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Final Technical Report
July 1981

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MICROWAVE TRANSVERSAL EQUALIZER OPEN LOOP ADAPTIVE COMPUTER CONTROL TECHNIQUES

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20. ABSTRACT (Continue on reverse side if necessary and identify by block number)

Adaptive equalization techniques, as well as suitable algorithms and computer programming procedures, have been developed to provide open loop control of the Microwave Transversal Equalizer. The resultant computer analysis will determine the necessary MTE control adjustments to minimize time sidelobe distortion.

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SUMMARY

This final report on Contract F30602-80-C-0042 summarizes the AIL investigation to determine and demonstrate suitable algorithms and corresponding computer software in order to provide open-loop adaptive control of the Microwave Transversal Equalizer (MTE) previously developed by AIL under Contract F3062-78-C-0352.

The program task objectives have been successfully accomplished by AIL with the application of Fast Fourier Transform (FFT) techniques to provide an algorithm programmed on the RADC HP 2100A computer. This procedure will ultimately be utilized to determine the MTE amplitude and time delay adjustments necessary to equalize the distortion introduced by an arbitrary network in series with the equalizer.

Verification of the developed FFT and associated software program was accomplished at AIL with a DEC-20 computer using artificially generated simulated time sidelobe distortion typical of the expected range of levels. Translation of the algorithm from the DEC-20 computer format to the HP 2100A computer format has been completed, and verification of the software program using RADC I and Q data is underway. Preliminary data suggests that the program translation task has been successfully accomplished by AIL. It is expected that optimal data acquisition techniques and/or preferred computer interface procedures will be defined as a result of related post-delivery activities at RADC.

ACKNOWLEDGEMENTS

The authors wish to express their appreciation for the technical direction of J. Polniaszek and W. Peterson of RADC, Griffiss Air Force Base, Rome, New York.

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1. INTRODUCTION

The objective of Contract F30602-80-C-0042 is to develop an algorithm, and relevant computer software programming for the RADC HP 2100A computer, to facilitate open-loop control of the MTE in order to minimize time sidelobe distortion. The resultant algorithm using FFT techniques and associated programming has been validated with artificially generated simulated distortion data using the AIL DEC-20 computer and the HP 2100A computer at RADC. A similar post-delivery validation effort will be conducted by RADC personnel using real-time distortion data and the HP 2100A computer.

Section 2.0 will contain a brief discussion of MTE operational fundamentals in order to establish an inter-relationship with the algorithm development program. Section 3.0 will present a detailed treatment of the FFT techniques applied to algorithm development for the MTE. Section 4.0 will discuss the resultant computer program including applicable operating and test procedures. Finally, Section 5.0 will present conclusions and recommendations for a logical continuation of effort required to provide closed-loop operation of the MTE.

2. MTE TECHNICAL DISCUSSION

A discussion of distortion considerations and the fundamentals of the MTE operation are presented in order to provide the technical basis for computer control and adaptive equalization of resultant time sidelobe distortion. A detailed description of the MTE, including operating procedures, is included in Final Report RADC-TR-80-121 of 31 January 1980 titled "Microwave Transversal Equalizer".

2.1 PAIRED ECHO THEORY AND DISTORTION CONSIDERATIONS

Algorithms for determining the settings of attenuation and phase for each loop of a transversal equalizer rely heavily on the paired echo concept. Accordingly, a brief review of this theory is given below stressing its applicability to the present program.

2.1.1 Paired Echo Theory

The paired echo concept and its application to the design of a transveral equalizer (MTE) was described in Ref. 1. It should be noted that this work was supported by RADC. This paper also gives a bibliography of prior work on non-microwave MTE's and the now classic reference on Chirp radar by Klander, et al. The latter emphasized the use of paired echo theory to predict time sidelobe levels in Chirp radars. The essential aspects of the paired echo theory are summarized below.

Ref. 1 - J.J. Taub and G.P. Kurpis, "Microwave Transversal Equalizer", Microwave Journal, 1969.

Signals which can be characterized by a time function of its Fourier transform (amplitude and phase vs frequency functions), are subject to distortion when propagating through a microwave component or a system of microwave components (such as a Chirp radar). This distortion occurs because the system's transfer function possesses neither perfectly constant gain nor perfectly linear phase over the spectrum of the signal.

We typically represent the transfer function in the frequency domain and need to predict its effect on time domain distortion. By using the paired theory we can make relatively quick conversions between the frequency domain and the time domain and vice versa.

The system transfer function, or frequency response, of an aribtrary system can be defined as:

$$H(\omega) = A(\omega)\ e^{jB(\omega)} \tag{2-1}$$

where A and B are respective gain and phase functions defined in Reference 1.

We must now assume that the signal transmitted through the system has spectral components that are band limited. This is a safe assumption for most systems. For example, in many radar applications the signal's spectrum is confined to the electronic bandwidth of the final power amplifier. Within a band limit of ω_ℓ to ω_h we can rigorously represent the A and B functions as Fourier series:

$$A(\omega) = a_o \left[1 + \sum_{n=1}^{N} \frac{a_1}{a_o} \cos (n\omega' e) + \emptyset_n \right] \qquad (2\text{-}2)$$

and

$$B(\omega) = b_o \omega' + \sum_{n=1}^{N} b_n \sin (n\omega' e + \psi_n) + K \qquad (2\text{-}3)$$

where:

$$\omega' = \omega - \omega_o$$

$$\omega_o = \frac{\omega_\ell + \omega_h}{2}$$

$$e = \frac{2\pi}{\omega_o - \omega_\ell}$$

In a distortionless network $A(\omega) = a_o$ and $B(\omega) = b_o \omega'$ Hence the remaining terms contribute to distortion. This representation leads to the time domain response including distortion and so that for a band limited signal $V_i(\omega)$ and its time function equivalent $V_i(t)$ we obtain

$$V_o(t) = a_o V_i (T + b_o) + a_o \sum_{n=1}^{N} E_{n+} V_i (t + b_o - ne)$$
$$+ a_o \sum_{n=1}^{N} E_{n-} V_i (t + b_o + ne) \qquad (2\text{-}4)$$

where $V_o(t)$ is the output time function and the constants E_{n+} and E_{n-} represent echo or time sidelobe amplitude levels. These levels are related to the Fourier distortion coefficients in equations (2-2) and (2-3). For the case where $\emptyset_o$ and $\psi_o = 0$ they reduce to

$$E_{n+} = \frac{1}{2}\left(\frac{a_n}{a_o} + b_n\right) \qquad (2\text{-}5)$$

$$E_{n-} = \frac{1}{2}\left(\frac{a_n}{a_o} - b_n\right)$$

A typical example of a distorted microwave pulse is given in Figure 2-1. The echoes are clearly displayed. This analysis desmonstrates that a frequency domain transfer function can be represented by a Fourier series which enables rapid calculation of the time domain response. Conversely, a measurement of the time domain response which yields sidelobe levels can be used to rapidly calculate the a and b coefficients thereby yielding the frequency domain transfer function. The MTE settings cancel distortion by injecting equal and opposite echoes in cascade with the system.

2.1.2 Use of Paired Echo Theory

Since the MTE design and adjustment procedures are based on cancelling distortion echoes, algorithms can be developed by taking either frequency domain (I and Q data) measurements or time domain responses (time sidelobe levels in dB) and converting them into the necessary MTE loop attenuation and phase settings to produce cancelling echoes. The key point of this discussion is that use of paired echo theory simplifies computation and therefore significantly reduces computer time. Furthermore it provides the flexibility to work with either frequency or time domain data. In this program the actual procedure used was to use I and Q data (frequency domain measurements).

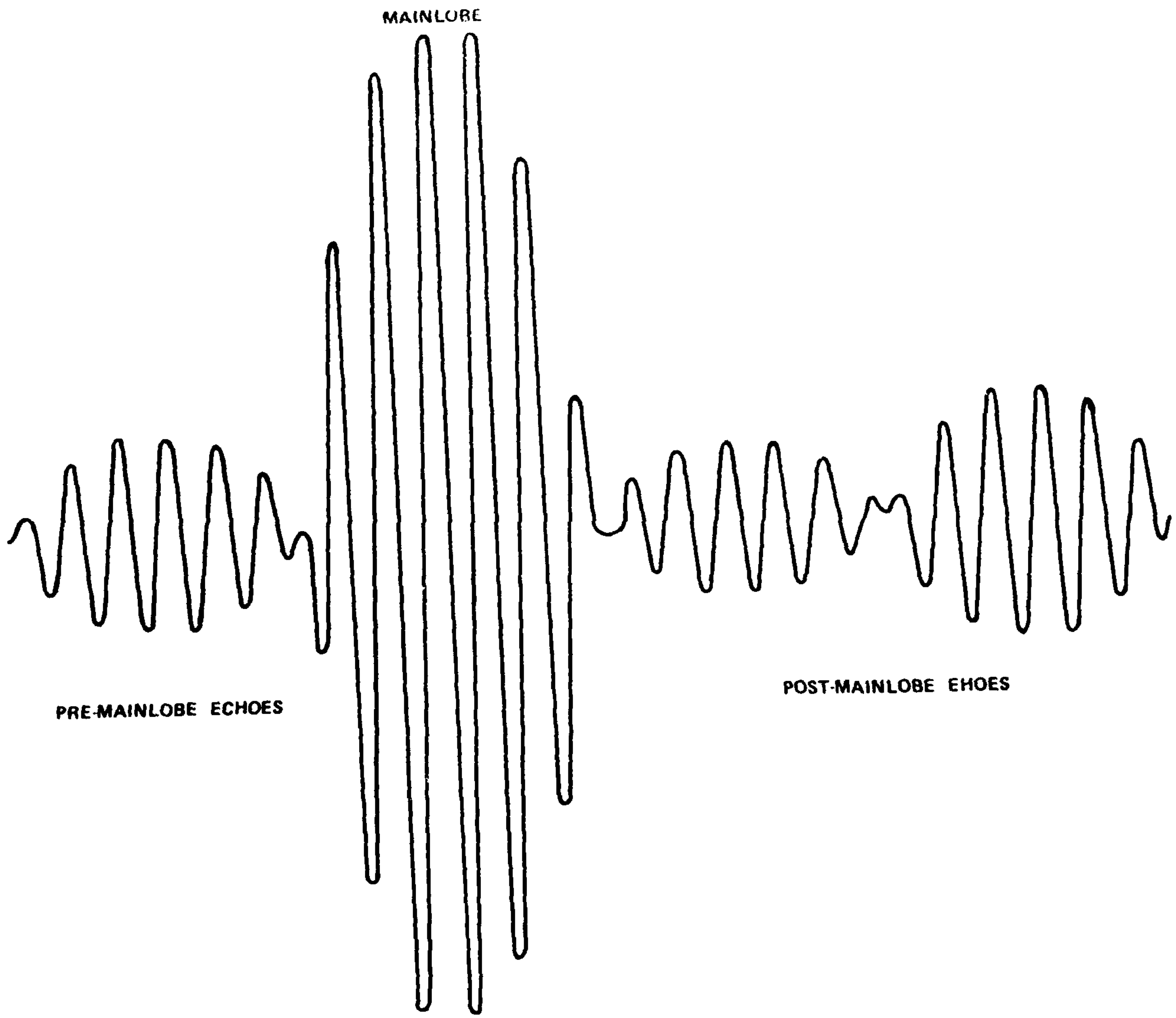


Figure 2-1. Time Sidelobe Distortion.

2.2 MICROWAVE TRANSVERAL EQUALIZER OPERATION

The MTE has been designed by AIL to cancel the parasitic time sidelobes caused by distortion, by generating its own artificial equalizer sidelobes, whose envelopes are properly delayed (or advanced) to coincide in time with the parasitic sidelobes. In addition, the amplitude and phase of each artificial sidelobe can be separately adjusted to produce an equal amplitude phase reversed replica of the corresponding parasitic sidelobe.

Generation of the equalizer echoes is accomplished by tapping off portions of energy from the main signal lobe on the main transmission path, directing these portions through properly adjusted delay lines and reintroducing the delayed signal portions back into the main transmission path with proper amplitudes and polarity. The time domain output response of an ideal MTE is shown in Figure 2-2. It should be noted however that the actual response will consist of a train of echoes with decreasing amplitudes (0.75 dB/tap) due to the insertion loss of the mainline tapped delay elements. A correction for the aforementioned MTE echo transfer characteristics will be necessary in order to achieve the desired tap attenuator setting with adaptive closed loop operation (computer controlled).

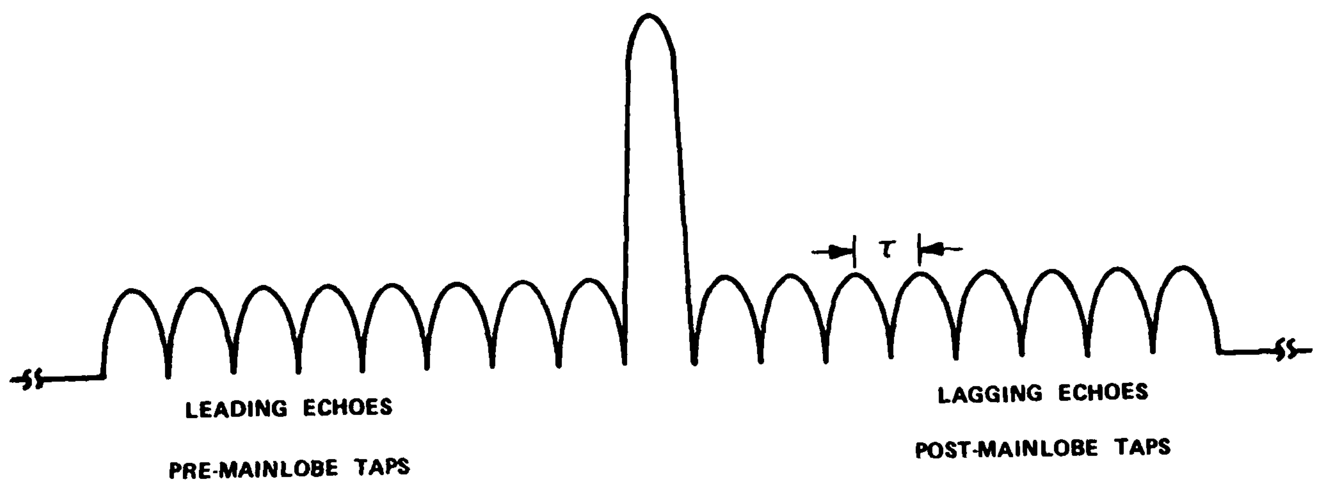


Figure 2-2. Time Domain Output of Microwave Transversal Equalizer.

The MTE consists of four cascade connected microwave integrated circuit mainlines and 32 secondary lines for the purpose of generating a train of 8 pre- and post-mainlobe echoes (16 total). A 3.3 nsec time delay between echoes of each train is provided.* The amplitude level and time delay of the selected 16 of the 32 possible echoes are individually adjusted with the PIN diode variable attenuator (electronically controlled), a line stretcher variable delay (mechanically controlled), included in each secondary line. The remaining 16 echo taps are terminated, however these taps are available for use as required.

It should be noted that the transfer characteristics of each of the 16 variable attenuators, as well as the 16 variable solid state time delay units (when they are developed) are required to properly achieve adaptive MTE control.

*Compatible with a 300 MHz bandwidth.

3. ALGORITHM DEVELOPMENT

An algorithm has been developed using FFT techniques to provide open loop control of the MTE. The technique will be described in terms of the relevant input and output formats, as well as the FFT algorithm.

3.1 INPUT FORMAT

The input to the FFT consists of a set of complex correlations as a function of frequency. These samples can be expressed by

$$H(j\omega_i) \simeq I\,(\omega_i) + jQ(\omega_i) \qquad (3\text{-}1)$$

where

ω_i is the i^{th} frequency

$I(\omega_i)$ is the in-phase frequency response

$Q(\omega_i)$ is the quad-phase frequency response

For the problem at hand, the following frequencies are considered:

$$\omega_i = 2\pi\,[f_L + (i-1)\,\Delta f],\ i = 1,2,\ldots.,\ 256 \qquad (3\text{-}2)$$

where

$f_L = 3.1$ GHz

$\Delta f = 1.17647\,\text{MHz} \simeq 1.18$ MHz

The last frequency is found by setting i = 256; that is,

$$f_H = 3.1 + 255 \times 1.18 \times 10^{-3} \doteq 3.4 \text{ GHz}$$

As a result, a total of 255 pairs of measurement data are involved for a bandwidth of 300 MHz as shown in the format presented in Table 3-1.

TABLE 3-1. TAPE FORMAT FOR THE FREQUENCY RESPONSE

Word Index	Notation	Frequency
1	$I(j\omega_1)$	3.1 GHz
2	$Q(j\omega_1)$	3.1 GHz
3	$I(j\omega_2)$	3.1 + Δf
4	$Q(j\omega_2)$	3.1 + Δf
0	0	0
0	0	0
0	0	0
509	$I(j\omega_{255})$	3.1 + 254 Δf
510	$Q(j\omega_{255})$	3.1 + 254 Δf
511	$I(j\omega_{256})$	3.4 GHz
512	$Q(j\omega_{256})$	3.4 GHz

Note:

1. $\Delta f = \dfrac{3.4 - 3.1}{255 \times 10^{-3}} \doteq 1.18$ MHz

2. Each word is represented by 8 bits in binary 2's complemert notation; i.e.:

$$-128 < I,Q < +127$$

The above representation defines the formula transformation required from the input medium to the computer.

The input samples are represented by an 8 bit binary two's complement notation. Thus, a word is defined to be 8 bit with the value of the word ranged from 127 to -128. In floating point notation the value is ranged from

$$127/128 = 0.992188 \quad \text{to}$$

$$-128/128 = -1$$

3.2 OUTPUT FORMAT

The output of the FFT consists of a set of $N_t = 17$ complex MTE tap coefficients which represents a truncated Fourier series representation of the equalizer transfer function. The total number of relative equalizer taps is sixteen with the center tap (number 9) taken as the reference tap.

The output tap coefficient will be presented by decimal representation with the absolute magnitude of the tap less than unity, that is: $-1 \leq C_i < 1$ for $i = 1,2,\ldots 17$ and phase angle is expressed in degree units.

Further data format transformation will be needed to match the exact setting on the MTE attenuator dial.

3.3 FFT ALGORITHM

Consider again the measured frequency response of a network with a transfer function $H(j\omega_i)$. The measurement can be functionally described by the block diagram shown in Figure 3-1. The input signal to the device can be written as

$$X_i(t) = A_i \cos(\omega_i t + \emptyset_o) \qquad (3\text{-}3)$$

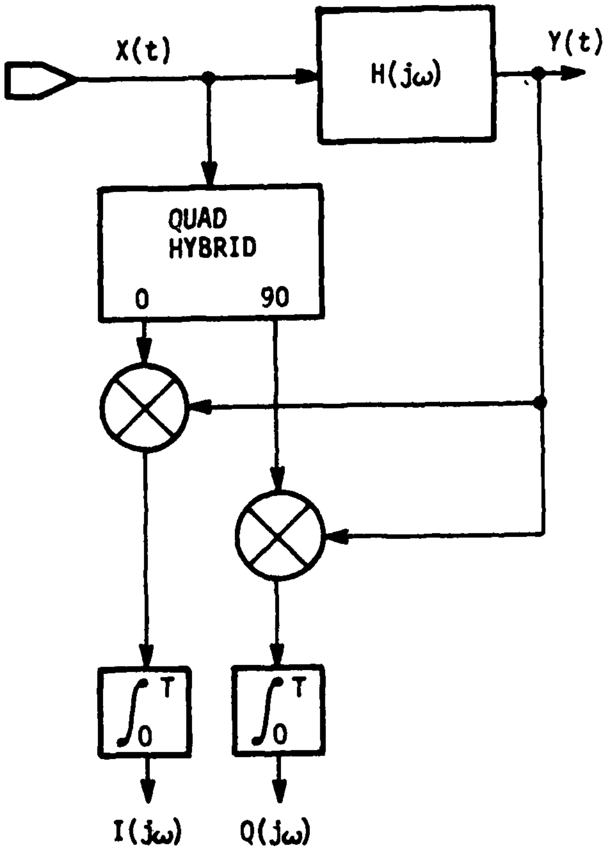


FIGURE 3-1. MEASUREMENT SETUP

where

$_i$ is the i^{th} in-band frequency

$\emptyset_o$ is the input phase angle

A is the amplitude of the signal

The response of the network is

$$Y_i(t) = B_i \cos(\omega_i t + \theta_i) \tag{3-4}$$

where

θ_i is the corresponding phase angle for the i^{th} signal

B_i is the corresponding amplitude for the i^{th} signal

The quadrature hybrid is used to generate the in-phase and the quad-phase components:

$$I(j\omega_i) = \frac{1}{T} \int_0^T A.B_i \cos(\omega_i t+\emptyset_o) \cos(\omega_i t+\theta_i)\, dt \tag{3-5}$$

and

$$Q(j\omega_i) = \frac{1}{T} \int_0^T A.B_i \sin(\omega_i t+\emptyset_o) \cos(\omega_i t+\theta_i)\, dt \tag{3-6}$$

where T is defined as the correlation time. The correlator output can be reduced to

$$I(j\omega_i) = \frac{AB_i}{2} \cos(\emptyset_o-\theta_i) \tag{3-7}$$

$$Q(j\omega_i) = \frac{AB_i}{2} \sin(\emptyset_o-\theta_i) \tag{3-8}$$

The gain and phase response as a function of frequency can be expressed by

$$G_i = \sqrt{I_i^2 + Q_i^2} = \frac{A}{2} . B_i \tag{3-9}$$

$$\theta_i = \tan^{-1}\left(\frac{Q_i}{I_i}\right) + \emptyset_o \tag{3-10}$$

where the short hand notation is adapted; that is

$$I_i = I(j\omega_i) \text{ and } Q_i = Q(j\omega_i)$$

Since A and $\emptyset_o$ are not a function of frequency, these measurement data represent the transfer function $H(j\omega_i)$. The impulse response h(nT) can be found by discrete Fourier transform method.

A few assumptions will be made such that the Fourier series approach can be applied:

(1) Signals passing through the network will be bandlimited to 300 MHz. To provide equalization over this bandwidth requires a corresponding time delay. This defines the delay between taps to be:

$$T = \frac{1}{f_H - f_L} = \frac{1}{300 \times 10^6} = 3.33 \text{ nsec}$$

(2) Since any digital filter spectrum to be realized will be made periodic of period f_s, then the spectrum can be represented by a Fourier series:

$$H(j\omega) = \sum_{k=-\infty}^{\infty} c_k \exp(jk\omega T) \tag{3-11}$$

where c_k is the k^{th} Fourier coefficient defined by

$$c_k = \frac{1}{\omega_s} \int_{-\frac{\omega_s}{2}}^{\frac{\omega_s}{2}} H(j\omega) \exp(-jk\omega T)\, d\omega \tag{3-12}$$

Generally, if the gain response is an even function and the phase response is an odd function, that is

$$G(j\omega) = G(-j\omega) \tag{3-13}$$

$$\theta(j\omega) = -\theta(-j\omega) \tag{3-14}$$

then the resultant coefficients will be real. Note that these requirements that $G(j\omega)$ and $\theta(j\omega)$ be even and odd respectively, lead to similar requirements that the in-phase and the quadrature-phase components to be even and odd, respectively. In practice, however, these symmetries cannot be assumed as shown in Figure 3-2. As a result, the coefficients will be complex; that is:

$$c_k = a_k + j\, b_k, \quad k = 1,2,\ldots\ldots,17 \tag{3-15}$$

where

$$g_k = \sqrt{a_k^2 + b_k^2} \tag{3-16}$$

$$\theta_k = \tan^{-1}(b_k/a_k) \tag{3-17}$$

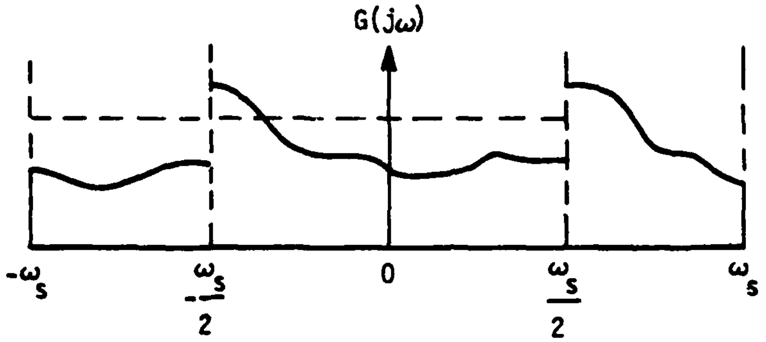
To achieve a set of real equalizer coefficients usually means to double the upper frequency bound of the signal samples (ω_s). When the upper frequency is doubled, the second half of the spectrum is copied from the original bandwidth with the proper polarity attached. Examples to illustrate these properties will be presented in the next section.

3.4 FFT ILLUSTRATIVE EXAMPLES

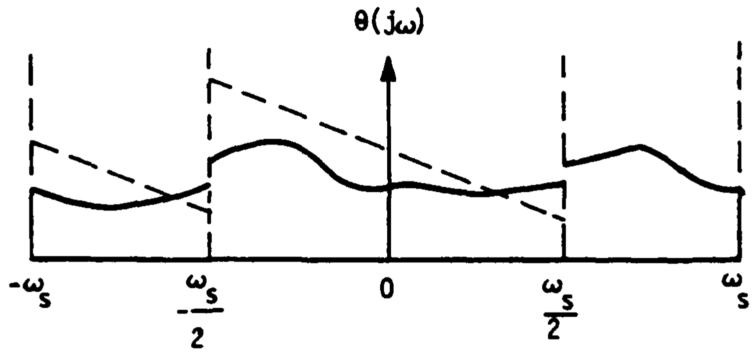
A few examples illustrates the Fourier transform approach to well defined $H(j\omega)$ will be given in this section.

3.4.1 Lowpass Filter

Figure 3-3 shows a specification for a lowpass transfer function with cutoff frequency set at 0.5 rad/sec and the upper frequency bound (ω_s) set at 2 rad/sec. The spectrum between $\omega_s/2$ to

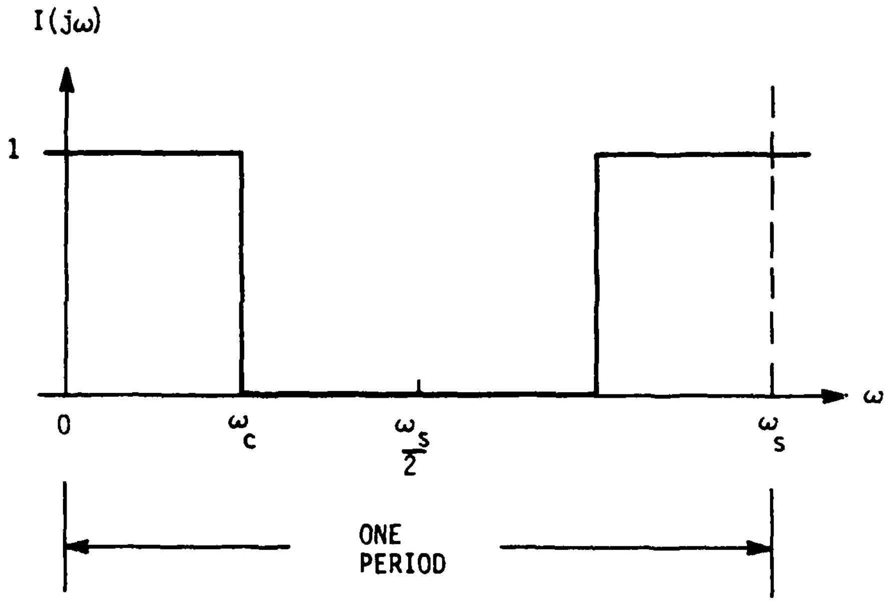


(a) GAIN SPECTRUM



(b) PHASE SPECTRUM

FIGURE 3-2. GAIN AND PHASE RESPONSE (GENERAL CASE - ASYMMETRIC)



ω_s = 2 RAD/SEC

ω_c = 0.5 RAD/SEC

$Q(j\omega)$ = 0 FOR ALL ω

FIGURE 3-3. IDEAL RESPONSE OF A LOWPASS FILTER

ω_s is duplicated to provide an even function for the in-phase response $I(j\omega)$. The quad-phase response is set to zero.

The Fourier coefficients can be readily found to be

$$c_k = \frac{1}{k_\pi} \sin (k_\pi), \quad k = -N, -N+1, \ldots N \quad k \neq 0 \text{ and } N = 8 \tag{3-18}$$

and $C_0 = 0.5$.

For the case $N = 8$, or 17-tap equalizer the response of the network can be plotted as shown in Figure 3-4.

Generally, the performance of such a filter indicates a passband ripple of 0.75 dB and a stopband rejection of approximately -20 dB. Direct truncation of the Fourier series leads to the well-known Gibbs phenomenon. The ripple, in generaly, cannot be reduced by simply including more taps as displayed by the 25 tap version shown in Figure 3-5.

The reduction of the passband ripple and stopband attenuation is later approached by finding a time-limited function whose Fourier transform best approximates a bandlimited function. This approach leads to the well-known Kaiser window expressed by:

$$\omega(K) = \frac{I_0 \left(\beta \sqrt{1 - (K/N)^2}\right)}{I_0 (\beta)} \quad -N < K < N$$

where β is a constant $(1 < \beta < 10)$

I_0 is the modified Bessel function of order zero

N is half the number of taps

The constant β can be determined experimentally. Figure 3-6 shows the same 17 taps frequency response with Kaisier window. The

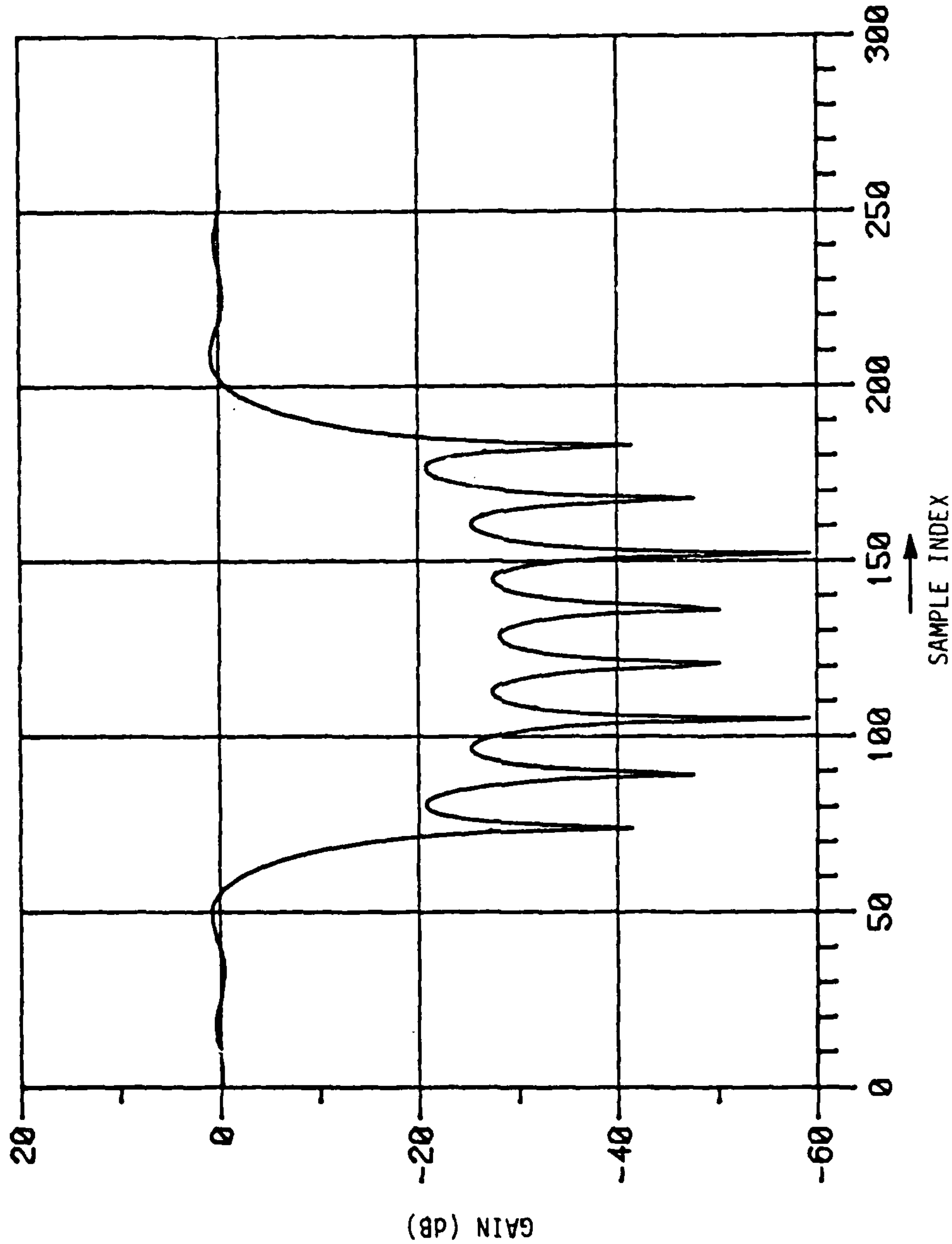


FIGURE 3-4. LOWPASS FILTER RESPONSE (17 Taps) DIRECT TRUNCATION

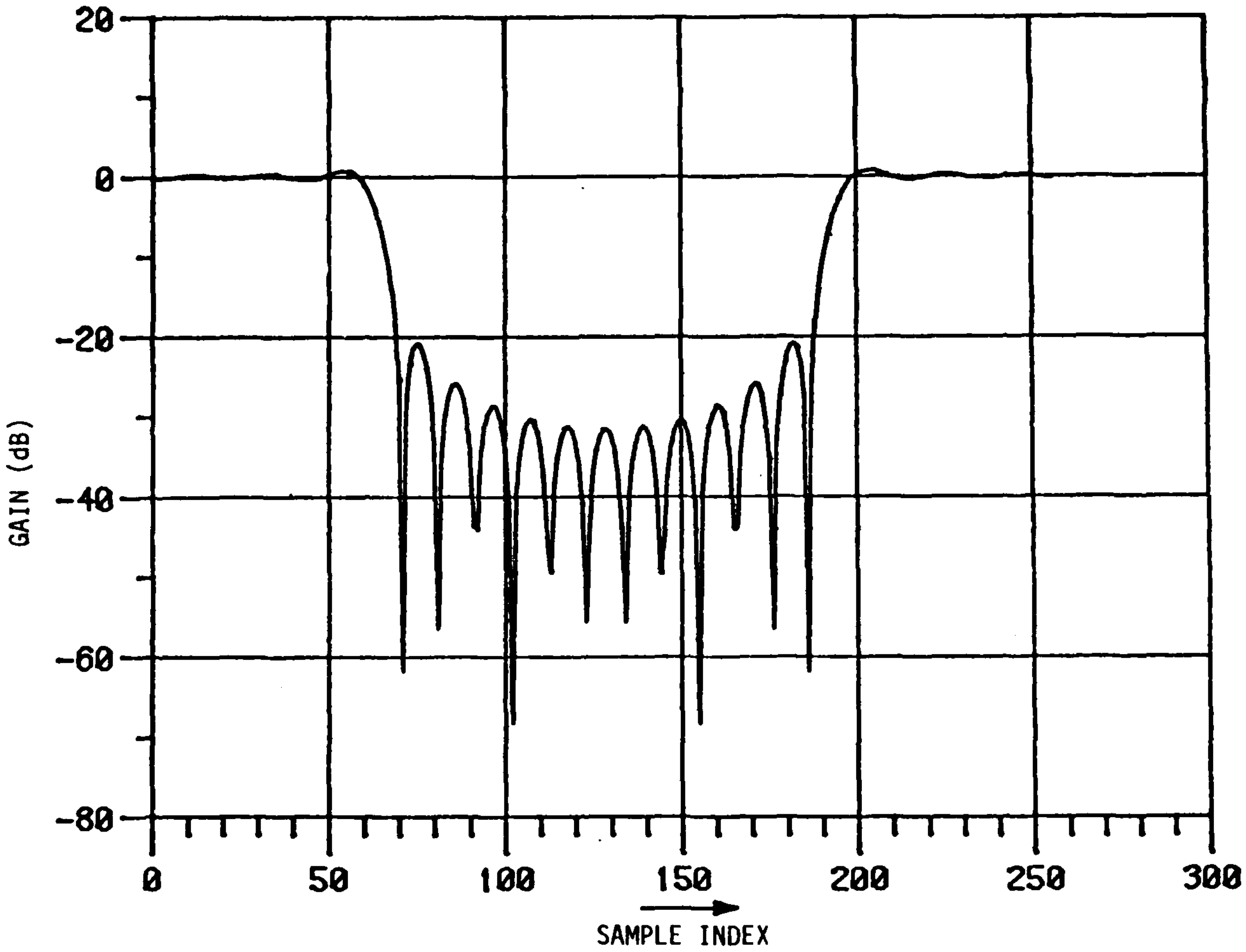


FIGURE 3-5. LOWPASS FILTER RESPONSE (25 Taps) DIRECTED TRUNCATION

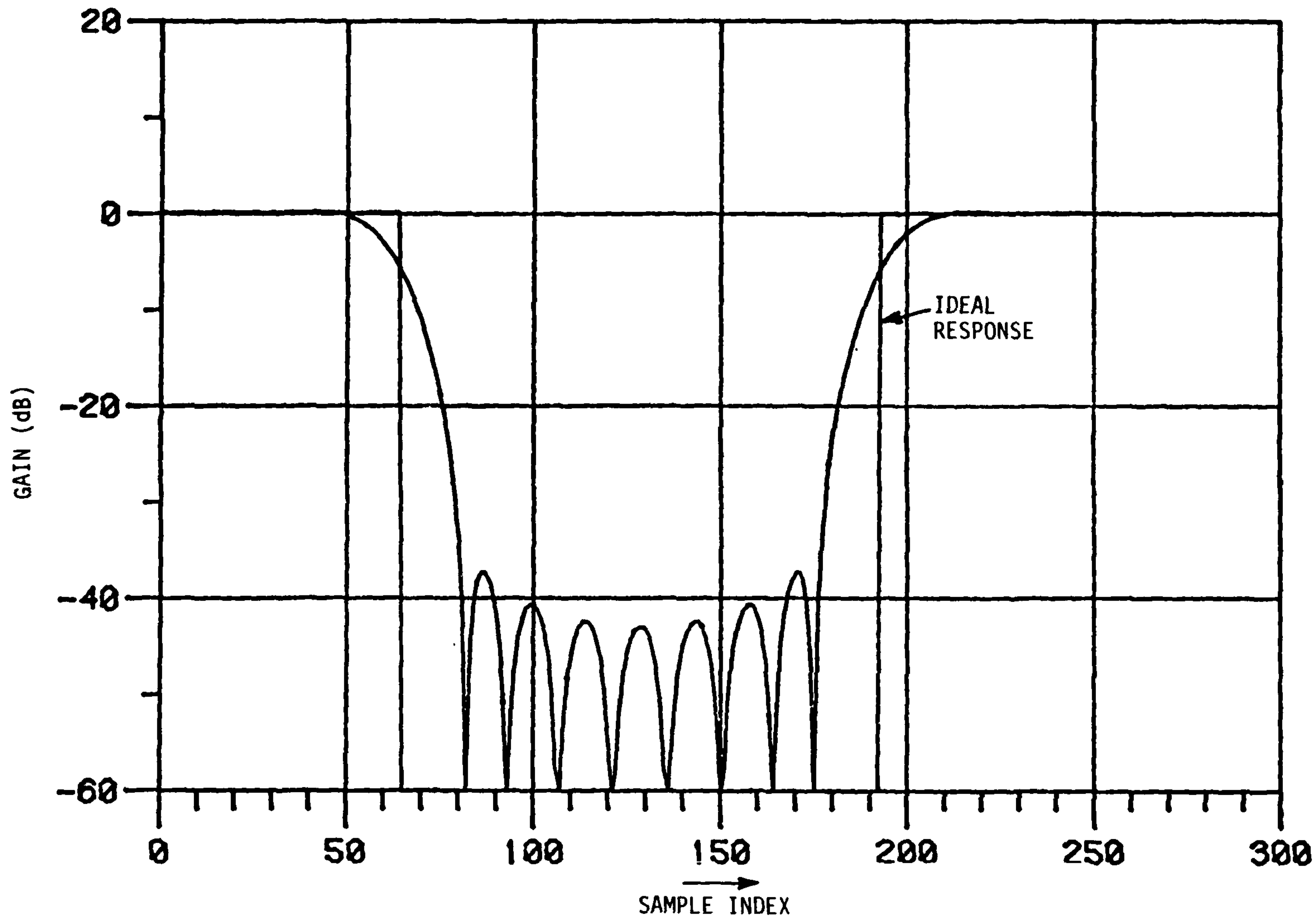


FIGURE 3-6. 17 - Tap LOWPASS FILTER WITH WINDOW

passband ripple is essentially vanished and the stopband rejection is reduced from -20 dB to -38 dB. Note that the windowing operation is a proper scale function applied to the impulse response of the filter. It does not change the hardware configuration of the filter.

3.4.2 A Differentiator

The response of a differentiator can be written by the equation

$$H(j\omega) = j\omega, \text{ for } -\omega_s/2 < \omega < \omega_s/2$$

This is the case that the in-phase response is zero and the quad-phase is an odd function. As a result, the coefficient (or the impulse response) is real and odd functions; that is,

$$c_k = \frac{2}{\omega_s} \int_0^{\frac{\omega_s}{2}} \omega \sin(k\omega T)\, d\omega$$

$$= \int_0^{1} \omega \sin(k\omega T)\, d\omega \text{ where } T = \frac{2\pi}{\omega_s} = \pi,\ \omega_s = 2$$

$$= -\frac{\cos(k\pi)}{k\pi},\quad k = -N, -N+1, -1, 1, \ldots N$$

$$k = 0$$

where $k = 0$, $C_o = 0$.

Figure 3-7 is the response of the filter with 17 taps and Figure 3-8 is the response with Kaiser window. The parameter β for the Kaiser window is set to 3 to reduce the ripple due to direct truncation of the Fourier series at $N = 8$.

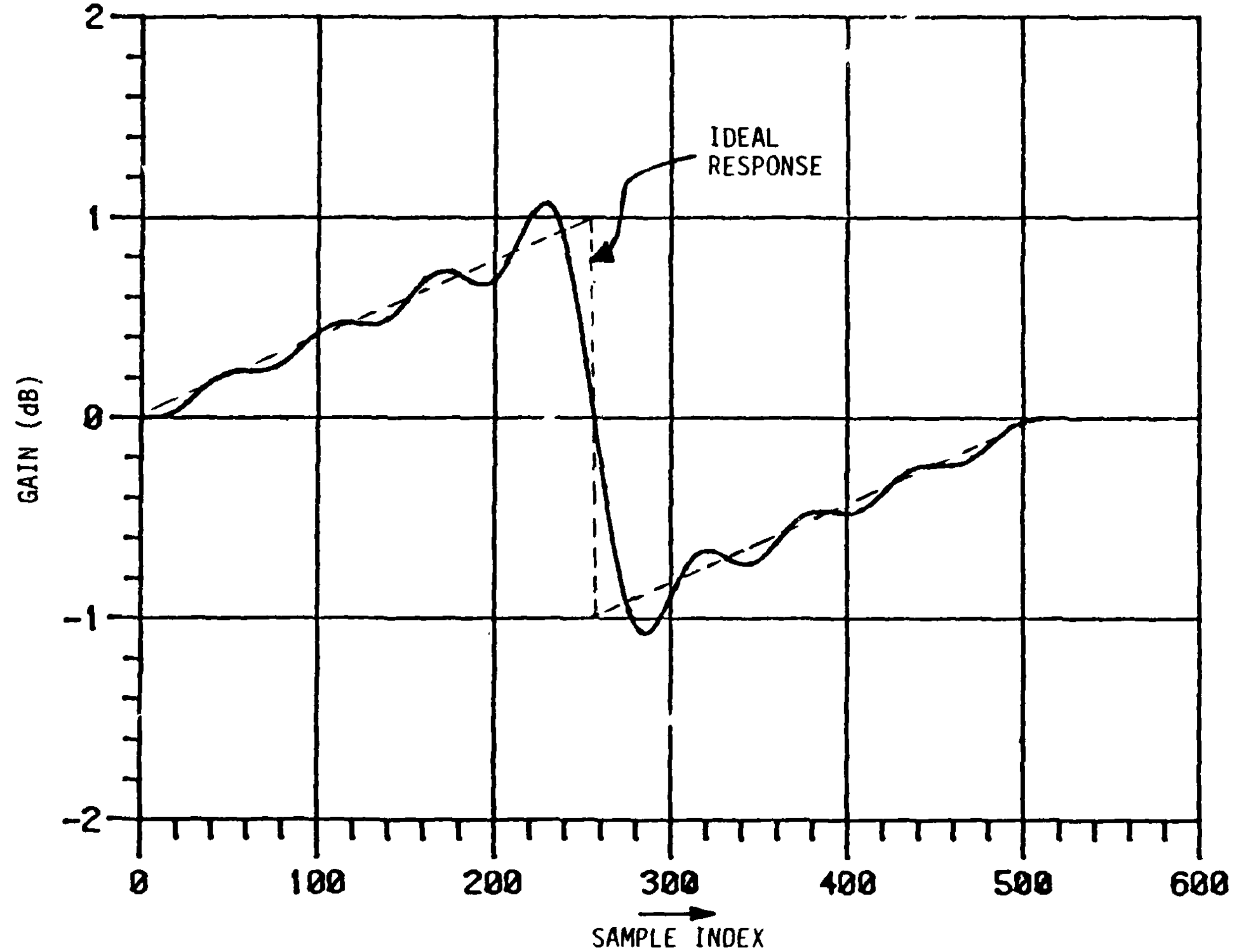


FIGURE 3-7. DIFFERENTIATOR RESPONSE (17 Taps) DIRECT TRUNCATION

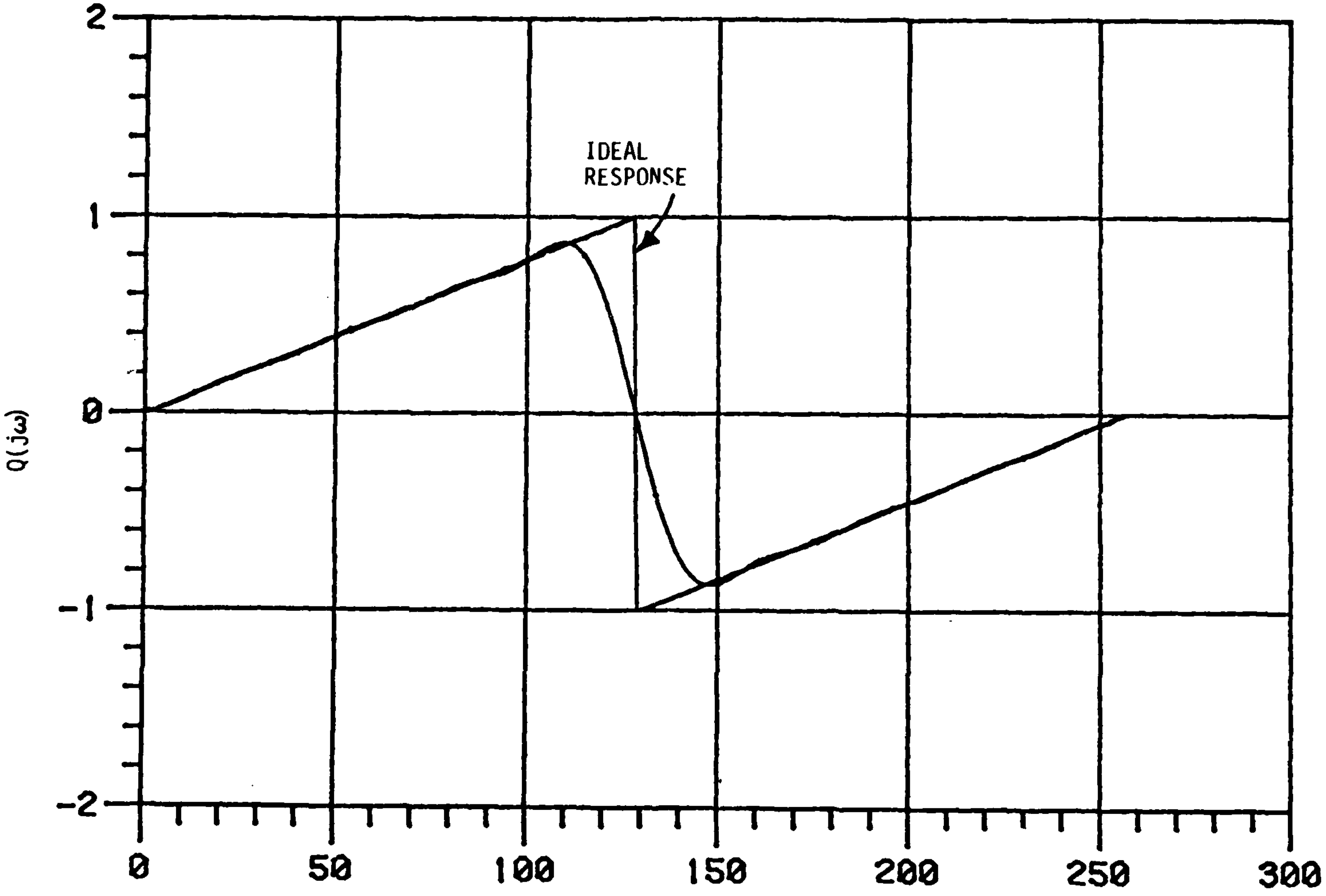


FIGURE 3-8. (17 Tap) DIFFERENTIATOR RESPONSE WITH WINDOW

3.4.3 Hilbert Transformer

The frequency response of a Hilbert transformer can be defined as

$$H(j\omega) = \begin{cases} -j, & 0 \leq \omega < \omega_s/2 \\ j, & \frac{\omega_s}{2} \leq \omega < \omega_s \end{cases}$$

The discrete Fourier transform can be expressed by the impulse response

$$h(n) = \begin{cases} \frac{\sin^2(n\pi/2)}{(n\pi/2)} & n \neq 0 \\ 0 & n = 0 \end{cases}$$

The comparison between a 17 tap filter and the ideal transform is shown in Figure 3-9.

3.4.4 Equalizer for an Arbitrary Function

The transfer function of an arbitrary frequency response

$$H(j\omega) = I(j\omega) + Q(j\omega)$$

over the band of interest is considered. Since the samples are not symmetrical in any sense, the resulting tap coefficients will be complex and asymmetrical. The matching of the ideal versus 17 tap equalizer is plotted in Figure 3-10 and the resulting complex taps are tabulated in Table 3-2. Note that the equalizer taps are complex:

$$c_k = X_k + j\ Y_k, \quad k = 1,2,\ldots.17$$

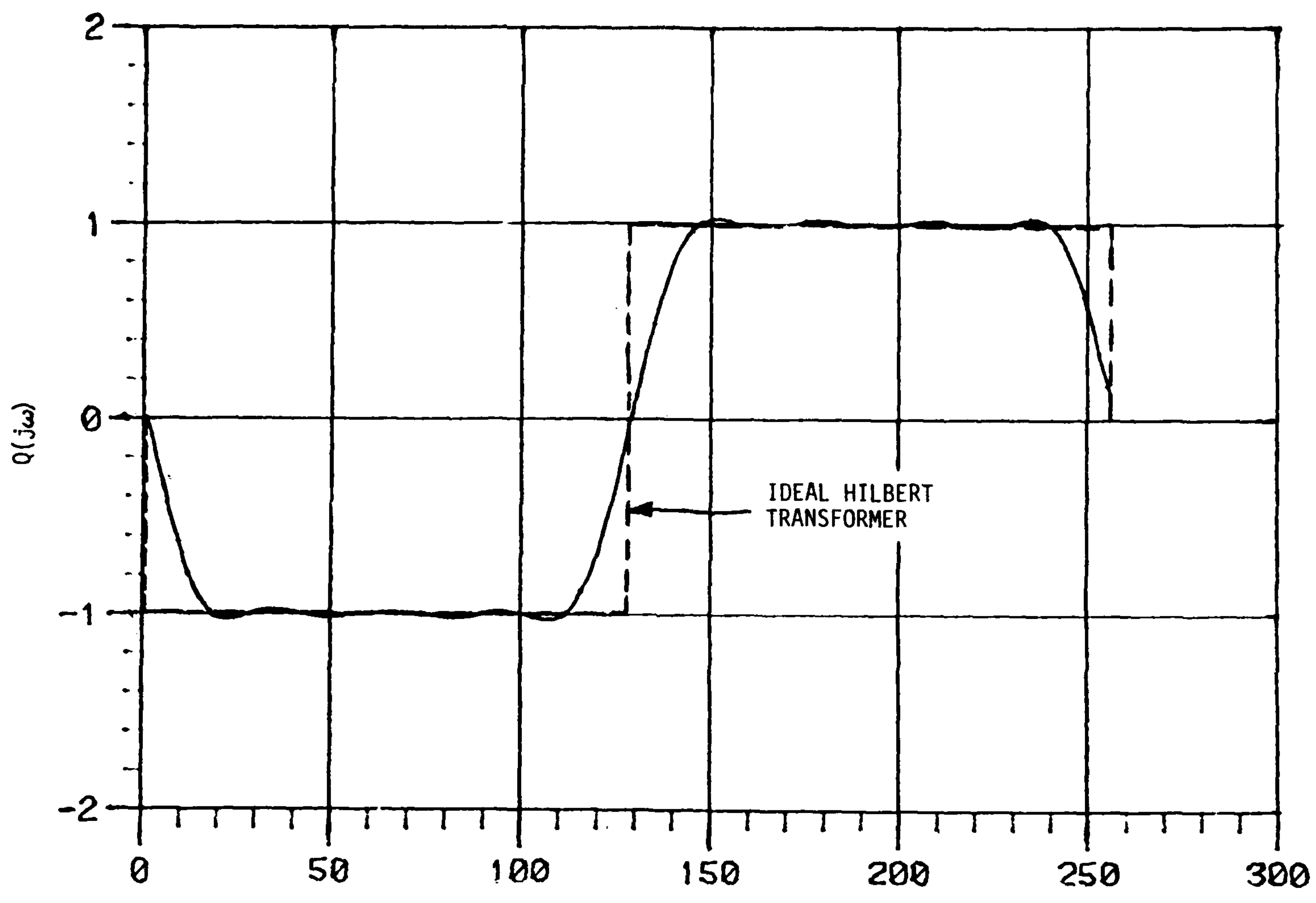


FIGURE 3-9. RESPONSE OF A (17 Tap) HILBERT TRANSFORMER (EXAMPLE 3)

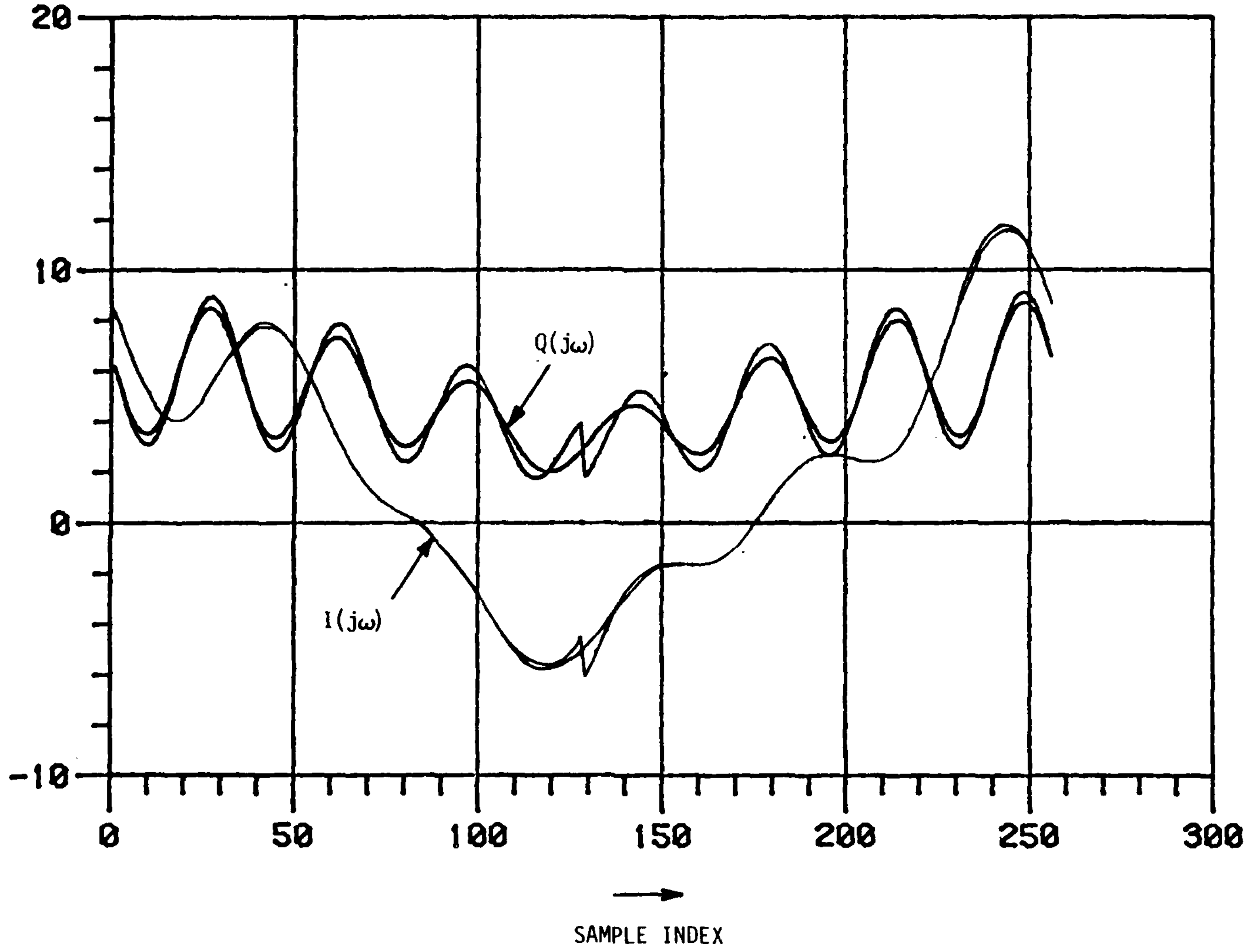


FIGURE 3-10. (17 Tap) EQUALIZER RESPONSE WITH AN ARBITRARY I(jω) AND Q(jω)

and the gain and phase adjustment can be found by the transformation

$$g_k = 10 \times \log (X_k^2 + Y_k^2)$$

$$\phi_k = \tan^{-1} (Y_k/X_k)$$

TABLE 3-2. RESULTING EQUALIZER TAPS

Index	X_i	Y_i
1	0.438043	-0.036516
2	0.931036	0.043327
3	-0.092596	-0.064665
4	0.086571	-0.686610
5	-0.074379	-0.578472
6	0.132042	-0.643479
7	-0.213581	-0.047679
8	3.133260	0.643330
9	2.183238	4.936495
10	3.120918	0.712413
11	-0.187885	-0.225917
12	0.090602	0.764618
13	-0.032419	0.508859
14	-0.005370	0.732557
15	0.036480	0.031661
16	-0.890202	-0.018254
17	-0.468842	0.016737

4. CONTROL PROGRAM

A Fast Fourier Transform (FFT) subroutine is employed to calculate the set tap coefficients for the equalizer. The equalizer taps are calculated in a recursive manner.

The input to the AMTE consists of a set of N complex correlations as a function of frequency. These samples can be expressed by the transfer function

$$H(j\omega_i) = I(\omega_i) + j\ Q(\omega_i),\ i = 1,2,\ldots.,N$$

where ω_i is the i^{th} radian frequency, $I(\omega_i)$ is the in-phase frequency response, and $Q(\omega_i)$ is the quad-phase frequency response, and N is the total number of sample points.

The output of the AMTE is a set of N_t equalizer taps. For the system at hand N_t is set equal to 17. These complex tap coefficients are obtained by a truncated Fourier series coefficients. The output tap coefficient will be presented by its magnitude setting (in dB's) and phase setting in degrees.

4.1 FLOW CHART

A simplified flow chart of the main routine is shown in Figure 4-1.

4.2 PROGRAM IMPLEMENTATION

The program is implemented on the RADC HP 2100, with the operating system configured on 2/6/81. The steps to be taken for execution of the program are:

After the logs on the HP 2100, the system will return the prompt sign ":". At this level, it is necessary to link the compiled version of the program $ZA::45. To link, one gives the command:

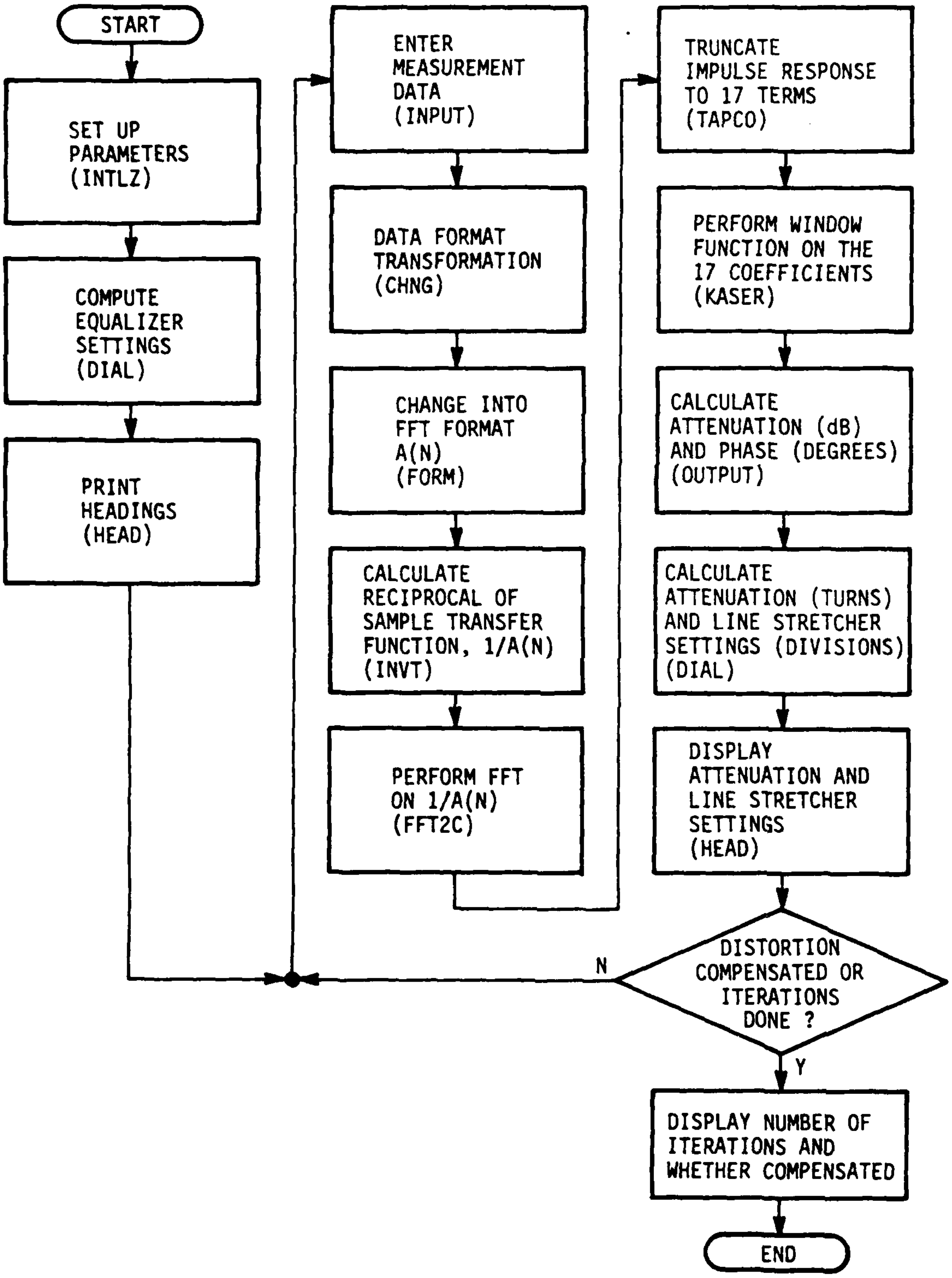


FIGURE 4-1. PROGRAM FLOW CKART

OF, AIL <CR>

Then after computer responds with ":" then type

RU, LOADR, XX <CR>

where XX is the peripheral number of the user's terminal (this may be optional) and <CR> is the carriage return key

The computer responds by giving a prompt of "/LOADR:". After this prompt, the user types:

RE, . ZA <CR>

The computer then lists the main line and the subroutines and functions called for by the user's program, in this case, $ZA::45 while linking them together. When the computer is through linking these together, it comes back with another "/LOADR:". The user should then type

/E <CR>

The computer than links and lists on the user's terminal as it is linking, library routines and functions necessary for executing the user's program. The computer then comes back with:

"XX PAGES RELOCATED" "XX PAGES REQ'D"

"NO PAGES EMA" "NO PAGES MSEG"

"/LOADR:XX READY AT XX (date, time)"

"LOADR· $END and it gives a ":" prompt

At the prompt, the user should run the program that has been loaded and linked, now called "AIL:. The command is:

"RU, AIL" <CR>

The computer then outputs anv messages and data from the program, as it is running. The sequence of output should be:

INITIAL SETTINGS AT STEP Ø SHOULD BE...

ENTER THE FILE NAME

At this point, the user types in the name of the input file (containing a distorted sinusoid, if created by AT10 as described), for example we could use ADØØØ, one input file from AT10; to use ADØØØ one types "ADØØØ" (return)

after the message "ENTER THE FILE NAME".

The computer then prints out this message:

INTERMEDIATE SETTINGS AT STEP 1 ARE:

The computer then prints out the attenuator and the line stretcher values for the equalizer, then continues.

If the distortion has not been compensated in the first iteration, the program goes back to subroutine INPUT. The program repeats this loop 10 times, or until the distortion is less than a threshold set to be -45 dB, whichever comes first. When the program reaches the point of exiting the loop, it prints out either:

"AFTER 10 ITERATIONS DISTORTION CANNOT BE COMPENSATED"

or:

"DISTORTION COMPENSATED AFTER XX ITERATIONS"

then:

"AIL: STOP"

The computer then returns to the system level, and the user can perform other tasks, or sign off.

A sample run of the program implemented with input data is contained in Appendix 1.

4.3 COMPUTER SUBROUTINES

The following discussion includes the entry points, input arguments and output arguments of each subroutine in the AMTE program, and what the subroutines or functions accomplish.

1: Subroutine INTLZ (DMY1, DMY2, N, NTAP, BETA, II0, IO, KSW) initializes constants and arrays.

Input: None

Output: DMY1, DMY2, N, BETA, IO, KSW

DMY1	A	Initial attenuator (dB) setting
DMY2	B	Initial phase (degrees) setting
N		Number of points to be transformed
NTAP		Number of taps
BETA		Constant used in KASER subroutine
IO		Output device number
KSW		Format Flag KSW = 0: the data is expressed in decimal form KSW = 1: the data is expressed in octal form
II0		Input device number (mag tape drive)

2: Subroutine DIAL (NTAP, DMY1, DMY2, X, Y)

This subroutine calculates attenuator and line stretcher settings.

Input: NTAP, DMY1, DMY2

DMY1, DMY2 - updated attenuation (dB) and phase (degrees)

Output: X, Y

X updated line stretcher (division) settings

Y updated attenuation (dB) settings

3: Subroutine Head (X,Y,NTAP, ISW, ICNT, IO)

This subroutine displays attenuator and line stretcher settings.

Input:	X, Y, NTAP, ISW, ICNT, IO
X, Y	line stretcher and attenuation settings
NTAP	number of taps
ISW	ISW = 0; sidelobe distortion < -45 dB
	ISW = 1; at least one sidelobe > -45 dB
ICNT	iteration counter

4. Subroutine INPUT (N, IBUF, IIO, IO, KSW)

INPUT defines input array of in-phase and quadrature-phase data.

Input:	N, IIO, IO, KSW, file name
File Name	file on which the data is stored
Output:	IBUF
IBUF	an array containing both I & Q data in octal form

5. Subroutine CHNG (N, IBUF)

This routine swaps the upper half of the FFT input array with the lower half.

Input: N, IBUF

Output: IBUF

6. Subroutine FORM (N, IBUF, A)

FORM will compute complex array $A = I + jQ$ and also gain = $5 \log_{10} (I^2+Q^2)$ and phase = $\tan^{-1} (Q/I)$

Input: N, IBUF

Output: A

7. Subroutine INVT (N,A)

This subroutine computes the sequence to be equalized:

$$A_i = \frac{1}{A_i} \text{ where } i = 1,2,3,\ldots.N$$

Input: N,A

Output: A

8. Subroutine FFT2C (A,M,IMK) computes the Fourier transform of the input array A

   Input: M,A

   Output: A

9. Subroutine TAPCO (N, A, NTAP, COEF, IO)

   TAPCO truncates array A(N) into finite terms. The number of terms is defined as NTAP (number of taps) to specify an equalizer. The variable IO is a device number for the output which is normally computer-dependent. The resulting equalizer taps are stored in COEF (NTAP).

10. Subroutine KASER (NTAP, BETA, COEF, IO)

    This subroutine applies the KAISER window function to the truncated Fourier transform output COEF (NTAP). A functional subroutine BESSL (X) is required to compute the window parameters. BESSL is the modified Bessel function of the first kind with zero order. The argument for the BESSL is X.

11. Subroutine OUTPT (DMY1, DMY2, COEF, NTAP, ISW)

    Subroutine OUTPT computes updated attenuation (dB) and phase shift (degrees).

$$X_i = 20 \log_{10} (10^A); \; A = \frac{DMYL_i}{20} + 1$$

$$Y_i = 20 \log_{10} (10^B); \; B = \frac{ATT_i}{20} + 1$$

$$\text{updated attenuation}_i = 20 \log_{10} (10^C)$$

where $C = \frac{X_i + Y_i}{20} + 1$

and $i = 1,2,\ldots.N$

DMY1 - previous attenuation (dB)

ATT - present tap indication (dB)

$$\text{phase shift} = \sum_{i=1}^{i=N} \text{phase}_i$$

Input: DMY1, DMY2, COEF, NTAP

DMY1 - updated attenuation setting (dB)

DMY2 - updated phase shift setting (degrees)

ISW - ISW = 0; sidelobe distortion < -45 dB

ISW = 1; at least one sidelobe > -45 dB

12: Subroutine DIAL (NTAP, DMY1, DMY2, ISW, ICNT, IO)

DIAL was described previously.

13: Subroutine HEAD (X, Y, NTAP, ISW, ICNT, IO)

HEAD was described previously.

14: Subroutine END (ISW, ICNT, IO)

This subroutine will print a message depending on what the inputs are:

Input: ISW, ICNT, IO

Output: If ISW = 0, message is "Distortion compensated after so many iterations"

If ICNT = 10, message is "After 10 iterations distortion cannot be compensated"

15: Function ZLOG2: This function calculates log to the base 10 of the input quantity X. It takes the natural log (ALOG) of X and multiplies by Y, where Y = $1/\log_e 10$.

16: Function ZN1: This calculates arctangent of input quantity (in radians).

4.4 COMPUTER PROGRAM LISTING

The following is a listing of the computer program:

```
 FTN4
C      REVISED FILE 1/22/81
       PROGRAM AIL
       COMPLEX A(256),B(256),COEF(256)
       DIMENSION IBUF(256),DMY1(17),DMY2(17),X(17),Y(17)
       DIMENSION IWK(9)
       ICNT=0
       M=8
C                INITIALIZE CONSTANTS
       CALL INTLZ (DMY1,DMY2,N,NTAP,BETA,IIO,IO,KSW)
C          CALCULATE INTITIAL ATTENUATOR AND LINE STRETCHER SETTING
       CALL DIAL(NTAP,DMY1,DMY2,X,Y)
C             DISPLAY ATTENUATOR AND LINE STRETCHER SETTINGS
       CALL HEAD  (X,Y,NTAP,ISW,ICNT,IO)
       DO 10 I= 1,10
       ICNT= I
       IF(ICNT.EQ.2) GO TO 20
C          ENTER SAMPLE TRANSFER FUNCTION AS I AND Q DATA
       CALL INPUT(N,IBUF,IIO,IO,KSW)
C                  SHIFT DATA N/2 SAMPLES
       CALL CHNG(N,IBUF)
C           COMPUTE TRANSFER FUNCTION A(N)=I(N)+JQ(N)
       CALL FORM(N,IBUF,A)
C            COMPUTE EQUALIZER TRANSFER FUNCTION 1/A(N)
       CALL INVT(N,A)
C                PERFORM FFT ON A
       CALL FFT2C(A,M,IWK)
C           TRUNCATE THE IMPULSE RESPONSE TO 17 TERMS
       CALL TAPCO (N,A,NTAP,COEF,IO)
C            PERFORM WINDOW FUNCTION ON TRUNCATED SERIES
       CALL KASER (NTAP,BETA,COEF)
C          CALCULATE ATTENUATION (DB) AND PHASE SHIFT (DEGREES)
       CALL OUTPT (DMY1,DMY2,COEF,NTAP,ISW)
C         CALCULATE ATTENUATION (TURNS) AND LINE STRETCHER SETTINGS
       CALL DIAL(NTAP,DMY1,DMY2,X,Y)
C             DISPLAY ATTENUATION AND LINE STRETCHER SETTINGS
       CALL HEAD  (X,Y,NTAP,ISW,ICNT,IO)
       IF(ISW.EQ.0)GO  TO 20
 10    CONTINUE
C      DISPLAY AFTER HOW MANY ITERATIONS DISTORTION WAS
C      COMPENSATED ,OR THAT IT WAS NOT COMPENSATED AFTER A
C      GIVEN NUMBER OF ITERATIONS.
 20    CALL END(ISW,ICNT,IO)
 9998  WRITE(37,99980)
99980  FORMAT(" AIL : STOP")
       END
```

```
C
C
C          CALCUATE LOG TO THE BASE 10
      FUNCTION ZLOG1 (X)
      Y=0.4342944819
      IF(X.LE.0.)X=1.E-10
       ZLOG1=Y*ALOG(X)
      RETURN
      END
C
C        CALCULATE ARCTANGENT
C
      FUNCTION ZN2(Y,X)
      IF(X.EQ.0.)X=1.E-10
      PI=3.141592654
      Z=Y/X
      ZN2=ATAN(Z)
      IF(X.LT.0) ZN2=ZN2+PI
C     WRITE (37,976) Z,ZN2
976   FORMAT (1X,E12.5,1X,E12.5)
      RETURN
      END
C
C
C         CALCULATE BESSEL FUNCTION OF THE FIRST KIND WITH 0 ORDER
      FUNCTION BESSL(X)
      Y=X/2
      DELTA=1E-8
      E=1.
      DE=1.
      DO 1 I=1,25
      DE=DE*Y/FLOAT(I)
      SDE=DE*DE
      E=E+SDE
      IF(E*DELTA.GT.SDE)GOTO 10
 1    CONTINUE
 10   BESSL=E
      RETURN
      END
```

```
C
C
C           SWAP THE UPPER HALF OF THE ARRAY WITH THE LOWER HALF
      SUBROUTINE CHNG(N,IBUF)
      INTEGER IBUF(1),IBUF1(300)
       K1=(N+1)/2
      K2=K1-1
      K=N/2
      DO 10 I=K1,N
      J=I-K
      IBUF1(J)=IBUF(I)
   10 CONTINUE
      DO 20 I=1,K
      J=I+K1
      IBUF1(J)=IBUF(I)
   20 CONTINUE
      DO 30 I=1,N
      IBUF(I)=IBUF1(I)
   30 CONTINUE
  998 FORMAT (1X,12(I6,1X))
      ZX=1
      CALL TYPE(ZX)
      RETURN
      END
```

```
C
C      PERFORM POLYNOMIAL FUNCTION TO CONVERT FROM ATTENUATION (DB)
C        TO ATTENUATION (DIAL TURNS) AND FROM PHASE SHIFT(DEGREES)
C          TO LINE STRETCHER SETTINGS (DIVISIONS)
C
       SUBROUTINE DIAL (NTAP,DMY1,DMY2,X,Y)
       DIMENSION A0(17),A1(17),A2(17),A3(17),
      1DMY1(1),DMY2(1),X(1),Y(1)
       DATA A0/-.9102,-.7261,-.9251,-.6881,-.8607,-1.2297,
      1-1.1216,-1.1216,0.,-.8644,-.8416,-1.0381,-.9169,
      2-2.1413,-.8422,-1.0342,-.7343/
       DATA A1/-.1166,-.07328,-.1116,-.06238,-.1026,-.12932,
      1-.1075,-.1075,0.,-.08205,-.07806,-.1176,-.0946,-.1399,
      2-.05173,-.1192,-.07771/
       DATA A2/-.0009174,.001295,-.0003383,.001829,.0003621,
      1.00007163,-.0003624,-.0003624,0.,.0003096,-.0002862,
      2-.00003543,.000672,.0006742,.002387,-.001177,.001285/
       DATA A3/-.0000383797,.00000377,-.00002436,.000011075,
      1-.0000212,-.000033318,-.00002773,-.00002773,0.,-.000025096,
      2-.00002957,-.0000273,-.000016696,-.00001571,.00001276,
      3-.00004344,-.000008216/
       DO 10 I=1,NTAP
       J=I
       IF(I.LE.8)J=9-I
       Y1=A0(I)+A1(I)*DMY1(J)+A2(I)*DMY1(J)*DMY1(J)
       Y2=A3(I)*DMY1(J)*DMY1(J)*DMY1(J)
       Y(I)=Y1+Y2
       X(I)=DMY2(J)*3.634/(360.*1.091)
       IF(Y(I).LT.0.0)Y(I)=0.0
       IF(Y(I).GT.10.0)Y(I)=10.0
       IF(X(I).LT.0.0)X(I)=0.0
       IF (X(I).GT.11.0) X(I)=11.0
 10    CONTINUE
C      WRITE (37,979) (X(I),I=1,NTAP)
C      WRITE (37,979) (Y(I),I=1,NTAP)
 979   FORMAT (1X,4(E12.5,1X,E12.5,1X))
       ZX=2
       CALL TYPE (ZX)
       RETURN
       END
```

```
C
C
C        PRINT OUT WHETHER DISTORTION IS COMPENSATED AFTER THE
C          LAST ITERATION
      SUBROUTINE END(ISW,ICNT,IO)
      IF(ISW.EQ.0)GO TO 10
      WRITE(37,20)INCT
   20 FORMAT(/,3X,5HAFTER,I3,31H ITERATIONS DISTORTION CAN NOT,
     114HBE COMPENSATED,/)
      GO TO 30
   10 CONTINUE
      ICNT=ICNT-1
      WRITE(37,40)ICNT
   40 FORMAT(/,3X,28HDISTORTION COMPENSATED AFTER,I2,11HITERATIONS
   30 CONTINUE
      ZX=3
      CALL TYPE(ZX)
      RETURN
      END
C
C
C
      SUBROUTINE FORM(N,IBUF,A)
      COMPLEX A(1)
      DIMENSION IBUF(1)
C          COMPUTE COMPLEX ARRAY A=I+JQ
      DO 10 I=1,N
      IA1=IBUF(I)/256
      IB1=IBUF(I)-256*IA1
      IF(IA1.GE.128)IA1=IA1-256
      IF(IB1.GE.128)IB1=IB1-256
      XI=FLOAT(IA1)/128.
      XQ=FLOAT(IB1)/128.
      A(I)=CMPLX(XI,XQ)
   10 CONTINUE
  998 FORMAT (1X,4(E12.5,1X,E12.5,1X))
      ZX=4
      CALL TYPE(ZX)
      RETURN
      END
C
```

```
C
      SUBROUTINE HEAD   (X,Y,NTAP,ISW,ICNT,ID)
      DIMENSION X(1),Y(1),ZA(17)
      DATA ZA/4H1 A=,4H2 A=,4H3 A=,4H4 A=,4H5 A=,4H6 A=,4H7 A=,
     14H8 A=,4H M =,4H1 B=,4H2 B=,4H3 B=,4H4 B=,4H5 B=,4H6 B=,
     14H7 B=,4H8 B=/
      WRITE(37,145)
  145 FORMAT(1H1)
      ICNT1=ICNT+1
      GO TO (10,20,20,20,20,20,20,20,20,20,30)ICNT1
   10 CONTINUE
      WRITE(37,150)ICNT
C           DISPLAY ATTENUATION AND LINE STRETCHER SETTINGS
  150 FORMAT(/,18X,25HINITIAL SETTINGS AT STEP ,I1,11H SHOULD BE:
      GO TO 205
   20 CONTINUE
      ISW1=ISW+1
      GO TO (30,200)ISW1
  200 CONTINUE
      WRITE(37,155)ICNT
  155 FORMAT(/,18X,29HINTERMEDIATE SETTINGS AT STEP,I2,5H ARE: )
      GO TO 205
   30 CONTINUE
      WRITE(37,160)ICNT
  160 FORMAT(/,21X,23HFINAL SETTINGS AT STEP ,I2,5H ARE: )
  205 CONTINUE
      WRITE(37,965)
 965  FORMAT (//10X,17HATTENUATOR(TURNS),31X,19HLINE STRETCHER(DIV
      DO 210 I=1,NTAP
      WRITE(37,170)ZA(I),Y(I),ZA(I),X(I)
  170 FORMAT(10X,A4,F6.3,34X,A4,F4.1)
  210 CONTINUE
      ZX=5
      CALL TYPE (ZX)
      RETURN
      END
C
```

```
C
      SUBROUTINE INPUT(N,IBUF,IID,IO,KSW)
      DIMENSION IDCB(256),NAM(3),IBUF(256)
      DIMENSION IBUF0(128),IBUF1(128)
      WRITE(37,10)
 10   FORMAT(/,' ENTER THE FILE NAME ')
      READ(37,15)NAM
 15   FORMAT(3A2)
      IL=N
      CALL OPEN(IDCB,IERR,NAM)
      IF(IERR.LT.0)GO TO 900
C            ENTER DATA IN OCTAL FORMAT
      CALL READF(IDCB,IERR,IBUF0)
      CALL READF(IDCB,IERR,IBUF1)
      IF(IERR.LT.0)GO TO 910
      CALL CLOSE(IDCB,IERR)
C        FILL ARRAY WITH IN-PHASE AND QUADRATURE-PHASE COMPONENTS
      DO 60 J=1,128
      IBUF(J)=IBUF0(J)
 60   CONTINUE
      DO 70 K=1,128
      IBUF(K+128)=IBUF1(K)
 70   CONTINUE
 998  FORMAT (1X,12(I6,1X))
      GO TO 999
 900  WRITE(37,30)IERR
 30   FORMAT(/," FMP ERROR "I4)
      GO TO 999
 910  WRITE(37,20)
 20   FORMAT(/,' ERROR READING THE FILE ')
 999  ZX=6
      CALL TYPE(ZX)
      RETURN
      END
C
```

```
C
C                 INITIALIZE CONSTANTS AND ARRAY
       SUBROUTINE INTLZ(DMY1,DMY2,N,NTAP,BETA,IIO,KSW)
       DIMENSION DMY1(1),DMY2(1),A(17),B(17)
       DATA A/-59.964,-59.964,-51.072,-56.743,-77.661,-60.551,
      1-72.714,-57.353,0.,-58.449,-62.143,-55.373,-57.741,
      2-54.419,-69.504,-57.047,-58.965/
       DATA B/659.23,756.49,810.53,983.44,162.1,399.86,
      1670.03,859.16,443.09,129.68,399.86,561.96,778.1,
      254.04,151.3,345.82,670.03/
       N=256
       NTAP=17
       DO 10 I=1,NTAP
       DMY1(I)=A(I)
       DMY2(I)=B(I)
   10  CONTINUE
       BETA=1.0
       IIO=37
       IO=37
C      FOR DECIMAL I&Q,KSW=0
C      FOR OCTAL I&Q,KSW=1
       KSW=1
       ZX=7
       CALL TYPE(ZX)
       RETURN
       END
C
```

```
C
C            CALCULATE THE RECIPROCAL OF THE TRANSFER FUNCTION
        SUBROUTINE INVT(N,A)
        COMPLEX A(1)
        DO 10 I=1,N
        A(I)=1./(A(I)*FLOAT(N))
10      CONTINUE
C       WRITE (37,998) (A(I),I=1,N)
998     FORMAT (1X,4(E12.5,1X,E12.5,1X))
        ZX=8
        CALL TYPE(ZX)
        RETURN
        END
C
C
C          APPLY KAISER WINDOW FUNCTION TO SMOOTH OUT THE RIPPLE DUE
C           TO TRUNCATION OF THE FOURIER SERIES
        SUBROUTINE KASER(N,B,C)
        COMPLEX C(1)
        IF(B.EQ.0.0)GO TO 9999
        NV2=(N+1)/2
        DO 10 I=1,N
        Z2=FLOAT(I-NV2)/FLOAT(N/2)
        Z3=B*SQRT(1.-Z2*Z2)
        C(I)=C(I)*BESSL(Z3)/BESSL(B)
 10     CONTINUE
C       WRITE (37,979) (C(I),I=1,N)
979     FORMAT (1X,4(1X,E12.5,1X,E12.5))
9999    ZX=9
        CALL TYPE(ZX)
        RETURN
        END
C
```

```
C
      SUBROUTINE OUTPT(DMY1,DMY2,COEF ,NTAP,ISW)
      COMPLEX COEF(1)
      DIMENSION DMY1(1),DMY2(1),ATT(17),PHS(17)
      DO 10 I=1,NTAP
      XR=REAL(COEF(I))
      XI=AIMAG(COEF(I))
C         COMPUTE ATTENUATION OF DISTORTION
      AA=XR*XR+XI*XI
      ATT(I)=10.*ZLOG1(AA)
C         COMPUTE THE PHASE SHIFT OF TAP COEFFICIENTS
      PHS(I)=ZN2(XI,XR)*45./ATAN(1.)
      ZZ=ZN2(XI,XR)
C     WRITE (37,978) PHS(I),ZZ
978   FORMAT (1X,E12.5,1X,E12.5)
 10   CONTINUE
      DO 20 I=1,NTAP
      ATT(I)=ATT(I)-ATT(9)
      PHS(I)=PHS(I)-PHS(9)
 20   CONTINUE
      ISW=0
      NTAP1=NTAP-1
      DO 30 I=1,NTAP1
      J=I
      IF(I.GE.9)J=I+1
      IF(ATT(J).LE.-45.)GO TO 15
      ISW=1
      GO TO 30
15    CONTINUE
      ISW=ISW
 30   CONTINUE
      IF(ISW.EQ.0)GO TO 40
      DO 50 I=1,NTAP
      X=20.*ZLOG1((10.**(DMY1(I)/20.))+1)
      Y=20.*ZLOG1((10.**(ATT(I)/20.))+1)
      DMY1(I)=20.*ZLOG1((10.**((X+Y)/20.))-1)
      DMY2(I)=DMY2(I)+PHS(I)
 50   CONTINUE
 40   CONTINUE
      WRITE (37,996)
996   FORMAT (1X,14HATTENUATOR(DB),37X,19HLINE STRETCHER(DEG),/)
888   ZX=10
      DO 881 I=1,NTAP
      WRITE (37,977)  DMY1(I),DMY2(I)
977   FORMAT (1X,E12.5,37X,E12.5)
881   CONTINUE
979   FORMAT (1X,4(1X,E12.5,1X,E12.5))
      CALL TYPE(ZX)
      RETURN
      END
C
```

```
C
      SUBROUTINE TAPCO(N,A,NTAP,COEF,ID)
      COMPLEX A(1),COEF(1)
      NV2=(NTAP+1)/2
      NV21=N+NV2+1
      NV1=NV2+1
C       TRUNCATE THE ARRAY A(N) INTO A GIVEN NUMBER(NTAP) OF
C         TERMS
      DO 20 I=1,NV2
      J=NV2-I+1
      COEF(I)=A(J)
   20 CONTINUE
C       OBTAIN THE UPPER HALF OF THE COEFFICIENTS CENTERED
C         AROUND (NTAP+1)/2
      DO 30 I=NV1,NTAP
      J=NV21-I
      COEF(I)=A(J)
   30 CONTINUE
C     WRITE (37,979) (COEF(I),I=1,NTAP)
  979 FORMAT (1X,4(1X,E12.5,1X,E12.5))
      ZX=11
      CALL TYPE(ZX)
      RETURN
      END
```

```
C
C
      SUBROUTINE FFT2C (A,M,IWK)
      INTEGER  M,IWK(1)
      COMPLEX A(1)
C
C
      INTEGER I,ISP,J,JJ,JSP,K,K0,K1,K2,K3,KB,KN,MK,MM,MP,N,
     1  N4,N8,N2,LM,NN,JK
      REAL RAD,C1,C2,C3,S1,S2,S3,CK,SK,SQ,A0,A1,A2,A3,
     1  B0,B1,B2,B3,TWOPI,TEMP,
C    2  ZERO,ONE,Z0(2),Z1(2),Z3(2)
     2  ZERO,ONE,Z0(2),Z1(2),Z3(2),Z2(2)
      COMPLEX ZA0,ZA1,ZA2,ZA3,AK2
      EQUIVALENCE (ZA0,Z0(1)),(ZA1,Z1(1)),(ZA2,Z2(1)),
     1 (ZA3,Z3(1)),(A0,Z0(1)),(B0,Z0(2)),(A1,Z1(1)),
     2 (B1,Z1(2)),(A2,Z2(1)),(B2,Z2(2)),(A3,Z3(1)),
     3 (B3,Z3(2))
      DATA SQ/.707106781/
      DATA SK/.382683432/
      DATA CK/.923879533/
      DATA TWOPI/6.28318531/
      DATA    ZERO/0.0/,ONE/1.0/
      MP = M+1
      N = 2**M
      IWK(1) = 1
      MM=(M/2)*2
      KN = N+1
      DO 5  I=2,MP
         IWK(I) =IWK(I-1)+IWK(I-1)
    5 CONTINUE
      RAD = TWOPI/N
      MK = M - 4
      KB = 1
      IF (MM .EQ. M) GO TO 15
      K2 = KN
      K0 = IWK(MM+1) + KB
   10 K2 = K2 - 1
      K0 = K0 - 1
      AK2 = A(K2)
      A(K2) = A(K0) - AK2
      A(K0) = A(K0) + AK2
      IF (K0 .GT. KB) GO TO 10
   15 C1=ONE
      S1 = ZERO
      JJ = 0
      K = MM - 1
```

```
       J = 4
      IF (K .GE. 1) GO TO 30
      GO TO 70
   20 IF (IWK(J) .GT. JJ) GO TO 25
      JJ = JJ - IWK(J)
      J = J - 1
      IF (IWK(J) .GT. JJ) GOTO 25
      JJ=JJ-IWK(J)
      J=J-1
      K = K + 2
      GO TO 20
   25 JJ = IWK(J) + JJ
      J = 4
   30 ISP = IWK(K)
      IF (JJ .EQ. 0) GO TO 40
      C2 = JJ * ISP * RAD
      C1 = COS(C2)
      S1 = SIN(C2)
   35 C2 = C1 * C1 -S1 * S1
       S2 = C1 *(S1 +S1)
       C3 = C2 * C1 -S2 *S1
       S3= C2 * S1 + S2 *C1
   40  JSP = ISP + KB
C     WRITE (37,997) S1,S2,S3
C     WRITE (37,997) C1,C2,C3
       DO 50 I = 1 , ISP
      K0 = JSP - I
      K1 = K0 + ISP
      K2 = K1 + ISP
      K3 = K2 + ISP
      ZA0 = A(K0)
      ZA1 = A(K1)
      ZA2 = A(K2)
      ZA3 = A(K3)
      IF (S1 .EQ. ZERO) GO TO 45
      TEMP = A1
```

```
      A1 = A1 * C1 -B1 *S1
      B1 = TEMP * S1 + B1 * C1
      TEMP = A2
      A2 = A2 * C2 - B2 * S2
      B2 = TEMP * S2 + B2 * C2
      TEMP = A3
      A3 = A3 * C3 - B3 *S3
      B3 = TEMP * S3 + B3 * C3
 45   TEMP = A0 + A2
      A2 = A0 - A2
      A0 = TEMP
      TEMP = A1 + A3
      A3 = A1 -A3
      A1 = TEMP
      TEMP = B0 + B2
      B2 = B0 - B2
      B0 = TEMP
      TEMP = B1 + B3
      B3 = B1 - B3
      B1 = TEMP
      AZ1=A0+A1
      AZ2=B0+B1
      AZ3=A0-A1
      AZ4=B0-B1
      AZ5=A2-B3
      AZ6=B2+A3
      AZ7=A2+B3
      AZ8=B2-A3
997   FORMAT (1X,4(E12.5,1X))
      A(K0) = CMPLX(AZ1,AZ2)
      A(K1) = CMPLX(AZ3,AZ4)
      A(K2) = CMPLX(AZ5,AZ6)
      A(K3) = CMPLX(AZ7,AZ8)
999   FORMAT (1X,'A COEFFICIENTS')
50    CONTINUE
C     WRITE (37,998) A(K0),A(K1),A(K2),A(K3)
C     WRITE (37,995) K0,K1,K2,K3
995   FORMAT (1X,4(I4,1X))
998    FORMAT(1X,4(E12.5,1X,E12.5,1X))
      IF (K .LE. 1) GO TO 55
```

```
      K = K - 2
      GO TO 30
55    KB = K3 + ISP
      IF (KN .LE. KB) GO TO 70
      IF (J .NE. 1) GO TO 60
      K = 3
      J = MK
      GO TO 20
60    J = J - 1
      C2 = C1
      IF (J .NE. 2) GO TO 65
      C1 = C1 * CK + S1 * SK
      S1 = S1 * CK - C2 *SK
      GO TO 35
65    C1 = (C1-S1) * SQ
      S1 = (C2 + S1) * SQ
      GO TO 35
70    CONTINUE
C     WRITE (37,998) (A(KZ),KZ=1,256)
      IF (M .LE. 1) GO TO 9005
      MP = M + 1
      JJ = 1
      IWK(1) = 1
      DO 75  I = 2,MP
      IWK(I) = IWK(I-1)*2
75    CONTINUE
      N4 = IWK(MP-2)
      IF (M .GT. 2) N8 = IWK(MP-3)
      N2 = IWK(MP-1)
      LM = N2
      NN = IWK(MP) + 1
      MP = MP - 4
      J = 2
```

```
80     JK =JJ + N2
       AK2 = A(J)
       A(J) = A(JK)
       A(JK) = AK2
       J = J+1
       IF (JJ .GT. N4) GO TO 85
       JJ = JJ + N4
       GO TO 105
85     JJ = JJ - N4
       IF (JJ .GT. N8) GO TO 90
       JJ = JJ + N8
       GO TO 105
90     JJ = JJ - N8
       K = MP
95     IF (IWK(K) .GE. JJ) GO TO 100
       JJ = JJ - IWK(K)
       K = K - 1
       GO TO 95
100    JJ = IWK(K) + JJ
105    IF (JJ .LE. J) GO TO 110
       K = NN - J
       JK = NN - JJ
       AK2 = A(J)
       A(J) = A(JJ)
       A(JJ) = AK2
       AK2 = A(K)
       A(K) = A(JK)
       A(JK) = AK2
 110   J = J + 1
       IF (J .LE. LM) GO TO 80
9005   ZX=12
       CALL TYPE(ZX)
C      WRITE (37,998) (A(KZ),KZ=1,256)
       RETURN
       END
C
C
       SUBROUTINE TYPE(ZX)
C      WRITE(37,20)ZX
20     FORMAT(/,' SUBROUTINE ENTERED ',F4.1)
       RETURN
       END

:\
```

4.5 COMPUTER SOFTWARE VERIFICATION

To assure that the software works properly, the following verification procedure has been established. Before running the program on real data, it is recommended that the program be run on an idealized, sinusoidal waveform with slight distortion added.

Two cases of distortion will be introduced; the first one involves amplitude only, and the second one involves phase only. The distortions are presented as sinusoids of various amplitudes but always small enough so as to permit the use of the paired echo theory concepts to be valid. Also, the period of the sinusoid is always harmonically related to the bandwidth of the equalizer network and as such is consistent with the concept of representing the distortion as a Fourier series.

The test cases of distortion are separated into two groups, one of which assumes only amplitude distortion and the other of which assumes only phase distortion, but of varying levels. The FFT presents the coefficients for the taps which in effect represent the amplitude and phase of the echo associated with each tap. These coefficients are then compared to the calculated values based on the paired echo theory.

CASE 1: A sinusoidal amplitude distortion of various peak to peak values.

$$|G| = 10 \log_{10} \sqrt{I^2+Q^2} = 5 \cdot \log_{10} (I^2+Q^2) = \text{amplitude}$$

$$\emptyset = \tan^{-1} \frac{Q}{I} = \text{phase}$$

To simplify, we consider the effects of only amplitude distortions so that assuming no phase distortion and letting Q = 0 we get

$$|G| = 5 \log_{10} I^2 = 10 \log_{10} I$$

$$\emptyset = \tan^{-1}\left(\frac{Q}{I}\right) = 0$$

Since a sinusoidal distortion is needed:

$$\frac{\sin(\Delta XN) - 1}{K} = 10 \log_{10} I$$

where ΔX = incremental frequency

N = *number of points*

K = scale factor (for $K = 1$, numerator goes 0 to -2 or 2 dB

and 10 log I represents the distortion level in dB (peak-to-peak). In order to fit the RADC data form so that $-128 \leq I < 128$, we multiply the given values of I by 128.

TEST 1: Amplitude distortion, 0.5 dB peak-to-peak with first echo pair

$$-0.5 = \frac{\sin(\Delta XN) - 1}{K} = 10 \log_{10} \quad \text{therefore } K = 4$$

For 215 points in 2π radians or 360°, there $\Delta X = \frac{360°}{215}$ and $0 \leq N \leq 214$*

$$I = 10\left[\frac{\sin(\Delta XN) - 1}{40}\right]$$

From the test run we get the first echo peak value distortion to be -30.44 dB. Analytically (for small amp distortion) we have:

*Note that the number of sample points is not germane to the analysis (as long as it is a large number) and relates only to the expected number of samples to be used. We later change this to 256 points.

$$20 \log_{10} \left(1+ \frac{a_1}{a_o}\right) = \text{amplitude ripple in dB (peak value)}$$

for $\frac{a_1}{a_o} << 1$; $\frac{1}{2}\ \frac{a_1}{a_o}$ = peak echo amplitude

$$20 \log_{10} \left(\frac{a_1}{2a_o}\right) = \text{echo in dB}$$

$$20 \log_{10} \left(1+ \frac{a_1}{a_o}\right) = 0.5 \text{ dB}$$

$$\frac{a_1}{2a_o} = \frac{10^{\left[\left(\frac{0.5}{20}\right)-1\right]}}{2} = 0.0296$$

$$\text{echo} = 20 \log_{10} \left(\frac{a_1}{2a_o}\right) = -30.566 \text{ dB}$$

TEST 2: Distortion is introduced in the second echo pair with an amplitude ripple of 0.5 dB

$$I = 10^{\left[\frac{(\sin \Delta x N)-1}{40}\right]}$$

where $\Delta x = \frac{2x360}{215}$ and $0 \leq N \leq 215$

From the test run we get second echo peak value to be -30.53 dB where the calculated value is the same as before -30.566 dB.

TEST 3: Distortion is introduced in the first and second echo pairs with an amplitude ripple of 0.5 dB

$$I = \left(10^{\left[\frac{\sin (\Delta x_1 N)-1}{40}\right]} + 10^{\left[\frac{\sin (\Delta x_2 N)-1}{40}\right]}\right)/2$$

where $\Delta x_1 = \frac{360}{215}$; $\Delta x_2 = \frac{2x360}{215}$ and $0 \leq N \leq 214$

Because we are summing two sinusoidal amplitude distortions of 0.5 dB each, the resultant distortion needs to be divided by 2 in order for it to be 0.5 dB.

From the test run we get both the first and second echo peak to be -36.2 dB. The calculated first and second echo peak is:

$$20 \log_{10} \left(1 + \frac{a_1}{4a_o}\right)$$

$$\frac{a_1}{4a_o} = \frac{10\left[\frac{0.5}{20}\right]}{4} = 0.0148$$

$$20 \log 10 \frac{a_1}{a_o} = 36.58 \text{ dB}$$

Tests 4 through 6 are the same as Tests 1 through 3 respectively, except the peak amplitude distortion is set at 1 dB instead of 0.5 dB (then K = 2). Test 7 is the same as Tests 4 or 5 except the distortion is introduced into the eighth echo pair. The results are given in Table 4-1.

TEST 8: Distortion is introduced in the first and eighth echo pairs with an amplitude ripple of 1 dB (K = 2)

$$I = \left(10^{\frac{\sin(\Delta X_1 N)-1}{20}} + 10^{\frac{\sin(\Delta X_2 N)-1}{20}}\right)/2$$

where $\Delta X_1 = \frac{360}{215}$; $\Delta X_2 = \frac{8x360}{215}$ and $0 \leq N \leq 214$

CASE II; A sinusoidal phase distortion of various peak-to-peak values. As defined previously the gain and phase are:

$$|G| = 10 \log_{10} \sqrt{I^2+Q^2}$$

$$\emptyset = \tan^{-1} \frac{Q}{I}$$

Now we want to have a sinusoidal phase variation, so $\emptyset = A \sin(\Delta XN)k$. To simplify, let us set $|G| = 0$ dB so that:

TABLE 4-1. TEST RESULTS OBTAINED WITH DEC-20 COMPUTER

	Peak Ampl. Distortion (dB)	Echo	Simulated* Distortion Peak (dB)	Calculated Distortion Peak (dB)
Test 1	0.5	1	-30.44	-30.566
Test 2	0.5	2	-30.53	-30.566
Test 3	0.5	1	-36.26	-36.58
Test 3	0.5	2	-36.27	-36.58
Test 4	1	1	-23.76	-24.29
Test 5	1	2	-23.85	-24.29
Test 6	1	1	-29.69	-30.31
Test 6	1	2	-30.04	-30.31
Test 7	1	8	-25.8	-24.29
Test 8	1	1	-29.77	-30.31
Test 8	1	8	-31.82	-30.31

*Actual Distortions outputted by the algorithm are in echo pairs. The value shown is an average.

(1) $10 \log_{10} [I^2+Q^2]^{1/2} = 0$ and $[I^2+Q^2]^{1/2} = 1$ or $I^2+Q^2 = 1$

(2) $\frac{Q}{I} = \tan (A \sin(\Delta XN)-k)$

$Q = I \tan (A \sin (\Delta XN)-k)$

Let $A \sin (XN)-k = \emptyset$ so that $Q = I \tan \emptyset$

From (1) we have:

$$I^2 = 1 - Q^2$$

From (2) $Q^2 = I^2 \tan^2 \emptyset = (1-Q^2) \tan^2 \emptyset = \tan^2 \emptyset - Q^2 \tan^2 \emptyset$

$$Q^2 + Q^2 \tan^2 \emptyset = \tan^2 \emptyset$$

$$Q^2 (1+\tan^2 \emptyset) = \tan^2 \emptyset$$

$$Q = \frac{\tan \emptyset}{\overline{1+ \tan^2 \emptyset}}$$

$$I = [\frac{1- \tan^2 \emptyset}{1+\tan^2 \emptyset}]^{1/2} = \frac{1}{(1+\tan^2 \emptyset)^{1/2}}$$

$$\emptyset = \tan^{-1} \frac{Q}{I} = \text{phase}$$

To simplify, let us always work in the I-V quadrant so that $Q \leq 0$ and $I \geq 0$, therefore when $\tan \emptyset \leq 0$, then $\sqrt{1 + \tan^2 \emptyset} \geq 0$. In order to fit the RADC format (2's complement) we multiply I and Q by 128.

TEST 1: Sinusoidal phase distortion of 6° peak value is introduced in the first echo pair

$\emptyset = 3 \sin (\Delta XN) - k$ where $\Delta X = \frac{360}{215}$, $0 \leq N \leq 214$ and

k = offset ≈ 42° places $\emptyset$ in fourth quadrant

$$Q = \frac{\tan(3 \sin(\Delta x N) - 42)}{\sqrt{1 + \tan^2(3 \sin(\Delta x N) - 42)}} \text{ and } I = \sqrt{1 - Q^2}$$

From computer simulation we get the first echo to be -31.49 dB. Analytically we get

$$20 \log_{10} \left(\frac{2\pi x\ b_1}{4x360}\right) = \text{echo in dB}$$

where b_1 = ripple in degrees

$$b_1 = 6^o,\ 20 \log_{10} \left(\frac{12\pi}{4x360}\right) = -31.64 \text{ dB}$$

TEST 2: Cosinusoidal phase distortion of 6^o peak value is introduced in the first echo

$$\emptyset = 3 \cos(\Delta XN) - 42$$

The computer simulation is basically the same except that for an odd (sin) distortion function the phase angle for the two echoes is 180^o apart, whereas for an even (cos) distortion function the two echoes are in phase. The opposite is true for amplitude distortion. That is, for odd function distortions the phase angle for the two echoes is in phase whereas for even distortion function the phase angle for the two echoes is 180^o apart.

TEST 3: Sinusoidal phase distortion of 22.92^o peak value is introduced in the first echo

$$\emptyset = 11.46 \sin(\Delta XN) - 42$$

Calculated result is $20 \log_{10} \left(\frac{2\pi\ x\ 22.92}{4x360}\right) = -20$ dB and the simulated result is -20.02 dB.

The results for the phase distortion test cases are shown in Table 4-2.

TABLE 4-2. TEST RESULTS OBTAINED WITH DEC-20 COMPUTER

	Peak Phase Distortion (degrees)	Echo	Simulated Distortion Phase (dB)	Calculated Distortion Phase (dB)
Test 1	6	1	-31.49	-31.64
Test 2	6	1	-31.49	-31.64
Test 3	22.92	1	-20.02	-20.0

4.5 PROGRAM TEST ON THE HP 2100A

Three tests were run on the algorithm contained in the HP 2100A computer at RADC. The first test is a single echo test involving a peak sinusoidal amplitude distortion of 0.5 dB. The second test is a double echo test involving a peak sinusoidal amplitude distortion of 0.5 dB with one sine wave at twice the frequency of the other. The third test is a single echo test involving a peak phase distortion of six degrees.

To perform the first test, a program has been implemented (now on disk cartridge 45) called "AT10". A listing of this program is given in Appendix 3.

AT10 outputs M sample points of a distorted sine wave given by:

$$I = 10^{\frac{[\sin \Delta XN]^{-1}}{40}}$$

where: $\Delta X = \frac{360^{\circ}}{M}$, and $N = 0,1,2,\ldots M$

In our case, we have run AT10 with 256 sample points ($M = 255$). As mentioned, the phase component of this test is Ø. AT10 produces both the single echo pair and two pair test cases. The case just described (one echo pair) is printed out in a data file called "ADØØØ"; this file contains 256 octal mode (06 integers, in two records, 128 numbers each; this file is the sequential output of the "I" or amplitude function as N goes from 1 to 256.

To run the double echo-pair, amplitude-distortion-only case, we executed AT10 again but such that AT10 adds to the first waveform, a second one given by

$$I_2 = 10 \frac{\sin(\Delta X_2 N)-1}{40}$$

where: $\Delta X_2 = \frac{2 \times 360}{M}$, and $N = 0,1,2,\ldots.M = 255$

The superimposed waveform (this is what is printed out into the data file) is

$$I_{sup} = (I + I_2)/2$$

Again, there are 256 samples, written into two 128-number records in 06 format. The file containing I_{sup} is called "ADØØ1". This double echo-pair that is equivalent to Test 3 of Table 4-1 which shows the theoretical peak distortion to be -36.58 dB.

For the third case, phase distortion only, the program "TEST 1" was executed. The waveform produced by "TEST 1" is a sine whose phase is given by:

$$Q = \frac{\tan\,(3\,\sin(\Delta X_1 N)-42)}{1+\tan^2\,(3\,\sin(\Delta X_1 N)-42)}, \quad X = \frac{360}{256}, \; N = 0,1,2,\ldots.255$$

To run this test case, we executed "TEST 1" (also on disk 45), and used the data file "ADØØ2". This results in a test case of 6° phase distortion equivalent to Test 1 in Table 4-2.

Tables 4-3 through 4-5 give printouts of the data contained in the output files for these three test cases, "ADØØØ", "ADØØ1", and "ADØØ2" respectively.

We can now compare the results obtained with the test cases run on the DEC-20 and HP 2100A computers with their respective equalizer algorithms. Slight differences are to be expected because of the differences in precision and the internal mathematical algorithms. Table 4-6 shows the results.

The comparison shows excellent agreement between the two equalizer algorithm outputs (DEC-20 and HP 2100A) and demonstrates that they are performing the way they are supposed to.

TABLE 4-3. OUTPUT TEST 1 (ONE ECHO PAIR)

File AD000

Tap No.	Attenuator (dB)	(Turns)	Line Stretcher (deg)	(Div)
8A	-.59754E+02	9.924	.77894E+03	7.2
7A	-.57500E+02	9.133	.84826E+03	7.8
6A	-.50899E+02	9.931	.72472E+03	6.7
5A	-.54626E+02	9.280	.12426E+04	11.0
4A	-.76775E+02	9.870	.13916E+03	1.3
3A	-.57659E+02	9.054	.48888E+03	4.5
2A	-.63654E+02	8.213	.82400E+03	7.6
1A	-.30575E+02	2.894	.94832E+03	8.8
0	.95424E+01	0.000	.44309E+03	4.1
1B	-.30102E+02	2.571	.40519E+02	0.4
2B	-.58178E+02	8.552	.60589E+03	5.6
3B	-.53561E+02	9.354	.47294E+03	4.4
4B	-.57641E+02	9.961	.80104E+03	7.4
5B	-.52656E+02	9.388	.15492E+03	1.4
6B	-.68077E+02	9.716	.23711E+03	2.2
7B	-.55111E+02	9.231	.61405E+03	5.7
8B	-.58767E+02	9.938	.91032E+03	8.4

TABLE 4-4. OUTPUT TEST 2 (DOUBLE ECHO PAIR)

File AD001

Tap No.	Attenuator (dB)	(Turns)	Line Stretcher (deg)	(Div)
8A	-.58976E+02	9.646	.78231E+03	7.2
7A	-.58854E+02	9.603	.86500E+03	8.0
6A	-.50607E+02	9.816	.85919E+03	7.9
5A	-.55226E+02	9.481	.90270E+03	8.4
4A	-.74326E+02	9.505	.32962E+03	3.0
3A	-.56551E+02	8.710	.36137E+03	3.3
2A	-.36679E+02	3.518	.93857E+02	8.7
1A	-.36044E+02	3.898	.95012E+03	8.8
0	.95424E+01	0.000	.44309E+03	4.1
1B	-.35637E+02	3.589	.39872E+03	3.7
2B	-.35844E+02	2.950	.49132E+03	4.5
3B	-.52810E+02	9.094	.60045E+03	5.6
4B	-.57316E+02	9.856	.97058E+03	9.0
5B	-.53165E+02	9.563	.13478E+03	1.2
6B	-.66063E+02	9.314	.10264E+03	0.9
7B	-.56190E+02	9.654	.59731E+03	5.5
8B	-.58025E+02	9.706	.90695E+03	8.4

TABLE 4-5. OUTPUT CASE 3 (PHASE DISTORTION ONLY)

File AD004

Tap No.	Attenuator (dB)	(Turns)	Line Stretcher (deg)	(Div)
8A	-.59834E+02	9.953	.79680E+03	7.4
7A	-.59511E+02	9.837	.73877E+03	6.8
6A	-.50643E+02	9.831	.96246E+03	8.9
5A	-.56282E+02	9.841	.11579E+04	10.7
4A	-.72772E+02	9.269	.38686E+03	3.6
3A	-.57695E+02	9.066	.39645E+03	3.7
2A	-.66269E+02	8.720	.67455E+03	6.2
1A	-.31102E+02	2.984	.10384E+04	9.6
0	.95424E+01	0.000	.44309E+03	4.1
1B	-.31104E+02	2.742	.17235E+03	1.6
2B	-.58022E+02	8.500	.44508E+03	4.1
3B	-.53881E+02	9.466	.78023E+03	7.2
4B	-.57201E+02	9.818	.86518E+03	8.0
5B	-.53831E+02	9.794	.10693E+03	1.0
6B	-.68513E+02	9.803	.20575E+03	1.9
7B	-.56951E+02	9.961	.33624E+03	3.1
8B	-.57922E+02	9.674	.66682E+03	6.2

TABLE 4-6. COMPARISON OF TEST RESULTS FOR THE DEC-20 AND HP 2100A EQUALIZER ALGORITHMS

Test	Type of Distortion	Echo Affected	Simulated Echo Level in dB HP 2100A	Simulated Echo Level in dB DEC-20	Theoretical Level in dB
Test 1,	0.5 dB	1A	-30.58	-30.44	-30.57
One Echo Pair	Peak Amplitude Only	1B	-30.10	(AVG)	-30.57
Test 2,		1A	-36.04	36.36	-36.58
Double Echo Pair	0.5 dB	1B	-35.64	(AVG)	-36.58
		2A	-36.68		-36.58
		2B	-35.84		-36.58
Test 3,	6^{o} Peak	1A	-31.10	-31.49	-31.64
	Phase Distortion Only	1B	-31.10	(AVG)	-31.64

While this is not 100% conclusive, it does provide a high degree of confidence that the algorithm is working properly. The only real test is the use of the algorithm over many real situations which completely exercise it. Some comments about what to expect are in order.

First of all, the Microwave Transversal Equalizer (MTE) to be used to provide corrections to the distortion has its own iherent second order distortions which are not constant. That is, under one set of amplitude and phase adjustments for each of the taps there is a given "self-distortion" which is included in the data supplied to the algorithm to analyze. When the amplitude and phase settings are now changed to correct for the total distortion reflected in the data, then the new settings of the MTE will correct much of the distortion but because the MTE settings are different from the original settings there will be a new "self-distortion" introduced by the MTE. This means that the correction is imperfect and the process must be iterated until the distortion residue is less than the allowable value (or until the "self-distortion" changes are greater than the corrected indicated by the algorithm).

Another source of distortion lies in the hardware used to provide the data samples. These distortions are:

1. the phase response of the components
2. the amplitude response of the components
3. noise introduced on the data (Gaussian)
4. random spikes (transients induced by interference)

5. sinusoidal variations (coupled interference from motors, oscillators, etc.)

6. timing errors (clocks, circuit switching speeds, etc.)

7. inaccuracies in data samples (improper scaling, etc.)

8. digitizing errors (bit errors in digital operations)

These errors introduce their own "self-distortion" and just as in the MTE limit the residual distortion that can be obtained. In addition, some of the errors are random so that iterations won't help.

Finally, there is the algorithm itself. It cannot be considered fully debugged until all combinations of operation have been exercised. This is not a practical thing to do in the laboratory but instead requires a perigd of field use. In addition, the algorithm has limits of precision which will ultimately limit the amount of distortion that can be corrected even if the hardware were perfect, although this is the area that is least likely to present a problem.

5. CONCLUSIONS AND RECOMMENDATIONS

Computer algorithms have been successfully developed to provide open-loop adaptive control of the MTE. The algorithms are based upon the application of FFT techniques, and the necessary corallary software programming procedures have been developed and described.

Verification of the aforementioned FFT and associated software programming has also been accomplished at AIL with a DEC-20 computer. Such verification has involved the generation of output adjustment data for MTE control based upon computer analysis of artificially simulated time sidelobe input distortion levels. A similar procedure will be undertaken at RADC using real-time I and Q data acquired at the A. Froelich High Power Tube Facility. Such distortion levels will be processed at RADC using translation of format programs between the AIL DEC-20 and the RADC HP 2100A computers. Preliminary data suggests that the post-delivery program translation will be successful after the normal debugging procedures have been completed.

AIL recommendations for future MTE related efforts include the following:

- Continuation and/or extension of contracted efforts to include AIL post-delivery support for RADC.

- Investigation and definition of suitable closed-loop techniques and procedures for adaptive operation of MTE.
- Development of suitable solid-state time delay devices to replace the present manually adjustable line stretchers.
- Modification of the MTE to expand capability by providing 32 tap operation.
- Development of suitable electronic interface to implement fully adaptive closed-loop operation of the MTE.

APPENDIX 1

SAMPLE COMPUTER RUN WITH
INPUT DATA

APPENDIX 1: SAMPLE RUN

This appendix includes a sample run of the program with the input data entitled ADØØØ.

```
:RU,AIL
1

                INITIAL SETTINGS AT STEP 0 SHOULD BE:


ATTENUATOR                                       LINE STRETCHER

1 A=10.000                                         1 A= 7.9
2 A=10.000                                         2 A= 6.2
3 A=10.000                                         3 A= 3.7
4 A=10.000                                         4 A= 1.5
5 A=10.000                                         5 A= 9.1
6 A=10.000    Initial                              6 A= 7.5    Initial
7 A=10.000    Attenuator                           7 A= 7.0     Line
8 A=10.000    Settings                             8 A= 6.1    Strecher
 M = 0.000                                          M = 4.1    Settings
1 B=10.000                                         1 B= 1.2
2 B=10.000                                         2 B= 3.7
3 B=10.000                                         3 B= 5.2
4 B=10.000                                         4 B= 7.2
5 B=10.000                                         5 B=  .5
6 B=10.000                                         6 B= 1.4
7 B=10.000                                         7 B= 3.2
8 B=10.000                                         8 B= 6.2
```

APPENDIX 1   SAMPLE RUN (continued

```
 ENTER THE FILE NAME         (Attenuator in dB Units)
AD000
   -.58372E+02  -.59486E+02  -.50322E+02  -.55731E+02  -.67727E+02  -.59106E+02
 -.52821E+02  -.24613E+02
   .95424E+01  -.23590E+02  -.50158E+02  -.54445E+02  -.55841E+02  -.53580E+02
-.64164E+02  -.56657E+02
  -.57362E+02
    .65446E+03   .67758E+03   .96441E+03   .92669E+03   .17776E+03   .31625E+03
  .84035E+03   .94840E+03
   .44309E+03   .40437E+02   .58954E+03   .64557E+03   .76244E+03   .11079E+03
 .35742E+03   .42473E+03
   .67480E+03
```

APPENDIX 1 SAMPLE RUN (continued)

```
                    INTERMEDIATE SETTINGS AT STEP 1 ARE:

          ATTENUATOR                                   LINE STRETCHER

          1 A= 1.976                                      1 A= 8.8
          2 A= 6.202                                      2 A= 7.8
          3 A= 9.519                                      3 A= 2.9
          4 A= 8.486                                      4 A= 1.6
          5 A= 9.669                                      5 A= 8.6
          6 A= 9.705                                      6 A= 8.9
          7 A= 9.828                                      7 A= 6.3
          8 A= 9.434                                      8 A= 6.1
           M = 0.000                                       M = 4.1
          1 B= 1.573                                      1 B=  .4
          2 B= 6.085                                      2 B= 5.5
          3 B= 9.665                                      3 B= 6.0
          4 B= 9.368                                      4 B= 7.1
          5 B= 9.707                                      5 B= 1.0
          6 B= 8.934                                      6 B= 3.3
          7 B= 9.842                                      7 B= 3.9
          8 B= 9.502                                      8 B= 6.2

   AFTER  0 ITERATIONS DISTORTION CAN NOT,BE COMPENSATED

 AIL : STOP
```

Note: XD = 20 in $AT10 when generating input for this run.

In subsequent runs, XD = 40

APPENDIX 2

PROGRAM TO GENERATE
TEST DATA

APPENDIX 2. PROGRAM TO GENERATE TEST DATA

This routine generates ideal sinusoidal distortion as the data file for the program.

```
FTN4,L
      PROGRAM AT10
      DIMENSION IBUF2(256),IBUF3(256),MBUF(256)
      DIMENSION IDCB(272),NAM(3),IBUF0(256),ISIZE(2)
      DIMENSION LBUF(256),IBUF1(256)
      DIMENSION IDCB2(272),NAM1(3)
      ITYPE=2
      ISIZE=2
      ISIZE(2)=128
      DATA NAM/ 2HAD,2H00,2H0 /
      DATA NAM1/2HAD,2H00,2H1 /
      CALL CREAT (IDCB,IERR,NAM,ISIZE,ITYPE)
      CALL CREAT (IDCB1,IERR,NAM1,ISIZE,ITYPE)
      N=256
      XD=40.
      DX1=360./FLOAT(N-1)
      DX2=DX1*2.
      DO 20 J=1,N
      X=DX1*FLOAT(J-1)
      Y=DX2*FLOAT(J-1)
      X1=(10.**((SIN(X*3.1415/180.)-1)/XD))
      X2=(10.**((SIN(Y*3.1415/180.)-1)/XD))
      LBUF(J)=128*X1
      XM=(X1+X2)/2.
      MBUF(J)=128*XM
      IF (MBUF(J).GE.128) MBUF(J)=127
      MBUF(J)=256*MBUF(J)
      IF(LBUF(J).EQ.128)LBUF(J)=127
      LBUF(J)=256*LBUF(J)
 20   CONTINUE
      DO 30 I=1,128
      IBUF0(I)=LBUF(I)
      IBUF2(I)=MBUF(I)
 30   CONTINUE
      DO 40 K=1,128
      IBUF1(K)=LBUF(128+K)
      IBUF3(K)=MBUF(128+K)
 40   CONTINUE
      CALL WRITF(IDCB,IERR,IBUF0)
      CALL WRITF(IDCB,IERR,IBUF1)
      CALL WRITF(IDCB1,IERR,IBUF2)
      CALL WRITF(IDCB1,IERR,IBUF3)
      CALL CLOSE(IDCB,IERR)
      CALL CLOSE (IDCB1,IERR)
      STOP
      END
```

This program generates a sinusoid with phase distortion only.

```
LIST,$TEST1::45
$TEST1 T=00004 IS ON CR00045 USING 00005 BLKS R=0034
AVL3L1   FTN4
              PROGRAM TEST1
              INTEGER IDCB(272),NAM(3),IBUF0(256),IBUF1(256),LBUF(256)
              INTEGER ISIZE(2)
              ITYPE=2
              N=256
              ISIZE=2
              ISIZE(2)=128
C             CALL CREAT(IDCB,IERR,NAM,ISIZE,ITYPE)
              DATA NAM/ 2HAD,2H00,2H4 /
              CALL CREAT(IDCB,IERR,NAM,ISIZE,ITYPE)
              DX1=360./FLOAT(N-1)
              DO 10 I=1,N
              X1=DX1*FLOAT(I-1)
              X=3.*SIN(X1*3.1415/180.)-42.
              X2=SIN(X*3.1415/180.)
              X3=COS(X*3.1415/180.)
              T=X2/X3
              T2=T*T
              X4=SQRT(1+T2)
              IF(T.GT.0) X4=-X4
              Q=T/X4
              IQ=128.*Q
              XI=SQRT(1.-Q*Q)
              II=128*XI
              IF(II.GE.127) II=127
              IF (II.LE.-128) II=-128
              IF (IQ.GE.127) IQ=127
              IF (IQ.LE.-128) IQ=-128
              IF(II.LT.0) II=II+256
              IF (IQ.LT.0) IQ=IQ+256
              LBUF(I)=256*II+IQ
     10       CONTINUE
              DO 30 J=1,128
              IBUF0(J)=LBUF(J)
              IBUF1(J)=LBUF(J+128)
     30       CONTINUE
              CALL WRITF(IDCB,IERR,IBUF0)
              CALL WRITF(IDCB,IERR,IBUF1)
              CALL CLOSE(IDCB,IERR)
              STOP
```

Input for Case 1: 256 samples of sinusoid listed from sequential file, in decimal

30720	30976	30976	30976	30976	30976	30976	31232
31232	31232	31232	31232	31232	31488	31488	31488
31488	31488	31488	31744	31744	31744	31744	31744
31744	31744	[illegible]	32000	32000	32000	32000	32000
32000	32256	32256	32256	32256	32256	32256	32256
32256	32256	32256	32512	32512	32512	32512	32512
32512	32512	32512	32512	32512	32512	32512	32512
32512	32512	32512	32512	32512	32512	32512	32512
32512	32512	32512	32512	32512	32512	32512	32512
32512	32512	32512	32512	32512	32512	32512	32512
32512	32512	32512	32512	32512	32512	32256	32256
32256	32256	32256	32256	32256	32256	32256	32000
32000	32000	32000	32000	32000	32000	31744	31744
31744	31744	31744	31744	31744	31488	31488	31488
31488	31488	31488	31232	31232	31232	31232	31232
31232	30976	30976	30976	30976	30976	30976	30720
30720	30720	30720	30720	30720	30464	30464	30464
30464	30464	30464	30208	30208	30208	30208	30208
30208	29952	29952	29952	29952	29952	29952	29952
29696	29696	29696	29696	29696	29696	29696	29696
29440	29440	29440	29440	29440	29440	29440	29440
29440	29440	29184	29184	29184	29184	29184	29184
29184	29184	29184	29184	29184	29184	29184	29184
29184	29184	29184	29184	29184	29184	29184	29184
29184	29184	29184	29184	29184	29184	29184	29184
29184	29184	29184	29184	29184	29184	29184	29184
29184	29184	29184	29184	29184	29440	29440	29440
29440	29440	29440	29440	29440	29440	29440	29440
29696	29696	29696	29696	29696	29696	29696	29952
29952	29952	29952	29952	29952	29952	29952	30208
30208	30208	30208	30208	30208	30464	30464	30464
30464	30464	30464	30720	30720	30720	30720	30720

Input for Case 2:

30720	30976	30976	30976	30976	31232	31232	31232
31232	31488	31488	31488	31488	31744	31744	31744
31744	31744	32000	32000	32000	32000	32000	32000
32256	32256	32256	32256	32256	32256	32256	32256
32256	32256	32512	32512	32512	32512	32512	32512
32512	32512	32512	32256	32256	32256	32256	32256
32256	32256	32256	32256	32256	32256	32000	32000
32000	[illegible]	[illegible]	[illegible]	32000	31744	31744	31744
31744	31744	31744	31488	31488	31488	31488	31488
31232	31232	31232	31232	31232	31232	30976	30976
30976	30976	30976	30976	30976	30720	30720	30720
30720	30720	30720	30720	30720	30720	30720	30720
30464	30464	30464	30464	30464	30464	30464	30464
30464	30464	30464	30464	30464	30464	30464	30464
30464	30464	30464	30464	30464	30720	30720	30720
30720	30720	30720	30720	30720	30720	30720	30720
30720	30720	30976	30976	30976	30976	30976	30976
30976	30976	30976	30976	30976	30976	30976	31232
31232	31232	31232	31232	31232	31232	31232	31232
31232	31232	31232	31232	31232	31232	31232	31232
30976	30976	30976	30976	30976	30976	30976	30976
30976	30976	30720	30720	30720	30720	30720	30720
30720	30464	30464	30464	30464	30464	30464	30208
30208	30208	30208	30208	30208	29952	29952	29952
29952	29952	29696	29696	29696	29696	29696	29696
29696	29440	29440	29440	29440	29440	29440	29440
29440	29440	29440	29440	29184	29184	29184	29184
29184	29184	29184	29184	29184	29184	29184	29440
29440	29440	29440	29440	29440	29440	29440	29440
29440	29696	29696	29696	29696	29696	29696	29952
29952	29952	29952	29952	30208	30208	30208	30208
30464	30464	30464	30464	30720	30720	30720	30720

STOP

Input for Case 1: Octal Format

```
:LIST,AD000
AD000  T=00002 IS ON CR00002 USING 00002 BLKS R=0128

REC# 00001

074000 074400 074400 074400 074400 074400 074400 075000◆
075000 075000 075000 075000 075000 075400 075400 075400◆
075400 075400 075400 076000 076000 076000 076000 076000◆
076000 076000 076400 076400 076400 076400 076400 076400◆
076400 076400 077000 077000 077000 077000 077000 077000◆
077000 077000 077000 077400 077400 077400 077400 077400◆
077400 077400 077400 077400 077400 077400 077400 077400◆
077400 077400 077400 077400 077400 077400 077400 077400◆
077400 077400 077400 077400 077400 077400 077400 077400◆
077400 077400 077400 077400 077400 077400 077400 077400◆
077400 077400 077400 077400 077400 077400 077000 077000◆
077000 077000 077000 077000 077000 077000 077000 076400◆
076400 076400 076400 076400 076400 076400 076000 076000◆
076000 076000 076000 076000 076000 075400 075400 075400◆
075400 075400 075400 075000 075000 075000 075000 075000◆
075000 074400 074400 074400 074400 074400 074400 074000◆

REC# 00002

074000 074000 074000 074000 074000 073400 073400 073400◆
073400 073400 073400 073000 073000 073000 073000 073000◆
073000 072400 072400 072400 072400 072400 072400 072400◆
072000 072000 072000 072000 072000 072000 072000 072000◆
071400 071400 071400 071400 071400 071400 071400 071400◆
071400 071400 071000 071000 071000 071000 071000 071000◆
071000 071000 071000 071000 071000 071000 071000 071000◆
071000 071000 071000 071000 071000 071000 071000 071000◆
071000 071000 071000 071000 071000 071000 071000 071000◆
071000 071000 071000 071000 071000 071000 071000 071000◆
071000 071000 071000 071000 071000 071400 071400 071400◆
071400 071400 071400 071400 071400 071400 071400 072000◆
072000 072000 072000 072000 072000 072000 072000 072400◆
072400 072400 072400 072400 072400 072400 073000 073000◆
073000 073000 073000 073000 073000 073400 073400 073400◆
073400 073400 073400 074000 074000 074000 074000 074000◆
```

Input for Case 2: Octal Format

```
LIST,AD001
AD001  T=00002 IS ON CR00002 USING 00002 BLKS R=0128

REC# 00001

074000 074400 074400 074400 074400 075000 075000 075000◆
075000 075400 075400 075400 075400 076000 076000 076000◆
076000 076000 076400 076400 076400 076400 076400 076400◆
077000 077000 077000 077000 077000 077000 077000 077000◆
077000 077000 077400 077400 077400 077400 077400 077400◆
077400 077400 077400 077000 077000 077000 077000 077000◆
077000 077000 077000 077000 077000 077000 076400 076400◆
076400 076400 076400 076400 076400 076000 076000 076000◆
076000 076000 076000 075400 075400 075400 075400 075400◆
075000 075000 075000 075000 075000 075000 074400 074400◆
074400 074400 074400 074400 074400 074000 074000 074000◆
074000 074000 074000 074000 074000 074000 074000 074000◆
073400 073400 073400 073400 073400 073400 073400 073400◆
073400 073400 073400 073400 073400 073400 073400 073400◆
073400 073400 073400 073400 073400 074000 074000 074000◆
074000 074000 074000 074000 074000 074000 074000 074000◆

REC# 00002

074000 074000 074400 074400 074400 074400 074400 074400◆
074400 074400 074400 074400 074400 074400 074400 075000◆
075000 075000 075000 075000 075000 075000 075000 075000◆
075000 075000 075000 075000 075000 075000 075000 075000◆
074400 074400 074400 074400 074400 074400 074400 074400◆
074400 074400 074000 074000 074000 074000 074000 074000◆
074000 073400 073400 073400 073400 073400 073400 073000◆
073000 073000 073000 073000 073000 072400 072400 072400◆
072400 072400 072400 072000 072000 072000 072000 072000◆
072000 071400 071400 071400 071400 071400 071400 071400◆
071400 071400 071400 071400 071000 071000 071000 071000◆
071000 071000 071000 071000 071000 071000 071000 071400◆
071400 071400 071400 071400 071400 071400 071400 071400◆
071400 072000 072000 072000 072000 072000 072000 072400◆
072400 072400 072400 072400 073000 073000 073000 073000◆
073400 073400 073400 073400 074000 074000 074000 074000◆
```

Input for Case 3: Octal Format

```
LIST,AD004
AD004  T=00002 IS ON CR00002 USING 00002 BLKS R=0128

REC# 00001

057653 057653 057653 057653 057653 057653 057654 057654◆
057654 060254 060254 060254 060254 060254 060255 060255◆
060255 060255 060655 060655 060655 060655 060655 060656◆
060656 060656 060656 060656 060656 061256 061256 061256◆
061256 061257 061257 061257 061257 061257 061257 061257◆
061257 061257 061257 061257 061257 061657 061657 061660◆
061660 061660 061660 061660 061660 061660 061660 061660◆
061660 061660 061660 061660 061660 061660 061660 061660◆
061660 061660 061660 061660 061660 061660 061660 061660◆
061660 061660 061660 061660 061660 061660 061660 061660◆
061660 061657 061657 061657 061257 061257 061257 061257◆
061257 061257 061257 061257 061257 061257 061257 061256◆
061256 061256 061256 060656 060656 060656 060656 060656◆
060656 060656 060655 060655 060655 060655 060255 060255◆


060255 060255 060254 060254 060254 060254 060254 060254◆
OP CODE ERR
057654 057654 057654 057653 057653 057653 057653 057653◆

REC# 00002

057653 057253 057253 057252 057252 057252 057252 057252◆
057252 057252 056652 056651 056651 056651 056651 056651◆
056651 056651 056651 056651 056250 056250 056250 056250◆
056250 056250 056250 056250 056250 056250 056250 055647◆


055647 055647 055647 055647 055647 055647 055647 055647◆
OP CODE ERR
055647 055647 055647 055647 055647 055246 055246 055246◆
055246 055246 055246 055246 055246 055246 055246 055246◆
055246 055246 055246 055246 055246 055246 055246 055246◆
055246 055246 055246 055246 055246 055246 055246 055246◆
055246 055246 055246 055246 055246 055246 055246 055246◆
055246 055246 055646 055647 055647 055647 055647 055647◆
055647 055647 055647 055647 055647 055647 055647 055647◆
055647 056250 056250 056250 056250 056250 056250 056250◆
056250 056250 056250 056251 056651 056651 056651 056651◆
056651 056651 056651 056651 056652 057252 057252 057252◆
057252 057252 057252 057252 057252 057253 057653 057653◆
```

Input for Case 3:
Decimal Format

24491	24491	24491	24491	24491	24491	24492	24492
24492	24748	24748	24748	24748	24748	24749	24749
24749	24749	25005	25005	25005	25005	25005	25006
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25262	25263	25263	25263	25263	25263	25263	25263
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25520	25520	25520	25520	25520	25520	25520	25520
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25520	25520	25520	25520	25520	25520	25520	25520
25520	25520	25520	25520	25520	25520	25520	25520
25520	25519	25519	25519	25263	25263	25263	25263
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25262	25262	25262	25006	25006	25006	25006	25006
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24749	24749	24748	24748	24748	24748	24748	24748
24492	24492	24492	24491	24491	24491	24491	24491
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24234	24234	23978	23977	23977	23977	23977	23977
23977	23977	23977	23977	23720	23720	23720	23720
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23206	23205	23462	23463	23463	23463	23463	23463
23463	23463	23463	23463	23463	23463	23463	23463
23463	23720	23720	23720	23720	23720	23720	23720
23720	23720	23720	23721	23977	23977	23977	23977
23977	23977	23977	23977	23978	24234	24234	24234
24234	24234	24234	24234	24234	24235	24491	24491 STOP

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