

$$
H(z) = \frac{c_m z^m + c_{m-1} z^{m-1} + \cdots + c_0 z^0}{d_m z^m + d_{m-1} z^{m-1} + \cdots + d_0 z^0}
$$
 (2.26)

which can be rearranged as follows:

$$
X(z)[c_{m}z^{m} + c_{m-1}z^{m-1} + \cdots + c_{0}z^{0}] =
$$

$$
Y(z)[d_{m}z^{m} + d_{m-1}z^{m-1} + \cdots + d_{0}z^{0}] =
$$
 2.27)

Applying **the inverse z transform, the equation becomes :**

$$
c_{m}x(nT+mT) + c_{m-1}x(nT+(m-1)T) + \cdots + c_{0}x(nT) =
$$

$$
d_{m}y(nT+mT) + d_{m-1}y(nT+(m-1)T) + \cdots + d_{0}y(nT) \qquad 2.28)
$$

The inputs ($x(nT+iT)$) and the outputs ($y(nT+iT)$) both depend on values at time t=nT and all future time values (t=nT+T, nT+2T, \cdots). The equation can be converted so that the inputs and outputs depend only on present ($t=nT$) and past values of time ($t=nT-T$, $nT-2T$, ...). This can be accomplished by allowing

$$
n = n-1 \qquad \qquad 2.29)
$$

where n is the nth coefficient. This amounts to a shift in time. **The difference equation now becomes:**

$$
c_m x(nT) + c_{m-1} x(nT-T) + \cdots + c_0 x(nT-mT) =
$$

\n
$$
d_m y(nT) + d_{m-1} y(nT-T) + \cdots + d_0 y(nT-mT)
$$
 (2.30)

The output at present time, $y(nT)$, can be expressed as a function **of the present input and all past inputs and outputs of the equation,** as follows:

$$
d_{m}y(nT) = c_{m}x(nT) + c_{m-1}x(nT-T) + \cdots + c_{0}x(nT-mT) - d_{m-1}y(nT-T) - \cdots - d_{0}y(nT-mT)
$$
 (2.31)

which can be rewritten as:

$$
y(nT) = \frac{\sum_{i=0}^{M} c_{M-i} x(nT-iT) - \sum_{i=1}^{D} d_{M-i} y(nT-iT)}{d_{M}}
$$
 2.32)

The equations necessary to perform a bilinear transformation on an H(s) have been developed. Also, the necessary equations have **been developed that will output a string of values based on a string of input values . What has been derived 1s a set of equations that allows a programmable implementation of a digital filter on a digital computer . By a proper adjustment of the output rate of the string of values from equation 2 .)2, the input-output operation could be performed in a 'real time I mode . By updating the original H(s) equation and allowing the bilinear transform to compute a new** $H(z)$, the digital filter could become ' continuously programmable' **and run in 'real time' .**

The implementation of the above bilinear transform algorithm and a corresponding input- output routine are discussed in the following sections. The implementation is a direct result of the equations developed in this section.

III . DATA FORMAT CONSIDERATIONS

•

Implementation of the bilinear transform algorithm on an 8 bit microcomputer poses some questions as to how the software 1s to handle the program data. The areas of concern in dealing with the data handling problems are:

- a. should the program use fixed point binary or floating point binary?
- b. should the program use single or double precision?
- c. what is the highest order transfer function that can be implemented, with respect to points a and b.

These are the software data handling problems that must be answered before the actual software programs can be written .

The first data handling question concerns the method of representing the data during algebraic manipulations. The use of floating point notation allows data to be described over a wide range of values. Floating point notation has a unique data structure and cannot be represented with a normal 8 or 16 bit data word. Due to the long data word required for floating point notation, execution times for floating point routines are excessively long when compared to analagous routines that are performed in a fixed point notation. Since a requirement in executing these transform algorithms is a rapid execution speed, the use of any floating point notation would cause a considerable increase in the total execution time of

a program, which is a feature that cannot be tolerated in executing these routines. Another disadvantage of using a floating point notation 1s that the number of bits allocated for the data (mantissa) are not the full 16 bits that are used in the double precision fixed point notation. This means that the floating point notation will not carry a full 16 bit accuracy 1n data and therefore 1s less accurate than the fixed point notation in describing data. This factor reinforces the undesireable aspects of using floating point notation .

This leaves the fixed point representation of data to be considered. Using a signed binary notation, data can be ranged over $+127$ for single bit precision and ranged over $+32767$ for double precision. If the sign bit is stored somewhere else than with the data, the double precision data could be ranged over ~65535 . In all cases, all integer values can be accounted for in the fixed point representation. There still exists a problem in describing data that exists in a fractional form or has some part of the data in fractional form $(i.e. 123.78)$, where the .78 is the fractional part). To use data in fractional form, all the data can be scaled to a pure fractional form (i.e. all data ranged ' between -1 and $+1$, excluding endpoints). This can be accomplished by dividing all the data by a value, R, which is greater in magnitude than any of the data, to convert all the data to a fractional form.

The result of scaling all the data to be less than the magnitude of one provides a method of describing all data combinations with a

high degree of accuracy. For a single precision notation, numbers as small as 2^{-8} (3.90625 x 10⁻³) can be described and for double precision notation, numbers as small as 2^{-16} (1.525 x 10⁻⁵) can be described. In both of the above fractional cases, it is assumed that the sign bit 1s carried elsewhere and 1s not part of the 8 or 16 bit data word. Therefore, by properly scaling all of the data to a fractional form, the accuracy of the data can be maintained.

From all the information known about fixed point binary and floating point binary data, and the knowledge that the bilinear transform algorithm requires rapid machine algebraic computations and accurate data handling, one can postulate that the fixed point binary data technique is best. To maintain the high accuracy of the data during the algebraic computations, a 16 bit double precision fractional format is necessary. To maximize the data accuracy, the sign bit of the double precision data word is stored elsewhere than with the data word itself.

Having answered the data handling questions to the first and second areas, there remains the question as to what is the highest order transfer function that can be implemented. With the knowledge that double precision fixed point notation is used, it 1s necessary to determine what is the smallest data word that can be accurately described. Part of this question can be quickly determined by examining the bilinear transform. An examination of equation 2.21, which is:

$$
f_{\mathbf{i}} = c^{\mathbf{N}} \alpha^{\mathbf{i}} \mathbf{e}_{\mathbf{i}} \tag{3.1}
$$

depicts that the α is raised to a power, i, which is directly related

to the order of the e coefficient . For the bilinear transform of

$$
s = \frac{z-1}{z+1}
$$
 (3.2)

with the $\frac{2}{T}$ factor set equal to one, the value of α becomes

$$
\alpha = \frac{b}{c} - \beta \gamma = -1 - 1 = -2 \tag{3.3}
$$

and

 $c = 1$

With this information , equation J.J becomes

$$
f_{\mathbf{i}} = (-2)^{\mathbf{i}} e_{\mathbf{i}}
$$
 (3.4)

For an Nth order system, the e_N coefficient would be multiplied by a $(-2)^N$ value. To insure that the f_N coefficient be less than the magnitude of one, the e_N can be divided by a 2^{N+1} .

There still exists the problem of a data overflow in equations 2 . 20 and 2 . 22 , due to the summations . Since the summed val ue is determined by all the higher order factors and these higher order **factors can range in value between** ~ **1 , there is no absolute factor to d.1 vide al l the data by to insure against an over flow . Therefore, it was necessary to determine a scaling factor based on sample** problems. By inspection of these sample problems and extrapolation of the scaling factors determined for these sample problems, an overall data scaling factor of 2^{2N-1} has been determined for all realizeable filter functions. From the data scaling factor and **the need to maintain some degree of accuracy in the data , an initial** limit on transfer functions has been determined to be fifth order. **Using the double precision fixed point notation, the data would be**

maximally scaled by 2^9 (512), which leaves, at most, seven bits of data that can be retained after the scaling process.

Based on the information presented and the knowledge of the bilinear transform algorithm, filter transfer functions should be no greater than fifth order. This allows sufficient data accuracy for the double precision fixed point binary data format, which is to be used 1n the algebraic computations. The basic questions as to what data handling techniques the software should use have been answered. The next step is to scale the differential equation for use by the bilinear transform.

IV. SCALING THE FILTER FUNCTION

•

Any given fllter transfer function in differential equation form will contain coefficients for each power of s. For any general case, the coefficients will be any real number. These coefficients must be converted to a double precision fixed point fractional binary number before being implemented. Therefore, the transfer function coefficients must all be scaled prior to implementing the bilinear transform algorithm. A generalized scaling technique must be obtained to handle any general transfer function.

Based on a bilinear transform of equation 3.2, a scaling factor of 2^{2N-1} was determined necessary to prevent data overflow during the bilinear transform algorithm. This scaling factor was determined with the $\frac{2}{T}$ factor set equal to one. In general, the $\frac{2}{T}$ factor is not equal to one and must be accounted for. If the $\frac{2}{T}$ factor were to be included in the a, b, c, d of equation 2.9, then equation 3.2 would really be expressed as:

$$
s = \frac{2z - 2}{Tz + T} \tag{4.1}
$$

and the α , β , γ factors would all be influenced by T. Due to this influence by T, the α , β , γ factors would have to be changed every time a different T is chosen. Since the α , β , γ factors must be included in the bilinear transform , the software must be alterable to handle the changes in α , β , γ .

The variations in α would complicate the implementation of equation 3.4 , since raising a number α to a power is not easily **done on a microprocessor. However, raising 2 to a power can be quickly accomplished on binary data by a sequence of shift oper**ations. Therefore, it would be convenient to keep the $(-2)^{1}$ factor in equation 3.4 . It is therefore necessary to scale the **transfer function to redefine the** $\frac{2}{\pi}$ **factor to be equal to one.**

The T factor must first be related to the fllter frequency . Consider a filter with a natural frequency of ω . The period of this filter is then **7**. The T factor is then some fractional part of τ , such that an integral multiple of T will equal τ . This **integral multiple can be defined as x and is called a sample interval .** Now, to obtain $\frac{2}{T}$ = 1, a frequency scaling technique must be incor**porated . Given a sample interval, x, which determines the number of data outputs (from the difference equation) per period,** the original transfer function (at *W*) yields:

$$
\omega = 2\pi f \qquad \qquad \mathcal{T} = \frac{2\pi}{\omega} = \frac{T}{x} \qquad \qquad 4.2)
$$

Therefore:

$$
\frac{2}{T} = \frac{2}{xT} = \frac{2U}{2\pi x}
$$
 (4.3)

Now, consider scaling the frequency to some ω ', such that $\frac{2}{T}$ = 1. **Under these conditions:**

$$
\frac{2}{T} = \frac{2}{xT'} = \frac{2\omega'}{2\pi x}
$$

or

$$
x = \frac{2\omega'}{2\pi}
$$
 (4.5)

To frequency scale from ω to ω' , substitute equation 4.5 into equation 4.1, as shown:

$$
\frac{2\omega}{2\pi x} = \frac{2\omega'}{2\pi} = \frac{\omega}{\omega'}
$$

which can be rewritten as:

$$
\omega' \frac{\omega}{\pi x} = \omega \qquad (4.7)
$$

Equation 4.7 is the factor necessary to frequency scale from ω to ω' . By using this scaling format, the $\frac{2}{T}$ factor will always be set equal to one. For a general polynomial in s, the coefficients are scaled using the formula:

$$
p_{\underline{i}}' = \left(\frac{mx}{\omega}\right)^{N-1} p_{\underline{i}} \qquad (4.8)
$$

For a normalized polynomial, with $\omega > \pi x$, the coefficients of P(s) are scaled down to a fractional value, with the exception of the Nth coefficient, which is one. Once all the polynomial coefficients are in a frequency scaled form, the additional scaling factor of 2^{2N-1} can be performed. The generalized scaling algorithm now becomes:

$$
p_{1}' = \frac{\left(\frac{\pi x}{\omega}\right)^{N-1}}{2^{2N-1}} p_{1} \tag{4.9}
$$

This scaling algorithm insures that all the coefficients are properly scaled to a fractional value and will not overflow during the bilinear transform algorithm process.

V. SOFTWARE IMPlEMENTATION

•

Knowing the necessary equations to perform the bilinear transform (equations $2.20 - 2.23$) and that the data is to be represented **1n a double precision fixed point signed binary format, the actual software programming can be implemented . Knowledge of the** bilinear transform equations only describes the algorithm, but does **not specify hoW' the equations are to be implemented 1n a software** program. These implementation procedures are based on the programmers' **interpretation of the equations and his experience of using a** particular programming language .

Based upon the transfer function 11m1 t of fifth order and the full 16 bit data word, certain initial configurations for memory **storage locations are possible . The data 1s stored as two 8 bit** words with a third 8 bit word storing the sign bit, described **as folloW's :**

with *M.S.B.* denoting most significant byte and L .S .B. denoting least significant byte. Only one bit of the sign byte is used, with the **other bits set to zero . For positive numbers, bit 7 is set to zero and for negative numbers bit 7 1s set to one . Since three memory locations are necessary to fully describe a data word and a fifth order polynomial can have six coefficients (0 - 5), there must be**

eighteen memory storage locations to store all the coefficients of a fifth order polynomial. A filter transfer function could possibly exist as a fifth order numerator over a fifth order denominator, therefore a total of thirty two memory locations are needed to store the coefficients of a transfer function in memory .

Knowing that the bilinear transform 1s to be performed on data 'in place', then once the transform algorithms are executed, the coefficients stored in the memory locations for the transfer function now store the coefficients for the difference equation. The inverse z transform then allows the coefficients of the difference equation to become the coefficients of the discrete time equation. Since every coefficient of a discrete time equation must have a discrete time factor associated with it (i.e. $p(nT-iT)$), there must be six discrete time factors each for the numerator and denominator discrete time equations. The discrete time factors are also described using the double precision fixed point signed binary format that is used on the transfer function coefficients. This requires another thirty two memory locations to store these discrete time factors . On the basis of this requirement for memory, an allocation for memory space was chosen, as shown in figure 1 .

The next step involves implementing the bilinear transform equations (equations $2.20 - 2.23$). One of the first questions is concerned wi th implementing the binomial factor

$$
\begin{pmatrix} \mathbf{i} \\ \mathbf{j} \end{pmatrix} = \frac{\mathbf{i} \mathbf{l}}{\mathbf{j}! (\mathbf{i} - \mathbf{j})!}
$$

Equation 5.1 can either be calculated each time equation 2.20 or

Figure 1. Memory map of data storage

•

2.23 is performed, or a lookup table, based on i and j, could be performed. Knowing that rapid computations are desired and that a factorial computation requires repeated multiplication, which requires an extensive amount of computer computation time , a lookup table would be easier to implement and faster to execute . To implement the lookup table, a means to uniquely describe every i and j combination must be determined. Examination of equations 2.20 and 2.23 show that i is less than j for all cases of i. These restrictions state that some combinations of i and j do not occur in these equations and can be disregarded . A means to determine a number that is unique for all the possible combinations of i and j is to multiply i and j such that

 $K = 1 x j$

 (5.2)

This K value can then be used to locate the position in memory of the proper binomial value. The binomial number can then be retrieved and used in the proper transform equation. The binomial lookup table, based on equations 5.1 and 5.2 , is shown in figure 2. The value of K is added to memory location 6060_{16} to 'point at' the bi nomial value to be retrieved from the table .

To implement the bilinear transform equations ($2.20 - 2.23$), a structured programming method is a desireable choice, both to aid in understanding the flow of the program and to break the transform process into 'blocks' that perform a specific equation on a specific section of data. Equations $2.20 - 2.23$ must be performed on both the numerator and denominator coefficients. Therefore, a software subprogram must be written for each transform equation twice, once

6060₁₆

•

 6074_{16}

Figure 2. Binomial lookup table

for the numerator coefficients and once for the denominator coefficients.

The first equation to be implemented 1s equation 2.20 , which is:

$$
e_j = d_j + \sum_{i=j+1}^{N} \binom{i}{j} \gamma^{i-j} d_i
$$
 (5.3)

Using the bilinear transform of

$$
s = \frac{z-1}{z+1}
$$
 5.4)

with

$$
\frac{2}{T} = 1 \tag{5.5}
$$

the factors α , β , γ become

$$
\begin{array}{l}\n\alpha = -2 \\
\beta = +1 \\
\gamma = +1\n\end{array}
$$
\n5.6a)
\n5.6b)
\n5.6c)

Equation 5.3 reduces to

$$
e_j = d_j + \sum_{i=j+1}^{N} \left(\frac{1}{j}\right) d_i
$$
 (5.7)

A flow chart depicting the implementation of equation 5.7 on the **numerator and denominator coefficients 1s displayed in figures J** and 4, respectively. In both subprograms (XFORM and XFORM2) , the program starts at j=0, evaluates the binomial factor and sums the partial products onto e_j. Once i=N, j is incremented and the **process repeats itself until j=N . The value of N 1s stored in the memory as NUM for the numerator and DEN for the denominator . These values must be placed in memory before the transformation process begins . Once j=N I equation 5 . 7 will have been implemented on all the coefficients and the program moves on to the next**

Figure 3. XFORM flow chart

 \mathcal{H}_α

•

Figure 4. XFORM2 flow chart

•

subprogram.

The next equation to be implemented is equation 2.21, which is:

$$
f_{\mathbf{1}} = e^{\mathbf{N}_{\alpha} \mathbf{1}} \mathbf{e}_{\mathbf{1}} \tag{5.8}
$$

which reduces to:

$$
f_{\pm} = (-2)^{\pm} e_{\pm} \tag{5.9}
$$

Equation 5.9 can be very easily implemented on a microcomputer. Any multiplication by two can be performed by a series of shift operations. A flow chart implementing equation 5 .9 on the numerator and denominator coefficients is shown in figures 5 and 6, respectively . Again, the subprogram ($X2NA$ or $X2NB$) starts with i=0, performs equation 5.9 and then increments i, repeating equation 5.9 until j=N, when the process is finished. The program then proceeds to the next subprogram.

The third equation to be implemented is equation 2.22, which is:

$$
g_1 = f_{N-1} \tag{5.10}
$$

This equation redefines the order of the coefficients. By keeping track of where all the coefficients are for both the numerator and denominator, the reassignment of the coefficients can be handled with software programming. This means that equation 5.10 does not have to be actually performed. This allows a saving of computation time since equation 5.10 is not actually implemented and this helps to reduce the total execution time of the program.

The last equation to be implemented is equation 2.23, which is:

$$
h_{j} = g_{j} + \sum_{i=j+1}^{N} \binom{i}{j} g^{i-j} g_{i}
$$
 (5.11)

•

o.

 $\mathcal{P}_{\mathcal{M}}$

•
This equation can be reduced to :

$$
h_j = g_j + \sum_{i=j+1}^{N} \left(\frac{i}{j}\right) g_i
$$
 (5.12)

since $\beta=1$. Equation 5.12 is identical to equation 5.7 in form, so **the actual programming should be similar. However, equation 5 . 12 must be executed on coefficients that have been reversed in order.** This difference must be accounted for in the subprogram (XFORMJ and XFORM4) . Figure 7 and 8 depict the flow charts of the sub**programs that operate on the numerator and denominator coefficients, respectively.**

Thoughout the subprograms that implement the bilinear transform , certain variables are used to allow the program to know the order of the transfer functions and properly implement the subprograms , These variables are dependent on the order of the transfer function and are obtained by using the following formulas:

> a. NUM = **order of the numerator** b . DEN = **order of the denominator** c . NUMPN = **order** of the numerator multiplied by three d. DENPN = **order** of the **denominator multiplied by three** e . !lUMM1 = **order** of the **numerator plus one** f . DENM1 = **order** of the **denominator plus one**

These variables must be determined and loaded into memory with the **transfer function coefficients before the bilinear transform program can be used.**

After equation 5 . 12 has been performed on the numerator and denominator coefficients, the coefficients that now reside in the memory allocated for the numerator and denominator transfer function coefficients are the coefficients of the difference equation . With the coefficients of the difference equation obtained , a routine

 -1

 ~ 10

Figure 8. XFORM4 flow chart

must be written to output a string of values based on a string of input values (based on the discrete time mode of the difference equation). Since a system response to a step function is a common **method to determine a systems' transient response , a discrete time** step function is used as the input string of values. Knowing that **the input and output values must be fractional numbers, the input values must be limited in value to prevent the output values from overflowing . KnOwing that a realizable transfer functions ' output will never exceed twice the input value, an input value limit** is chosen to be $\frac{1}{2}$ unit.

Having determined the constraints on the difference equation (equation 2.27), which is transformed into the discrete time domain **of equation 2 .)2, a program** can "be **written to evaluate equation 2 .)2 .** Equation 2.32 is restated here as:

$$
y(nT) = \frac{\sum_{i=0}^{m} c_{m-i} x(nT-iT) - \sum_{i=1}^{m} d_{m-i} y(nT-iT)}{d_m}
$$
 5.13)

This equation can be broken down into three simpler equations that can be used to design a structured software program. Equation 5.13 **can be divided into three subprograms:**

- **a . the summation over the x inputs**
- **b. the summation over the y outputs**
- **c . the division over the entire summation to obtain the present time output .**

A flow chart implementing the summation over the i nputs is shown in figure 9. This subprogram (DIFF) performs the discrete **time coefficient by discrete time input factor multiplication and**

Figure 9. DIFF flow chart

sums these partial products into a memory storage location. For the input at nT, the program inserts an input value of $\frac{1}{2}$ into both **the program and the discrete time input factor memory storage location . After the discrete time inputs have all been accounted for in equation 5 . 13, the discrete time output values must be subtracted from the memory location holding the partial summation over the inputs .** This program (STG2) performs the coefficient by discrete time **output factor multiplication , performs a twos' compliment on the product and subtracts the product from the overall summation factor .** Figure 10 depicts the flow chart for this subprogram. Once all **the discrete time output factors have been multiplied and subtracted from the discrete time input factor summation , the present discrete** time output, $y(nT)$, must be evaluated. The program ($STG3$) divides the total summation number by the coefficient d_m to determine the $y(nT)$. The $y(nT)$ is then outputted to an output device for viewing **and recording purposes. This subprogram is flow charted in figure 11 .**

When all the discrete time input and output factors have been evaluated for $t=nT$, the sampling time point must be incremented to t=nT+T. All the discrete time factors must be shifted back in **time by T so that the new sampling time point 1s nT . Since equation 5 . 13 1s deter mined from past and present time values for x and y , an increment in time, T, moves all the x and y values back in time by T. Therefore , all the x and y discrete time factors must be shifted in the memory location to match up with their respective position in time . Figure 12 demonstrates how the x and y values are shifted when the sample time point is incremented .**

)6

Figure 10. STG2 flow chart

Load H,L from TMP3

Load B,C from $[H,L]$

Increment H,L by 2

Load A from $[H,L]$

Load H,L with 607C

Load D,E from $[H,L]$

Determine quotient sign

Stack pointer = 8802

B,C = D,E ÷ B,C

Load H,L with 6045

Store B,C in $[H,L$ device

To MOVE

Figure 11. STG3 flow chart

 \times 10

X.

 \mathbf{v}_R

Figure 12 . Shifting of difference equation time values

TABlE I

Input - output execution time based on order of transfer function

•

TABLE II

Maximum omega that input - output routine can be run in real time

Finally, to ensure that the output string of values occur in a real time mode, any excess execution time must be used up before a new input-output sequence begins. This timing routine must be adjustable based on the updating rate of the output string. The execution time of the input-output routine, based on the order of the transfer function, must be included in the design of the timing routine. Table I displays the execution time of the input-output routine of first to fifth order functions. Based on this information, the maximum frequency that the input-output routine can be operated at, based on the sampling rate, is depicted in Table II.

The end result is a computer program that is capable of performing a bilinear transform algorithm on a differential equation to produce a difference equation. From this difference equation, an output string of discrete time values can be evaluated and produced in a real time mode. By using any transfer function that fits within the constraints of this software program , a real time simulated digital filter can be implemented using this program. With proper interfacing techniques, the software program could actually be used to synthesize a digital filter 'in circuit' in a real time mode.

VI. FILTER IMPLEMENTATION

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Having designed a software program to implement the bilinear transform , several fllter transfer functions have to be tested on the software program to determine the programs' accuracy . The accuracy of the program can be determined by comparing the output from the bilinear transform program with the output determined from the original transfer function, using the same input conditions . By comparing the two outputs, the sensitivity of the program to data format and scaling parameters can be determined. The performance of the program to known transfer functions will help determine the response from any general transfer function.

Filter designs are based on a set of frequency characteristics that are required for a circuit. Therefore, a filter is a frequency selective device. Normally, a test for a filter would involve implementing a frequency spectrum sweep on the filter and observing the output frequency spectrum. However, a digital filter has a different method to be used to check for accuracy . Based on the original transfer function in 5, a continuous time response can be obtained from the analog filter. This continuous time response can then be sampled at intervals of nT (or discretized) to obtain a time sampled response. This response can then be compared to the response from the digital filter, based on the same input, although now discretized. If the digital filter response is accurate, **this output should be the same as the discretlzed response of the** analog filter. Upon this basis, the digital filters are tested **in the time domain and not in the frequency domain.**

Based on the information in Tables I and II, an operating frequency for the test transfer functions is selected to be ω = 10. Sampling rates of .1T and .05T are used for the output rate of the discrete time equation, based on $\omega = 10$. Three transfer functions **are chosen to test the performance of the software program . These transfer functions are:**

> **a . second order low pass b. third order low pass Butterworth** c. third order low pass Chebyshev with 1 dB ripple

These three transfer functions are sufficient to test the software program, testing different types of transfer functions at different system orders .

The transfer function for the second order low pass filter is:

$$
H(s) = \frac{100}{s^2 + 10s + 100} \tag{6.1}
$$

Taking equation 6.1 and allowing $X(s)$ to be a $\frac{1}{2}$ unit step function and then performing an inverse Laplace transform, the resultant **transient response 1s:**

$$
y(t) = .5 - \frac{1}{\sqrt{3}} e^{-5t} \sin \left[10\sqrt{3} + \frac{\pi}{3} \right], t \ge 0 \qquad 6.2
$$

From equation 6.1 , the scaled transfer functions (using equation 4.9) using .17 and .057 sampling rates are determined and shown in Table III. Table III displays the coefficients of the transfer **function in both decimal and hexadecimal form .**

TABLE III

Scaled second order transfer functions

$$
H(s) = \frac{.012337}{.125s^2 + .0392699s + .012337}
$$

 \sim α

 $.17$ sampling rate - decimal format a_{\bullet}

$$
H(s) = \frac{.0328}{.2000s^2 + .0A0Ds + .0328}
$$

 $b.$.1 T sampling rate - hexadecimal format

$$
H(s) = \frac{.003084}{.125s^2 + .0196349s + .003084}
$$

c. . 05T sampling rate - decimal format

$$
H(s) = \frac{.00CA}{.2000s^2 + .0506s + .00CA}
$$

.05T sampling rate - hexadecimal format d_{\circ}

The transfer function for the third order Butterworth low pass filter is:

$$
H(s) = \frac{1000}{s^3 + 20s^2 + 200s + 1000}
$$
 (6.3)

Again using a $\frac{1}{2}$ unit step input and taking the inverse Laplace transform, the transient response of equation 6.3 becomes:

$$
y(t) = .5 - .5e^{-10t} - \frac{5}{\sqrt{75}} e^{-5t} \sin \left[\sqrt{75} t\right] \qquad 6.4
$$

Using equation 6.3, the scaled transfer functions using *. 1"T* **and** *.05 '1* sampling rates are shown in Table IV, in both decimal and hexadecimal **form .**

The transfer function for the third order Chebyshev low pass filter with 1 dB ripple is:

$$
H(s) = \frac{491.3}{s^3 + 9.8834s^2 + 123.84s + 491.3}
$$
 (6.5)

Taking equation 6.5 and allowing a $\frac{1}{2}$ unit step input and then taking an inverse Laplace transform, the transient response becomes:

$$
y(t) = .5 - .5e^{-4.9417t} - \frac{2.47}{\sqrt{93.314}} e^{-2.471t} \sin \left[\sqrt{93.314} t\right]
$$

 $t \ge 0$ 6.6)

From equation 6.5 , the scaled transfer functions using .17 and .057 **sampling rates are shown in Table V, in both decimal and hexadecimal form .**

From equation 6.2 , a plot of the response, $y(t)$, versus time is plotted in figure 13. Along with the transient response, the **outputs from the discrete time functions are also plotted . Similarly , equation 6.4 and the discrete time function outputs are plotted in**

TABlE IV

•

Scaled third order Butterworth transfer functions

$$
H(s) = \frac{.0009689}{.03125s^3 + .0196349s^2 + .0061685s + .0009689}
$$

a. .1T sampling rate - decimal format

$$
H(s) = \frac{.003F}{.0800s^3 + .0506s^2 + .0194s + .003F}
$$

b. . 17' sampling rate - hexadecimal format

$$
H(s) = \frac{.0001211}{.03125s^{3} + .0098174s^{2} + .0015421s + .0001211}
$$

c. .05T sampling rate - hexadecimal format

$$
H(s) = \frac{.0007}{.0800s^{3} + .0283s^{2} + .0065s + .0007}
$$

d. .057 sampling rate - hexadecimal format

 $\gamma_{\rm in}$

TABLE V

29

Scaled third order Chebychev transfer functions

$$
H(s) = \frac{.000476}{.03125s^{3} + .009703s^{2} + .0038195s + .000476}
$$

a. .17 sampling rate - decimal format

$$
H(s) = \frac{.001F}{.0800s^3 + .027Bs^2 + .00FAs + .001F}
$$

b. \therefore 17 sampling rate - hexadecimal format

$$
H(s) = \frac{.0000595}{.03125s^{3} + .0048515s^{2} + .0009548s + .0000595}
$$

c. .05T sampling rate - decimal format

$$
H(s) = \frac{.0003}{.0800s^{3} + .013Ds^{2} + .003Es + .0003}
$$

d. .05T sampling rate - hexadecimal format

S.

figure 14, and equation 6 .6 and the discrete time f unction outputs are plotted in figure 15.

Returning to figure 13, the outputs from the .17 and .05 T **discrete time functions are seen to closely follow the transient response.** The output from the .057 discrete time function 'tracks' **the transient response more accurately than the** *. 1T* **discrete time** function, due to more samples per time period. The steady state value for the transient response is .8000₁₆ (.5000₁₀) and the steady state value for the \cdot **17** discrete time function is \cdot 800B₁₆ ($.5001678_{10}$) and for the $.057$ discrete time function is $.7FD7_{16}$ $($.4993743₁₀). In both cases, the steady state error is less than . 125% for . $7FD7_{16}$ and less than . 033% for . 800B₁₆. Both cases **represent very close approximation to the transient response .**

Figure 14 shows that the .1 T discrete time function accurately follows the transient response, while the .05T discrete time function does not match the transient response characteristics. Both discrete time functions settle down to a steady state value, with the .17 discrete time function having a .7EC9₁₆ (.495254₁₀) value and the .05T discrete time function having a .74BF₁₆ (.456039₁₀) value. The transient response has a steady state value of .8000₁₆ (.5000₁₀). These steady state values represent a steady state error of $.949\%$ for $.7EC9_{16}$ and 8.792% for $.74BF_{16}$.

In figure 15, the .17 discrete time function accurately follows **the transient response , while the** *.05'T* **discrete time function does not match the transient response characteristi cs at all. The** steady state value of the .1 τ discrete time function is .7D10₁₆

(.488525₁₀) and for the .057 discrete time function is .60EB₁₆ $(.378585₁₀)$. The transient response has a steady state error of 2.295% for .7D10₁₆ and 24.283% for .60EB₁₆.

In all the discrete time functions, there exists a larger steady state error for the .057 sampling rate than for the .17 **sampling rate . An examination of the scaled transfer functions** for .1T and .05T sampling rates in hexadecimal format (parts b and d in Tables III, IV, V) show that the number of non-zero bits in the coefficient drops by as much as three bits as the sampling rate **increases from** *. 1T to .OST.* **For the higher order systems , this** leaves only two or three non-zero bits for the zero order coefficients. The result of the coefficients being rounded off and expressed in such small, truncated numbers is that these coefficients produce round off errors when shifted, added and multiplied by the bilinear **t ransform . As long as there are sufficient bits 1n the coefficients to retain data accuracy , the bil inear transform closel y matches the transient response (as in the .17 case, all transfer functions).** Once the data accuracy is lost, due to insufficient bits, the **bilinear transform is using truncated data words , and the output** from the discrete time equation is a poor approximation of the **transient response . The r esult is a tradeoff between sampling rate and data accuracy; data accuracy diminishes with higher sampling** rates and the output is inaccurate. With a low sampling rate, the **output has less than 1% error for two of the functions tested at** . 17 and less than 2.3% error for the Chebychev function at .17.

VII. RESULTS AND CONCLUSIONS

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The software program that 1s implemented in this thesis is basically two separate programs linked together. One program performs a bilinear transform on a transfer function to generate coefficients for a difference equation. The other program actually performs an input-output operation on the coefficients of the difference equation. Through the implementation of both of these software modules as one larger program, the performance of these programs can be evaluated. By evaluating these performance characteristics, the benefits/disadvantages of the programs are revealed .

The software program written to implement the bilinear transform algorithm was designed around the need to calculate the data as quickly as possible. To help increase calculation speed, a fixed point notation was used to represent the data. Double precision notation was needed to insure adequate word length during the calculations. To insure that the data was represented accurately, the data was scaled to a fractional form. To reduce calculation time on the bilinear transform equations, a scaling factor was designed to frequency scale the transfer function to a standardized frequency, based on the sampling rate of the discrete time equation. All these techniques were used in writing the bilinear transform algorithm software program .

Based on the results of filter transform functions implemented

in Chapter VI, the bilinear transform software program results in an output error less than 2.3% when the data is accurately represented (minimwn of last 7 of 16 bits are non zero or contain data information) . When the 16 bit data word truncates the value of the real coefficients, the bilinear transform can provide an output error greater than 8% of the real transient response. The truncation of data occurs when the sampling rate is increased, causing the scaling factor to decrease the values of the transfer functions' coefficients. From this knowledge, there are several solutions to retain data accuracy with increasing sampling speed. Among these ideas include:

- a. using a 16 bit microprocessor with double precision (32 bit) word length
- b. using a different scaling technique
- c. using a floating point notation
- d. developing new equations to implement the bilinear transform •

Using a 32 bit word would increase the data accuracy, until high sampling rates are needed, where the data would again be truncated. A different scaling technique could imply rewriting the algorithms, possibly slowing down execution time. Floating point notation would allow a wide range of data values, but would slow down execution time . Other new algorithm equations are not yet developed to execute the bilinear transform with minimum memory. There appears to be no single best solution to this problem. Using any alternate approach that will not drastically increase execution time can be considered a feasi ble solution.

From the information supplied in Tables I and II, the maximum operating frequency of the program is limited by the input-output

routine. An analysis of the input-output program reveals that a major amount of execution time 1s spent in software multiply and divide routines. The data acquisition and add routines are presently using minimal execution time based on the 16 bit data word . An improvement 1n this program would be the implementation of a hardware or firmware multiply/divide routine to decrease the execution time . By decreasing execution time , the maximum frequency obtainable 1s increased . Since the input- output routine is a very straightforward process, the algorithms need not be modified. The execution time can be reduced by using hardware or firmware multiply and divide routines .

The algorithms designed to perform an 'ln place ' operation , based on the bilinear transform, can result in output errors less than 2 . J% on a microprocessor system . Based on sampled outputs from the bilinear transform program versus outputs from the original transfer function, the program data matches the theoretical data within a 2.3% error provided that the last 7 bits of the 16 bit data word contain data information. Faster input-output operations can be obtained by substituting a hardware or firmware multiply/divide routine for the present soft**ware routine . With these modifications , a sufficiently powerful realtime digital filter can be designed around a small memory mi croproc**essor, with continuously programmable features that make this system **extremely attractive for digital filter design implementation. Furthermore , the generalized procedure for the second order accuracy bilinear approach implies a search for similar higher order accuracy algorithms** that could be beneficial to state of the art digital filter design.

APPENDIX A

INTEL 8080 Assembly Program Listing

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RENSO : F1 .FILTER SRC SYNBOLS XREF

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And

ISIS-II 6868/8865 MACRO RSSEMBLER, V2 8 XFORM PAGE 2 S TO Z TRANSFORM ALGORITHMS LOC OBJ SEQ SOURCE STATEMENT 6899 CD0362 36 CALL ADM : ADD MPY # TO LOW COEFFICIENT 609C 3R5R60 37 LDA NUM : CHECK FOR HIGHEST CORFFICIENT 689F B9 38 CMP C 68A8 C28968 39 JNZ CONT : LOOP BACK IF NOT DONE 6983 84 40 INR B **6084 48** 41 MOV C B : C GETS B 42 LXI H TABLE 60R5 210060 : H. L CETS 6000H **A GETS B** 60R8 78 43 MOV A.B 60R9 2C 44 INCR: INR L ; SET UP NEW COEFFICIEN" POINTER 60RR 2C 45 INR L 60AB 2C 46 INR L 60AC 3D 47 DCR A ; A GETS A - 1 60AD C28960 48 JNZ INCR : UPDATE COEFFICIENT POINTER 60B0 3R5A60 49 LDA NUM : A GETS NUMERATOR ORDER **6883 BS** 50 CMP B 51 JNZ CNTR 60B4 C28660 ; LOOP BACK IF ALL COEFFICIENTS 52 ; NOT DONE 6887 818988 53 XFORM2: LXI B.80H : B. C SET TO ZERO 54 LXI H TRELEMENT : H, L GETS 6012H
55 CNTR2: SHLD TMP1 : STORE H, L IN TMP1
56 CONT2: LXI D, BINOM : D, E GETS 6060H 68BR 211268 6060 222460 60C0 116060 68C3 8C 57 INR C ; A CETS B X C
: SET BINOMIAL POINTER 60C4 CDEA61 **58 CALL MULT** 60C7 83 59 ADD E 68 HOV E.R 6008 SF 60C9 2C 61 INR L 60CR 2C 62 INR L 600B 2C 63 INR L 60CC 18 : A GETS BINOMIAL NUMBER 64 LDAX D 60CD CD5362 ; MULTIPLY BIN. # BY COEFFICIENT 65 CALL MPY 6000 CD0362 ; ADD MPY # TO LOW COEFFICIENT 66 CALL ADM 6803 3A5B68 67 LDA DEN ; CHECK FOR HIGHEST COEFFICIENT 6006 B9 68 CMP C 6007 C2C060 69 JNZ CONT2 ; LOOP BACK IF NOT DONE 68DA B4 70 INR B I C GETS B 68DB 48 71 MOV C.B : H.L GETS 6012H 60DC 211260 72 LXI H TABLE+12H 68DF 78 73 MOV R.B. 74 INCR2: INR L 60E0 2C : SET UP NEW COEFFICIENT POINTER 60E1 2C 75 INR L 60E2 2C 76 INR L **GRE3 3D** 77 DCR A 68E4 C2E868 : UPDATE COEFFICIENT POINTER 78 JNZ INCR2 60E7 385B60 79 LDA DEN ; A GETS DENOMINATOR ORDER 6BEA BB SB CHP R 60EB C28D68 B1 JNZ CNTR2 : LOOP BACK IF ALL COEFFICIENTS 82 : NOT DONE : E GETS 1 60EE 0601 83 X2MH: MVI 8.1 68F0 3R5A60 84 LDA NUM : A GETS NUMERATOR ORDER 60F3 B8 85 CMP B 60F4 CR2E61 ; FREAK OUT IF NO COEFFICIENT AFFECTED 86 JZ X2NB 68F7 218368 : H.L CETS 6003H 87 LXI H, TRBLE+3 GBFA BEBB 88 RPT: MVI C.0 **GBFC 56** 89 MOV D. M ; LOAD COEFFICIENT INTO D.E 60FD 2C 98 INR L

 -1.8

 $\mathcal{L}_{\mathcal{A}}$

ISIS-II 8080/8085 MACRO ASSEMBLER, V2 0 NFORM PAGE 3 S TO Z TRANSFORM ALGORITHMS LOC 08J SEQ SOURCE STATEMENT **GREE SE** 91 HOY E.H 60FF 2C 92 INR L ; LOAD COEFFICIENT SIGN IN A 6100 7E 93 MOV R.M 6101 322660 94 STR THP2 ; STORE A IN TMP2 : SET CARRY BIT 6104 37 95 SHFT: STC ; CLEAR CARRY BIT
; MULTIPLY COEFFICIENT BY 2 6105 3F 96 CHC 6106 7B 97 HOY R.E 6197 17 98 RHL 6108 SF 99 MOV E.R. 6109 7R 100 MOV 8.0 610A 17 101 RAL 6108 57 182 MOY D. A : COMPLIMENT SIGN BIT 610C 3R2660 103 LDA THP2 610F 2F 104 CMR 6110 322660 105 STR TMP2 6113 BC 106 INR C : A GETS B 187 MOV A B 6114 78 6115 89 108 CMP C ; CONTINUE MULTIPLY IF NOT DONE 6116 020461 109 JNZ SHFT ; CLEAN UP SIGN BIT 6119 3R2668 110 LDR THP2 611C E688 111 ANI SOH ; RESTORE COEFFICIENT IN MEMORY 611E 77 112 MOV M.A. 611F 2D 113 DCR L 6120 73 114 MOV M.E. 6121 2D 115 DCR L 6122 72 116 HOY M.D ; INCREMENT POINTER TO NEW COEFFICIENT 6123 2C 117 INR L 6124 2C 118 INR L 6125 2C 119 INR L 6126 84 120 INR B ; A GETS TOP OF STORAGE STACK NUMBER - 6127 3R5E60 121 LDA NUMM 6128 BB 122 CMP B ; REPEAT SHIFT IF MEW COEFFICIENT AVAILABLE 612B C2FR60 123 JNZ RPT 124 ; B GETS 1 612E 8681 125 X2NB: MVI B.1 : H.L GETS 6015H 6130 211560 126 LXI H TABLE+15H 127 RPT2: MVI C. 8 6133 BEBB ; LOAD COEFFICIENT INTO D.E 6135 56 128 MOV D.M 129 INR L 6136 2C 6137 5E 139 MOV E.M. 6138 2C 131 INR L 6139 7E 132 HOV A.M. ; STORE SIGN BIT IN TMP2 613R 322668 133 STA TNP2 6130 37 134 SHFT2: STC ; SET CARRY ; CLEAR CARRY 613E 3F 135 CMC ; MULTIPLY COEFFICIENT BY 2 613F 7B 136 MOY R.E 6140 17 137 RAL 138 MOV E.A 6141 5F 6142 7R 139 MOV R.D 6143 17 148 RAL 6144 57 141 MOV D.A : COMPLIMENT SIGN BIT 6145 3R2660 142 LDA TMP2 6148 2F 143 CMR

1.4

6149 322660

614C 0C

144 STR THP2

145 INR C

ISIS-II 8888/8885 MACRO ASSEMBLER, V2.8 XFORM PAGE 5 S TO 2 TRANSFORM ALGORITHMS LOC OBJ SEQ SOURCE STATEMENT **H.L GETS 6012H** 61RB 211268 201 LXI H TABLE+12H 61AE 385D60 ; SET H.L TO POINT AT LOW COEFFICIENT 202 LDR DENPN 61B1 85 203 ADD L 61B2 6F 204 HOY L R 285 CNTR4: SHLD THP1 ; STORE H, L IN THP1
286 CONT4: LXI D.BINOM ; D, E GETS 6060H 61B3 222460 61B6 116060 61B9 AC 207 INR C ; A GETS B X C 61BA CDEA61 208 CALL HULT 61BD 83 209 ADD E : SET BINOMIAL POINTER **GIRE SE** 210 MOY E.A. 61BF 2D 211 DCR L 61C0 2D 212 DCR L 61C1 2D 213 DCR L : A GETS BINOMIAL NUMBER 61C2 1R 214 LDAX D ; MPY BIN. # BY COEFFICIENT 61C3 CD5362 215 CALL MPY : ADD MPY # TO LOW COEFFICIENT
: CHECK FOR HIGHEST COEFFICIENT 216 CALL ADM 6106 CD0362 6109 385860 61CC 89 218 CMP C ; LOOP BACK IF NOT DONE 61CD C2B661 219 JNZ CONT4 61D0 04 228 INR B ; C GETS B 61D1 48 221 MOV C B : H.L GETS 6012H
: SET H.L TO POINT AT NEW COEFFICIENT 222 LXI H TABLE+12H 6102 211268 61D5 385068 223 LDA DENPN 6108 85 224 ADD L 6109 FF 225 MOV L A 61DA 78 226 MOV A.B 6108 20 227 INCR4: DCR L 61DC 2D 228 DCR L 6100 20 229 DCR L 61DE 30 238 DCR A 61DF C20B61 61E2 3R5B60 61E5 88 233 CMP B ; LOOP BACK IF ALL COEFFICIENTS NOT DONE 61E6 C2B361 234 JNZ CNTR4 235 : END OF PROGRAM 61E9 76 **236 HLT** : MULTIPLY ROUTINE FOR BINOMIAL POINTER 237 ; PUSH REGISTERS ON STACK 238 HULT: PUSH B 61EA C5 61EB D5 239 PUSH D 61EC E5 248 PUSH H : A GETS B 61ED 78 241 HOV R. B **61EE FEBB** 242 CPI 8 $: TEST FOR B = 0$ 61F0 CAFE61 243 JZ ONE **A GETS O** 244 HVI R 8 61F3 3E00 ; ADD B TO A C TIMES 61F5 80 245 MLTY: RDD B 61F6 00 246 DCR C : CHECK IF $C = 0$ 61F7 C2F561 247 JNZ MLTY ; FOP REGISTERS OFF STACK 61FR E1 248 FINAL: POP H 61FB D1 249 POP D 61FC C1 250 POP B 61FD 09 ; END SUBROUTINE 251 RET 61FE 3E01 252 ONE: MVI A.1 : A GETS 1 6200 C3FA61 253 JMP FINAL ; ADD ROUTINE 254 255 HOM: PUSH H | PUSH ON STACK 6283 E5

ISIS-II 8090/8085 MACRO ASSEMBLER, V2 0 XFORM PAGE 6 S TO Z TRANSFORM ALGORITHMS LOC 083 SEQ SOURCE STATEMENT 6204 05 256 PUSH D 6205 CS 257 PUSH B 6296 F5 258 PUSH PSN 6207 2R2460 259 LHLD TMP1 ; H.L GET LOW COEFFICIENT ADDRESS ; D.E GET LOW COEFFICIENT 620A 56 268 HOY D. M 6208 23 261 INX H 620C 5E 262 MOY E.M. 6200 23 263 INX H 620E 7E 264 HOV A. M. : A GETS SIGN BIT 628F 47 265 MOY B.A 266 CPI 80H : 2'S COMPLIMENT IF NEGATIVE 6210 FE80 6212 CA3D62 267 JZ COMP 6215 282760 268 SET1: LHLD MPYL : H.L GET MPY NUMBER 6218 3R2968 269 LDA NPYS : A CETS SIGN BIT 621B 4F 278 MOY C.R 621C FE80 271 CPI 80H ; 2'S COMPLIMENT IF NEGATIVE 621E CR4562 272 JZ COMP2 273 SET2: MOV R.B 6221 78 : CHECK SIGNS OF BOTH NUMBERS 6222 R9 274 XRR C 6223 47 275 MOV B.A 276 MVI A 0 6224 3E00 6226 19 ; ADD D, E TO H, L 277 DAD D ; PUT SIGN BIT IN A 6227 1F 278 RAR 6228 R8 : FIGURE SIGN OF SUM 279 XRA B 6229 17 288 RAL 622R DR4B62 : 2'S COMPLIMENT IF NEGATIVE 281 JC COMP3 6220 3E88 282 MYI 8.8 622F EB 283 INSRT: XCHG ; SWAP D, E WITH H, L : H.L GET LOW COEFFICIENT ADDRESS ; STORE COEFFICIENT IN MEMORY 6238 282468 284 LHLD TIP1 6233 72 285 MOV M.D 6234 23 286 INX H 6235 73 287 HOV MLE 6236 23 288 INX H 6237 77 289 MOV M.A. 6238 Fi 298 POP PSW : POP REGISTERS OFF STACK 6239 C1 291 POP B 623R D1 292 POP D 623B E1 293 POP H 623C C9 294 RET : END SUBROUTINE 295 6230 EB 296 COMP: XCHG : SWAP D.E WITH H.L ; 2'S COMPLIMENT ROUTINE 623E CD6F62 297 CALL MOD1 ; SWAP D, E WITH H.L 6241 EB 298 XCHG 6242 C31562 299 JMP SET1 388 6245 CD6F62 301 COMP2: CALL MOD1 6248 032162 302 JMP SET2 624B CD6F62 303 COMP3: CALL MOD1 624E 3E80 304 MVI A 80H 6250 C32F62 305 JMP INSRT : M'LTIPLY ROUTINE 386 6253 05 387 HPY: PUSH B ; FUSH REGISTERS ON STACK 6254 05 388 PUSH D 6255 E5 389 PUSH H 6256 F5 310 PUSH PSW

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ISIS-II 8080/8085 MACRO RSSEMBLER, V2 0 XFORM PAGE 7 S TO Z TRANSFORM ALGORITHMS LOG $0BJ$ SEQ SOURCE STATEMENT 6257 56 311 MOV D.M ; D.E GET COEFFICIENT 6258 23 312 INX H 6259 SE 313 MOV E.M. 625A 23 314 INX H 625B 7E 315 MOV A.M 316 STR MPYS ; SIGN BIT STORED IN MPY5 625C 322960 625F F1 317 POP PSW $H, L = 0$ 6260 210000 318 LXI H 00H ; ADD D,E TO H.L 6263 19 319 MPY2: DAD D 6264 30 320 DCR A $\begin{array}{ll} \texttt{1} & \texttt{CONTINUE} & \texttt{TO ADD IF A} \neq 0 \\ \texttt{1} & \texttt{H,L} & \texttt{STORBD IN MPYL} \end{array}$ 6265 026362 321 JHZ MPY2 322 SHLD MPYL 6268 222760 ; POP REGISTERS OFF STACK 626B E1 323 POP H 626C D1 324 POP D 6260 C1 325 POP B ; END SUBROUTINE 626E 09 326 RET 327 328 MOD1: PUSH D I PUSH D ON STACK 626F D5 6278 70 329 HOY R.H. : 1'S COMPLEMENT H.L 6271 2F 330 CMR 6272 67 331 MOV H.A. 6273 7D 332 MOV R.L 6274 2F 333 CMR 6275 GF 334 MOV L A $; D, E = 1$ 335 MVI D.8 6276 1600 6278 1E01 336 MVI E.1 ; ADD D, E TO H, L , POP OFF STACK 6278 19 337 DAD D 338 POP D 627B D1 : END SUBROUTINE 339 RET 627C C9 348 341 END

PUBLIC SYMBOLS

V.

EXTERNAL SYMBOLS

ASSEMBLY CONPLETE NO ERRORS

CROSS REFERENCE COMPLETE

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RSH80 DIFFEQ SRC SYMBOLS XREF

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ISIS-II 8888/8865 MACRO ASSEMBLER, V2.8 DIFFEQ PRGE 2 DIFF. EQN. OUTPUT ROUTINE LOC OBJ SEG SOURCE STATEMENT 6346 7D 53 MOV AL ; CHECK FOR PRESENT TIME FACTOR 6347 FE38 54 CPI 30H 6349 CA7563 55 JZ INPX ; INPUT X FACTOR IF TIME IS RIGHT : LOAD B.C WITH INPUT FACTOR 6340 46 56 HOV B. M 634D 2C 57 INR L 634E 4E 58 MOV C.M 634F 2C 59 INR L 6350 7E 60 MOV A.M 6351 227868 61 RERTE: SHLD ADL ; STORE H.L IN ADL 6354 C5 62 PUSH B 6355 47 63 MOV B.A ; ADJUST PRODUCT SIGN BIT 6356 3R2960 64 LDA MPYS 6359 A8 65 XRR B 635R 322968 66 STR HPYS 6350 C1 67 POP B 635E CDD764 68 CALL MPYDB ; MULTIPLY COEFFICIENT BY X FACTOR 6361 222760 69 SHLD MPYL 6364 CD0362 78 CALL RDM : ADD MPY # TO SUMMATION 6367 287868 71 LHLD RDL ; LOAD H, L FROM ADL 636A 2D 72 DCR L ; L GETS L - 5 636B 2D 73 DCR L 636C 2D 74 DCR L 6360 2D 75 DCR L 636E 2D 76 DCR L 636F 227868 ; STORE H.L IN ADL 77 SHLD ADL 6372 C32A63 78 JMP BACK ; RETURN TO NEW INPUT 79 6375 8688 80 INPX: MVI B.89H : SET B.C = $BCOOH$ 6377 BEBB 81 MVI C 0 6379 3E88 82 HVI A.8 ; SET $A = 0$ 637B 70 83 HOY M.B ; STORE A.B.C IN X FACTOR TABLE 637C 2C 84 INR L 6370 71 85 MOV M.C 637E 2C 86 INR L 637F 77 87 HOV M. A. 6380 035163 ; RETURN TO INFUT ROUTINE 88 JMP RERTE 89 6383 214568 98 STG2:LXI H YCOEFF ; H, L GETS 6045H 6386 3R5C60 91 LDA NUMPN : SET H.L TO POINT AT OLDEST TIME 6389 85 ; FACTOR 92 ADD L 6388 6F 93 MOV L R 6388 227868 94 SHLD ADL ; STORE H.L IN ADL 638E 211268 95 LXI H TABLE+12H ; H.L GETS 8012H 96 LDA NUMPN 6391 3R5C60 ; SET H.L TO POINT AT CORRESPONDING 6394 85 97 ADD L ; COEFFICIENT 6395 6F 98 MOV L A 6396 227660 99 SHLD THP3 ; STORE H, L IN TMP3 188 BROK2: LHLD THP3 6399 287660 ; LOAD H.L FROM TMP3 639C 7D 101 MOV R.L 6390 FE12 182 CPT 12H : CHECK IF ALL COEFFICIENTS USED 639F CRF663 183 JZ 5TG3 6382 56 184 MOV D. M ; LOAD D.E WITH COEFFICIENT 63R3 2C 105 INR L 63R4 5E 186 HOV E.M. 63R5 2C 187 INR L

ISIS-II 8080/8085 MACRO ASSEMBLER, V2 0 0IFFEQ PAGE 3 DIFF. EQN. OUTPUT ROUTINE LOC OBJ SEQ SOURCE STATEMENT 63A6 7E 188 MOY R. H ; A GETS SIGN BIT 63R7 322960 109 STA MPYS 63RR 2D 110 DCR L ; L GETS $L - 5$ 63AB 2D 111 DCR L 112 DCR L 63RC 2D 63RD 2D 113 DCR L 63RE 2D 114 DCR L ; STORE H, L IN TMP3 63RF 227660 115 SHLD THP3 ; LOAD H, L FROM ADL 63B2 2R7860 116 LHLD ADL ; B, C GETS OUTPUT FACTOR 63B5 46 117 MOV B. M 63B6 2C 118 INR L 63B7 4E 119 MOV C. M 63B8 2C 120 INR L 121 MOV R. M 6389 7E 63BR 227860 122 SHLD RDL ; STORE H.L IN ADL 63BD CS 123 PUSH B 63BE 47 124 MOV B.A ; ADJUST SIGN OF PRODUCT 63BF 3R2960 125 LDA MPYS 63C2 R8 126 XRA B 6303 2F 127 CMR 63C4 E680 128 ANI 88H 6306 322960 129 STR HPYS 6309 C1 130 POP B ; MULTIPLY COEFFICIENT BY Y FACTOR 63CA COD764 131 CALL MPYDB 63CD 222768 132 SHLD MPYL ; CHECK FOR ZERO PRODUCT 63D0 C3E463 133 JMP CHK ; ADD TO SUMMATION 63D3 CD0362 134 CONT: CALL ADM 6306 287868 135 RRTE2: LHLD ADL : L GETS L - 5 6309 20 136 DCR L 63DA 2D 137 DCR L 63DB 2D 138 DCR L 63DC 2D 139 DCR L 6300.20 148 DCR L ; STORE H, L IN ADL 63DE 227868 141 SHLD RDL 63E1 039963 142 JMP BACK2 ; GO FOR NEW COEFFICIENT 143 63E4 70 144 CHK: MOV R.L. : CHECK H.L FOR ZERO NUMBER 63E5 FE00 145 CPI 0 63E7 C2F363 146 JNZ PRSS 63EA 7C 147 HOV A.H 63EB FE00 148 CPT 8 63ED C2F363 149 JNZ PRSS 63F8 C3D663 150 JMP RRTE2 151 PASS: JMP CONT 63F3 C3D363 152 153 63F6 287668 154 STG3: LHLD THP3 ; LOAD H, L FROM TMP3 , B.C GETS Y OUTPUT COEFFICIENT 63F9 46 155 HOV B. M 63FA 2C 156 INR L 63FB 4E 157 MOV C. M 63FC 2C 158 INR L 63FD 7E 159 HOV A. M 63FE C5 160 PUSH B ; IOAD H.L WITH SUMMATION # LOCATION 63FF 217068 161 LXI H ZXH ; LOAD D, E WITH SUMMATION NUMBER 6402 56 162 MOV D. M

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ISIS-II 8888/8885 MRCRO RSSEMBLER, V2.8 DIFFEQ PRGE 4 DIFF. EQN. OUTPUT ROUTINE SEG SOURCE STATEMENT LOC OBJ 6403 2C 163 INR L 6404 SE 164 HOV E.H. 6405 2C 165 INR L 6486 46 ; ADJ'ST SICN BIT 166 MOV B.M 6487 R8 167 XRR B 6488 324768 168 STA YCOEFF+2 : STORE SIGN BIT 648B C1 169 POP B 640C 310288 170 LXI SP, 8802H ; SET STACK POINTER 648F CDF664 171 CALL DIVOB ; DIVIDE D, E BY B, C : H.L GETS 6045H 172 LXI H YCOEFF 6412 214560 6415 70 173 MOV M.B ; STORE QUOTIENT IN MEMORY 6416 2C 174 INR L 6417 71 175 MOV M.C 6418 CS 176 PUSH B 6419 78 177 MOV R.B ; OUTFUT QUOTIENT TO OUTFUT DEVICE 6418 CDB264 178 CALL HEX1 6410 C1 179 POP B 641E 79 180 MOV A.C 641F CDB264 181 CALL HEX1 6422 3R4760 182 LDA YCOEFF+2 6425 CDB264 183 CALL HEX1 ; CUTPUT CAPRIAGE PETURN 6428 0E8D 184 HVI C 8DH 642R CDBFF8 185 CALL OUT1 642D BEBR 186 MVI C. 86H : OUTPUT LINE FEED 642F CDOFF8 187 CALL OUT1 188 189 6432 814568 198 HOVE: LXI B. YCOEFF ; H. L CETS 6045H 6435 3A5C68 191 LDA NUMPN ; SET H, L TO POINT AT OLDEST TIME FACTOR 6438 C602 192 ADI 02H 643A 81 193 ADD C 643B 4F 194 HOV C.R 643C 3C 195 INR A 643D 3C 196 INR A 643E 3C 197 INR A 643F 5F 198 MOV E R 6448 58 199 MOV D.B 6441 0A 200 SHFTDN: LDRX B ; MOVE ALL TIME FACTORS DOWN TO 6442 12 201 STRX D ; NEXT TIME FACTOR SLOT 6443 80 202 DCR C 6444 1D 203 DCR E 6445 78 284 MOV R.E. 6446 FE2F 205 CPI 02FH 6448 CR4E64 206 JZ TIMER ; JUMP TO TIMER WHEN FINISHED 6448 C34164 207 JMP SHFTDN 288 209 218 644E 218880 211 TIMER: LXI H, 8880H 212 THOUT: DCX H 6451 2B 6452 7D 213 MOV R.L 6453 FE00 214 CPI 8 6455 025164 215 JNZ THOUT 6458 7C 216 MOV A.H 6459 FE00 217 CPI 0

ISIS-II 8000/8085 MACRO ASSEMBLER, V2 0 DIFFEQ PAGE 5 DIFF. EQN. OUTPUT ROUTINE LOC OBJ SEQ SOURCE STATEMENT 645B C25164 218 JNZ TMOUT 645E C30063 219 JMP DIFF 6858 228 DS 58H 221 222 ; END OF PROGRAM 223 64B1 76 224 HLT 225 / OUTPUT HEX NUMBERS 226 HEX1: PUSH PSN 64B2 F5 ; PUSH REGISTER ON STACK 64B3 1F 227 RAR ; RIGHT SHIFT 4 TIMES 64B4 1F **228 RAR** 64B5 1F 229 RAR 64B6 1F **230 RAR** 64B7 E60F 231 ANI 8FH ; A GETS A LOGICAL AND OFH 64B9 C630 232 ADI 30H ; A GETS A + 30H 64BB FE3R 233 CPI 3RH 64BD FRC264 ; CHECK IF A LESS THAN 3AH 234 JM OUT2 64CB C687 235 ADI 87H ; ADD 7 IF NOT 64C2 4F 236 OUT2: MOY C.A 64C3 CDBFF8 237 CALL OUT1 ; OUTPUT MOST SIGNIFICANT PART 6406 F1 238 POP PSW ; POP REGISTER OFF STACK 64C7 E60F : A GETS A LOGICAL AND OHF 239 ANI OFH 64C9 C63B 240 ADI 30H : A CETS A + 30H 64CB FE3R ; CHECK IF A LESS THAN 3AH 241 CPI 3RH 64CD FRD264 242 JH GUT3 6400 C607 ; ADD 7 IF NOT 243 ADI 07H 64D2 4F 244 OUT3: MOV C.A. 6403 CDOFF8 ; CUTPUT LEAST SIGNIFICANT PART 245 CALL OUT1 6406 C9 : END OF SUBROUTINE 246 RET 247 248 249 / MULTIPLY/DIVIDE SUBROUTINE 258 251 MPYDB: LXI H 00 6407 210000 ; H.L SET TO ZERO 6408 3E10 252 MVI A.16 $A = 16$ 640C F5 253 MPY2: PUSH PSW 640D 7B 254 MOV R.E : CHECK IF LSB IS ZERO 640E E601 255 ANI 01H 64E0 CAE464 256 JZ MPY1 64E3 89 257 DAD B ; ADD B.C TO H.L IF LSB # ZERO 64E4 7C 258 MPY1: MOV A.H ; RIGHT SHIFT H.L AND D.E 64E5 1F 259 RAR 64E6 67 268 MOV H. A. 64E7 7D 261 MOV R.L 64E8 1F 262 RAR 64E9 6F 263 MW L R 64ER 7R 264 MOV R.D 64EB 1F 265 RAR 64EC 57 266 MOV D. A 64ED 7B 267 MOV A.E 64EE 1F 268 RHR **GAFF SF** 269 MOV E.A. 64Fe F1 270 POP PSH 64F1 3D 271 DCR A $; A GETS A - 1$ 64F2 C2DC64 ; CONTINUE MULTIPLY IF A \neq 0 272 JNZ MPY2

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ISIS-II 8888/8885 MHCRO ASSEMBLER, V2.8 DIFFEQ PAGE 6 DIFF. EQN. OUTPUT ROUTINE LOC OBJ SEQ SOURCE STATEMENT 64F5 C9 273 RET : END SUBROUTINE 274 ; CARRY SET 64F6 37 275 DIVDB: STC 64F7 3F 276 CMC ; CARRY CLEARED 64F8 7R ; RICHT SHIFT D.E 277 MOV A.D. 64F9 1F 278 RFR 279 MOV D. A 64FR 57 64FB 7B 288 MOV A.E 64FC 1F **281 RAR** 64FD SF 282 MOV E.R ; SET CARRY
; CLEAR CARRY 64FE 37 283 DIVD1: STC 64FF 3F 284 CMC ; RIGHT SHIFT AND 1'S COMPLIMENT B, C 6500 78 285 HOV R.B 6501 1F 286 RRR 6502 2F 287 CHR 288 MOV B.A 6583 47 6584 79 289 MOV A.C 6505 1F 290 RRR 6506 2F 291 CMR 6507 4F 292 MOV C. A 6588 83 293 INX B $:$ B GETS $B + 1$; H.L SET TO ZERO 6509 210000 294 LXI H 80 650C 3E11 295 MVI A.17 $; A = 16$; SWAP D, E WITH H, L 650E EB 296 XCHG 297 DV0: PUSH H 658F ES ; ADD B, C TO H, L 6518 89 298 DAD B ; CHECK FOR CARRY BIT 6511 DA1565 299 JC DV1 ; RESTORE OLD H, L IF NO CARRY 6514 E1 300 POP H 6515 F5 381 DV1: PUSH PSN 6516 7B 382 MOV R.E. ; LEFT SHIFT D, E AND H, L 6517 17 383 RAL 6518 SF 304 MOY E.R 6519 7R 385 MOV R.D. 6518 17 306 RAL 651B 57 387 MOV D. A 651C 7D 388 MOV R.L 651D 17 389 RAL 651E 6F 318 MOV L R 651F 7C 311 MOV R.H. 6528 17 312 RAL 6521 67 313 MOV H R 6522 F1 314 POP PSW 6523 30 315 DCR A ; A GETS A - 1 ; CONTINUE DIVISION IF A \neq 0 6524 C20F65 316 JNZ DV0 ; PLACE D.E IN B.C 6527 78 317 MOV R.D 6528 47 318 MOY B. A 6529 7B 319 MOV R.E 652A 4F 320 MOV C.A ; SET STACK POINTER 652B 318088 321 LXI SP, 8800H : END SUBROUTINE 652E C9 322 RET 727 324 325 326 END

USER SYNBOLS

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RSSEMBLY COMPLETE, NO ERRORS

REFERENCES

1. Close, C.M.; DeRusso, P.M.; and Roy, R.J. State Variables For Engineers. New York: John Wiley and Sons, 1965.

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- $2.$ Cohen, D., and Simons, F.O., Jr. "An In-Place Algorithm for Computing the Bilinear Transform of Polynomials." Unpublished research paper, California: University of Southern California / Information Sciences Institute, 1978.
- $3.$ Harden, R.C., and Simons, F.O., Jr. "Differential Equation Solutions For Up to 10th Order System Theory Models With H.P. - 67 Compulators." Unpublished research paper, Florida: University of Central Florida, 1978.
- 4. INTEL 8080 Assembly Language Programming Manual. California: INTEL Corp., 1976.
- $5.$ Stanley, W.D. Digital Signal Processing. Virginia: Reston Publishing Company, Inc., 1975.
- $6.$ Wavell, R.B. "Microcomputers: An Alternative for Digital Controllers." Unpublished Masters thesis, University of Central Florida, 1979.