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DIGITAL PULSE COMPRESSION TECHNIQUES.(U)  
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DIGITAL PULSE COMPRESSION TECHNIQUES

by

Peter N. Marinos  
Department of Electrical Engineering

Prepared Under:  
Office of Naval Research  
Contract No. N00014-67-A-0251-0023 *new*

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DUKE UNIVERSITY

Department of Electrical Engineering  
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⑩ Peter N. Marinos  
⑪ Jan 1977

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Prepared under: Office of Naval Research

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# ABSTRACT

The purpose of this work was to identify and study pulse compression techniques useful in LPI (i.e., Low Probability of Intercept) applications. More specifically, the technique proposed should be capable of compression ratios of from 128 to 1024 or higher, and integrated sidelobes of -35 to -30 db throughout the compression range; furthermore, it should be easy to mechanize, and should result in minimal S/N loss. It was concluded that nonlinear FM is the pulse-compression scheme capable of satisfying all these requirements, and existing technologies make its mechanization feasible.

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1. Nonlinear FM Pulse Compression System

## I. INTRODUCTION

The probability of detection of a radar signal in the presence of white Gaussian noise is maximized if the receiver is a matched filter to the signal [1]. In this case the receiver response becomes the signal autocorrelation function. In certain radar applications, it is also necessary that the transmitted waveform be rectangular and have as low an amplitude as possible but always consistent with the requirements for range resolution and probability of detection. These signal design constraints are best satisfied by using pulse compression techniques which will result in the minimal possible signal-to-noise power loss.

Nonlinear FM signals have relatively low time sidelobes and result in no loss in signal-to-noise ratio for as long as the receiver is matched to the transmitted nonlinear FM waveform. The compressed pulse, however, is highly sensitive to frequency variations, and becomes extremely distorted and develops high sidelobes as the doppler frequency is increased.

Since nonlinear FM suffers no signal-to-noise ratio loss under perfect match conditions, it becomes a serious candidate for use in a pulse compression system. In the sequel, a digital pulse-compression system is proposed based on nonlinear FM and capable of operating over an arbitrarily wide band of doppler frequencies without violating the requirements of specified sidelobe level and signal-to-noise ratio degradation.

## II. NONLINEAR FM PULSE COMPRESSION SIGNALS

Consider a narrow band signal  $s(t)$  centered at a carrier frequency  $f_c$  with an envelope  $a(t)$ , and a phase modulation given by  $\phi(t)$ . That is,

$$\begin{aligned} s(t) &= a(t)\cos[2\pi f_c t + \phi(t)], \quad -T/2 \leq t \leq T/2 \\ &= 0, \quad \text{elsewhere} \end{aligned} \quad (1)$$

where  $T$  is the pulse length. The equivalent complex representation of the modulation is given by

$$\tilde{a}(t) = a(t) \exp[j\phi(t)] \quad (2)$$

and its corresponding Fourier transform is expressed as

$$\begin{aligned} \tilde{A}(f) &= \int_{-\infty}^{\infty} a(t) \exp[j\{\phi(t) - 2\pi ft\}] dt \\ &= A(f) \exp[j\theta(f)] \end{aligned} \quad (3)$$

where  $A(f)$  denotes the modulus of  $\tilde{A}(f)$ . In a similar manner, one may also write

$$\begin{aligned} a(t) &= \int_{-\infty}^{\infty} A(f) \exp[j\{\theta(f) + 2\pi ft\}] df \\ &= a(t) \exp[j\phi(t)] \end{aligned} \quad (3a)$$

where  $a(t)$  is the modulus of  $\tilde{a}(t)$ .

The moduli  $a(t)$  and  $A(f)$  are related via Parseval's theorem which is analytically expressed by

$$\int_{-\infty}^{\infty} a^2(t) dt = \int_{-\infty}^{\infty} A^2(f) df = 2E \quad (4)$$

where  $E$  is the energy in the real signal.

The signal autocorrelation function,  $z(t)$ , which appears at the output of the matched filter receiver, is given by

$$z(t) = \int_{-\infty}^{\infty} \tilde{a}(\tau) \tilde{a}^*(t+\tau) d\tau \quad (5)$$

where  $\tilde{a}^*(t)$  is the complex conjugate of  $\tilde{a}(t)$ . Using the Fourier transform of  $\tilde{a}(\tau)$ , one may rewrite Eq. (5) as

$$z(t) = \int_{-\infty}^{\infty} A^2(f) \exp[j2\pi ft] df \quad (6)$$

or

$$A(f) = \left\{ \int_{-\infty}^{\infty} z(t) \exp[-j2\pi ft] dt \right\}^{1/2} \quad (7)$$

The general problem which includes the nonlinear FM case as a special case may be stated as follows:

"Given the two moduli  $a(t)$  and  $A(f)$ , is it possible to obtain a pair of phase functions  $\phi(t)$  and  $\theta(f)$  such that equation (3) (or 3a) is satisfied?"

With the exception of very few well-known cases, the general exact solution to this problem for arbitrary functions  $a(t)$  and  $A(f)$  is not known. This stems from the fact that Fourier theory does not provide a general set of necessary and sufficient conditions which must be satisfied by the moduli of a potential Fourier transform pair, that is, by  $a(t)$  and  $A(f)$  in this case. (Note, once again, that the reference here is to  $a(t)$  and  $A(f)$  and not to  $\tilde{a}(t)$  and  $\tilde{A}(f)$ .) Nevertheless, one may still insist on an approximate solution and accept it or reject it depending on how well it satisfies expressions (3) and (3a). Such approximate solutions may be pursued by making use of the stationary phase method of approximate integration [2]. This method may be applied either to expression (3) or (3a).

According to the principle of stationary phase, the major contribution to the integral of a rapidly oscillating function will result from regions over which the phase of the integrand functions is "stationary"; that is, from regions over which the derivative of the phase function is zero.

Applying this technique to expression (3), one has

$$\frac{d}{dt} [\phi(t) - 2\pi ft] \equiv 0$$

or  $\phi'(t) = 2\pi f$  (8)

Expression (8) implies that the points in time,  $t$ , at which the total phase becomes "stationary" will depend on the frequency  $f$ .

One obtains a similar relationship using expression (3a). That is,

$$\frac{d}{df} [\theta(f) + 2\pi ft] \equiv 0$$

or  $\theta'(f) = -2\pi t$  (9)

Expression (9) implies that the frequencies,  $f$ , at which the total phase becomes "stationary" will depend on the time,  $t$ .

It becomes apparent that in order to obtain  $\phi(t)$ , it is necessary that we determine the functional dependence of  $f$  on  $t$ , or

$$f = g(t) \quad (9a)$$

Use of (9a) in (8) and subsequent integration will provide the phase  $\phi(t)$ .

Similarly, one may use the inverse of (9a), that is,

$$t = g^{-1}(f) \quad (9b)$$

and expression (9) to determine  $\theta(f)$ .

Determining the functional dependence expressed by (9a) or (9b) is the ultimate objective of the stationary phase approximation method. This is illustrated by considering expression (3a) and expanding its phase function in a 3-term Taylor's series about the stationary point, say,  $f = \epsilon$ , and performing the indicated operations. Key et al. [3] have shown that this yields the approximate expression

$$\tilde{a}(t) \cong \sqrt{2\pi} \frac{A(\epsilon)}{\sqrt{|\theta''(\epsilon)|}} \exp[j\{2\pi\epsilon t + \theta(\epsilon) \pm \pi/4\}] \quad (10)$$

where (+) is associated with  $\theta''(\epsilon) > 0$  and (-) with  $\theta''(\epsilon) < 0$ . Expression (10) for  $a(t)$  may be separated into the modulus and phase functions as follows:

$$a(t) = \sqrt{2\pi} \frac{A(f)}{\sqrt{|\theta''(f)|}} \quad (11)$$

and

$$\phi(t) = 2\pi ft + \theta(f) \pm \pi/4 \quad (12)$$

where  $\epsilon$ , denoting the stationary point in frequency, has been replaced by  $f$  for clarity and in being consistent with expressions (8) and (9).

Differentiation of expression (9) and subsequent substitution in (11) yields

$$a^2(t) dt = A^2(f) df \quad (13)$$

which in a sense establishes in differential form the dependence of  $f$  on  $t$  or vice versa. Furthermore, expression (13) prescribes the manner in which the moduli  $a(t)$  and  $A(f)$  enter in this dependence.

According to Parseval's theorem (see expression (4)), Eq. (13) must yield

$$\int_{-\infty}^{\infty} a^2(t) dt = \int_{-\infty}^{\infty} A^2(f) df \quad (14)$$

But more importantly, given time  $t$  (or frequency  $f$ ) we want to know the corresponding frequency  $f$  (or time  $t$ ) such that

$$\int_{-\infty}^t a^2(y) dy = \int_{-\infty}^f A^2(x) dx \quad (15)$$

or equally valid

$$\int_{-\infty}^t a^2(y) dy = \int_f^{\infty} A^2(x) dx \quad (16)$$

Equations (15) and (16) provide two possible solutions regarding the dependence of  $f$  on  $t$ , or vice versa, given two moduli functions  $a(y)$  and  $A(x)$ . If the integrations implied by (15) and (16) are not tractable analytically, one may employ computational techniques to obtain the dependence of  $t$  on  $f$  or vice versa.

One special case of interest is the one in which the signal envelope or modulus  $a(t)$  is given by

$$\begin{aligned} a(t) &= 1, \quad -T/2 \leq t \leq T/2 \\ &= 0, \quad \text{elsewhere} \end{aligned}$$

and the spectral function or modulus  $A(f)$  is such that when processed by the matched filter receiver results in the prescribed response. The choice of  $A(f)$  for a specified sidelobe level and compressed pulse shape can be made from any of several well-known families of spectral functions [3,4].

Appendix-1 provides the computer program used to compute numerically the required phase function  $\phi(t)$  given  $a^2(x) = 1, -T/2 \leq t \leq T/2$ , and arbitrary spectral functions for  $A^2(f)$ . In the program, use of expression (15) is made to compute the value of  $f$  corresponding to a given  $t$ , and this information is subsequently utilized in Eq. (8) to evaluate the derivative  $\phi'(t)$  at that point in time. From the numerically established behavior of the derivative, the program proceeds and evaluates  $\phi(t)$  which is the phase to be assigned to the transmitted signal.

### III. NONLINEAR FM SIGNALS IN THE PRESENCE OF DOPPLER SHIFT

The compressed pulse resulting from nonlinear FM pulse-compression systems is highly sensitive to frequency shifts due to doppler. This is due to a resulting phase mismatch condition which is responsible for causing severe distortions in the receiver response both in terms of degradation in the main lobe and the level of the range sidelobes.

The received spectrum is given by

$$S(\omega) = A(\omega + \omega_d) \exp[j\theta(\omega + \omega_d)] \quad (17)$$

where  $\omega_d = 2\pi f_d$ , the doppler frequency shift in rad/sec. The matched-filter transfer function is given by

$$H(\omega) = A(\omega) \exp[-j\theta(\omega)] \quad (18)$$

and the output of the matched filter by

$$G(\omega) = A(\omega + \omega_d) A(\omega) \exp[j\{\theta(\omega + \omega_d) - \theta(\omega)\}] \quad (19)$$

The function  $\theta(\omega)$  is obtained using procedures developed in the preceding section.

In order to overcome the detrimental effects of phase mismatch due to doppler frequency shifts, it is proposed that a bank of filters be designed which are perfectly matched in phase to signals received at equally spaced frequencies spanning the entire interval of doppler frequencies of interest (i.e.,

$$f_c - f_{d_{\text{Max}}} \text{ to } f_c + f_{d_{\text{Max}}}).$$

The center frequencies of the matched filters used in adjacent doppler channels are so spaced that the resulting deterioration in range sidelobe levels is kept within prespecified bounds.

The design of matched filters for processing nonlinear FM signals is a very difficult technical undertaking, and the task becomes even more complex if

the filters are required to process a wide variety of spectral functions,  $A(f)$ . Such a need may arise either from technical or from tactical considerations, and it is best satisfied by implementing the matched filters and associated signal processing numerically. Such an approach requires that we design the matched-filter transfer function in a way which accounts for phase distortions due to doppler frequency shifts. Knowledge of the dependence of  $A(\omega)$  on  $\omega$  facilitates accounting of the doppler effects on the signal and its subsequent processing. In order to obtain the filter characteristics for perfect match in the presence of a doppler frequency shift equal to  $\omega_d$ , one must obtain  $\theta(\omega)$  from the moduli  $a(t)$  and  $A(\omega+\omega_d)$ . The modulus  $a(t)$  being the envelope of the transmitted signal is assumed to remain fixed, but  $A(\omega)$  is adjusted to reflect the frequency shift due to doppler.

The required spacing between the center frequencies of adjacent doppler filters for a specified sidelobe or signal-to-noise ratio deterioration is determined via numerical techniques utilizing the program given in Appendix-1. In the case of a square-root Taylor frequency spectrum, adjacent doppler channels with their center frequencies placed at points corresponding to a target velocity of 1.76 of MACH-1 result in a signal-to-noise ratio deterioration of less than -0.22 db; the corresponding distortion in the time response yields -18.1 db sidelobe level instead of the design value of -33.2 db.

Figure 1 offers a processing scheme for non-linear FM signal waveforms.

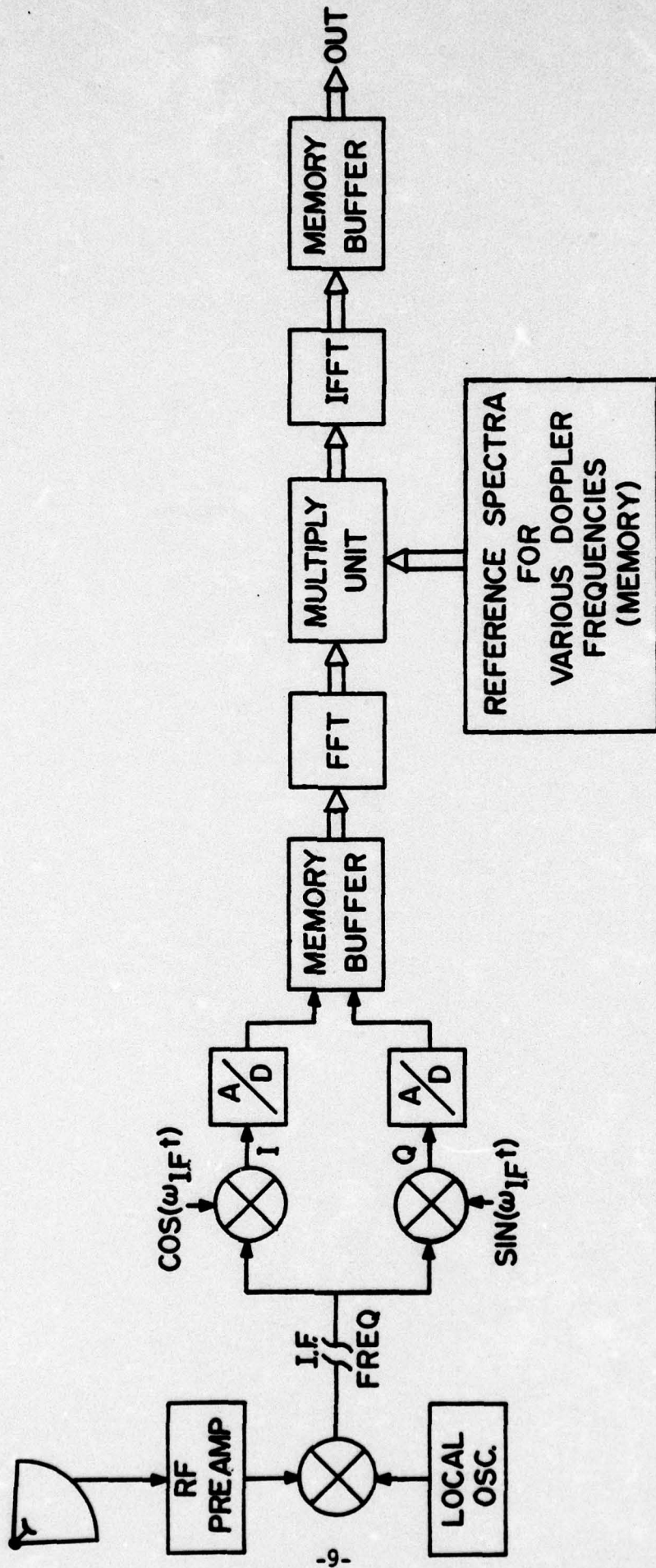


Fig. 1. - Nonlinear FM Pulse Compression System

#### IV. DESIGN CONSIDERATIONS FOR THE NON-LINEAR FM PULSE COMPRESSION SYSTEM

In order to accommodate information rates and signal waveforms compatible with modern radar system requirements, the design of the nonlinear FM pulse compression system must display

- a/ flexibility in the choice of transfer function, and
- b/ bandwidth capability of the order of many megahertz.

The issue of system flexibility is easily settled by assuming digital implementation of the processor. This choice usually puts extra pressure on the system designer to develop FFT (Fast-Fourier Transform) units capable of coping with large bandwidths. In the entire system shown in Fig. 1, the FFT units are the only design challenges, and with present-day technology and existing FFT algorithms [5-12] their implementation has become both economically and technically feasible. The factors which determine the size of the FFT unit, and thus its cost, are the time-bandwidth product, speed and precision requirements of the system.

At this point in time, the design of FFT processors has reached sufficient maturity to satisfy most of the signal processing needs encountered in modern radar systems. The analysis and error sensitivity of various FFT algorithms have been well documented [13-20], and there is no need for their duplication here. It should be pointed out, however, that new advances in the area of parallel computational structures [21-25] point to more cost-effective FFT processor designs in the future, and thus to more economical and flexible signal processing systems.

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## APPENDIX-1

The program given in this Appendix uses square-root Taylor weighting in the frequency domain for -40 db sidelobe level. The input quantities DEL1 and DELTA ( ) specify the doppler frequency shift we wish to center the matched filter under design at, and the sequence of doppler frequencies for use in testing the performance of the matched filter, respectively. Note that DELTA or DEL1 are normalized quantities corresponding to the parameter,

$$\delta = (2 v/c)(f_c/\Delta f)$$

where  $v$  = target radial velocity;  $c$  = velocity of light;  $f_c$  = carrier frequency; and  $\Delta f$  = signal bandwidth. Furthermore, many of the parameters are assigned values within the program and not thru a READ statement.

ND = Number of doppler frequencies to be investigated

N = Number of sample points to be processed ;  $M = N/16$ .

NSTAGE = A positive integer such that  $N = 2^{**} (NSTAGE)$ . It denotes the number of FFT stages.

AF( ) = Coefficients determining the polynomial signal taper employed. The illustrative values used in the program represent a -40 db sidelobe square-root Taylor weighting. AF( ) is DIMENSIONED to accommodate up to ten polynomial coefficients (see p. 20-18 of RADAR HANDBOOK by Skolnik.)

D = Compression ratio. The compression ratio is set equal to  $(N/4 - 0.25)$  which indicates four samples per cycle with the first sample at time  $t = 0$ , and the last sample omitted. (Also,  $D = \text{COMPR}$ )

MF =  $N/4$

MPSI = Number of Fourier coefficients wanted for approximating the signal phase function  $\text{PSI}(\ )$  in the time domain.

DEL1 = Center-frequency at which the nonlinear filter is tuned. It is given in normalized form and has been defined earlier in the Appendix.

DELTA( ) = Doppler frequencies expressed in normalized form and in a manner similar to DEL1.

#### Discussion of subroutines and functions

1. SUBROUTINE EVLPSI (NSTAGE, N, NN, D). It evaluates the Fourier coefficients  $B(\ )$  for the approximation of the real signal phase function  $\text{PSI}(t)$ .
2. SUBROUTINE EVFLTR (NSTAGE, N, NN, D, FACTOR, DEL1). It evaluates the transfer function of the nonlinear filter tuned at DEL1. The parameter FACTOR is only defined and computed within the subroutine for latter use by other subroutines.
3. SUBROUTINE RESPNS (NSTAGE, N, NN, D, DEL, FACTOR, XMAX). It evaluates the response of the nonlinear filter due to a unit-amplitude rectangular pulse of real phase  $\text{PSI}(t)$ .
4. SUBROUTINE TSOPLT (MM). It provides a quick "line-printer" plot of the filter response (or of any function computed within the program).
5. SUBROUTINE SAMPLE (NF, SIGN, D, DEL, NSTAGE). It samples the signal  $\exp(j\text{PSI}(t))$

6. SUBROUTINE FFT(NSTAGE, SIGN). It evaluates the discrete Fourier transform of a set of sample points.
7. SUBROUTINE TMPROC (NF, D, DEL, NSTAGE). It is used in conjunction with SUBROUTINE RESPNS( )
8. SUBROUTINE SELECT (NF, VARBLE, OUTPUT). It selects between the functions SGNL( ) and TAPER( ) depending on the value NF.
9. SUBROUTINE INVERT (NF, XTOL, FTOL, NTOL, Y). It determines the root(s) of a function using Newton's method.
10. SUBROUTINE AMPLE (THIGH, N, NN). It samples the derivative function of PSI(t).
11. SUBROUTINE INTGRN(NF, C, E, M, NMAX, STOL, S). It integrates either function SGNL or TAPER depending on the value of NF.
12. FUNCTION PSI (D, DEL, SIGN, T). It provides the real phase of the rectangularly shaped signal assuming square-root Taylor weighting.
13. FUNCTION TAPER (PHI). It provides the spectral function of the signal which will result in the desired compressed pulse. In this case, use is made of square-root Taylor weighting.
14. FUNCTION SGNL(T). It provides a constant, unit-amplitude taper.

```

DOPLNEW JOB DU.D07.AB1203,MARINOS,T=(2,00),P=50
DOPLNEW EXEC WATFIV,REGION=220K
SYSIN DD *
DB DU.D07.AN3066/MARINOS,TIME=120,PAGES=50
COMPLEX X,DATANL
DOUBLE PRECISION D,DT,DFLOAT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
DIMENSION DELTA(8)
READ (1,2102) ND, NSTAGE, M, MPST
102 FORMAT (4I10)
RFAD (1,2103) DEL1, COMPR
103 FORMAT (2F10.0)
AF(1)=0.3891154
AF(2)=-0.0094523
AF(3)=0.0048819
AF(4)=-0.0016105
AF(5)=0.0003474
N=2*NSTAGE
D=COMPR
MF=N/4
READ (1,102) (DELTA(K),K=1,ND)
102 FORMAT (8F10.0)
NN=2*N
MM=2*M-1
MFOUR=MPST
D1=0
WRITE (3,1)
1 FORMAT (1H1)
CALL EVLPSI (NSTAGE,N,NN,D)
IF (IFLAG.GT.0.OR.NFLAG.GT.0) GO TO 999
WRITE (3,9) D1
9 FORMAT (3X,31H THE COMPR. RATIO IS EQUAL TO ,F10.3,/)
WRITE (3,302)
302 FORMAT (3X,'THE FOURIER COEFFICIENTS OF PSI(T) ARE',/)
WRITE (3,303)
303 FORMAT (6X,5H N ,12X,8H B(N) ,/)
DO 304 I=1,MFOUR
WRITE (3,305) I,B(I)
305 FORMAT (3X,I6,E20.6,/)
304 CONTINUE
WRITE (3,1)
CALL EVFLTR (NSTAGE,N,NN,D,FACTOR,DEL1)
INITIALIZE THE LOOP FOR THE DELTA VALUES.
J=0
15 J=J+1
IF (J.GT.ND) GO TO 999
DFL=DELTA(J)
CALL RESPNS (NSTAGE,N,NN,D,DEL,FACTOR,XMAX)
WRITE (3,9) D1
WRITE (3,21) DEL
21 FORMAT (3X,20H DELTA IS EQUAL TO ,E15.4,/)
WRITE (3,298) DEL1
898 FORMAT (13X,26H FILTER CENTERED AT DELTA= ,E15.4,/)
WRITE (3,98) XMAX
98 FORMAT (3X,'THE SIGNAL TO NOISE RATIO IS EQUAL TO',E20.6,' DBS',
1 //)
WRITE (3,99)
99 FORMAT (3X,'THE AMPLITUDE OF THE RESPONSE IS EXPRESSED IN DBS',/)
CALL TSOPLT (MM)
WRITE (3,1)
GO TO 15
990 CONTINUE
STOP
END

```

```

SUBROUTINE EVFLTR (NSTAGE,N,NN,D,FACTOR,DEL1)
COMPLEX X,DATANL
DOUBLE PRECISION D,DT,DFLOAT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
COMPUTE REQUIRED FILTER COMPENSATION, NEXT!
DT=D/DFLOAT(N-1)
LSTAGE=NSTAGE+1
MF1=MF+1
CALL SAMPLE (2,+1.,D,0.,NSTAGE)
CALL FFT (LSTAGE,-1.)
DATANL(1)=X(2,1)
DO 250 K=2,MF
DATANL(K)=X(2,K)
DATANL(NN+2-K)=X(2,NN+2-K)
250 CONTINUE
DO 251 K=MF1,N
DATANL(K)=CMPLX(0.0,0.0)
DATANL(NN+2-K)=CMPLX(0.0,0.0)
251 CONTINUE
DATANL(N+1)=CMPLX(0.0,0.0)
CALL SAMPLE (2,+1.,D,DEL1,NSTAGE)
CALL FFT (LSTAGE,-1.)
DATANL(1)=DATANL(1)/X(2,1)
DO 8001 K=2,MF
DATANL(K)=DATANL(K)/X(2,K)
DATANL(NN+2-K)=DATANL(NN+2-K)/X(2,NN+2-K)
8001 CONTINUE
WRITE (3,1)
1 FORMAT (1H1)
CALL SAMPLE (2,-1.,D,0.,NSTAGE)
CALL FFT (LSTAGE,-1.)
DATANL(1)=DATANL(1)*X(2,1)
DO 8000 K=2,MF
DATANL(K)=DATANL(K)*X(2,K)
DATANL(NN+2-K)=DATANL(NN+2-K)*X(2,NN+2-K)
8000 CONTINUE
THE COMPLEX ARRAY DATANL CONTAINS THE FILTER
DAT= REAL(DATANL(1))**2+AIMAG(DATANL(1))**2
DO 8002 K=2,MF
GAT= REAL(DATANL(K))**2+AIMAG(DATANL(K))**2
DAT=DAT+2.*GAT
8002 CONTINUE
THE CONSTANT 'FACTOR' IS EQUAL TO THE DENOMINATOR IN THE DEFINI-
TION OF THE SIGNAL-TO-NOISE RATIO.
FACTOR=DAT/2.
RETURN
END

```

```

SUBROUTINE RESPNS (NSTAGE,N,NN,D,DEL,FACTOR,XMAX)
COMPLEX X,DATANL
DOUBLE PRECISION D,DT,DFLOAT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
PREPARATION OF THE OUTPUT OF THE RESPONSE.
CALL TMPROC(2,D,DEL,NSTAGE)
DO 11 K=1,NN
DATA(K)=REAL(X(1,K))*2+AIMAG(X(1,K))*2
DATA(K)=DATA(K)/FACTOR
TA(K)=-D+DFLOAT(K-1)*DT
11 CONTINUE
XMAX=DATA(1)
DO 97 I=1,NN
IF (DATA(I).GT.XMAX) XMAX=DATA(I)
97 CONTINUE
DO 12 K=1,MM
DATA(K)=DATA(N-M+K)
DATA(K)=10.*ALOG10(DATA(K))
TA(K)=TA(N-M+K)
12 CONTINUE
XMAX=10.*ALOG10(XMAX)
RETURN
END

```

```

SUBROUTINE SAMPLE (NF,SIGN,D,DEL,NSTAGE)
COMPLEX X,DATANL
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
N=2*NSTAGE
N1=N+1
NN=2*N
IF (NF.EQ.1) GO TO 100
T=-D/2.-DT
DO 2 I=1,N
T=T+DT
Y=PSI(D,DEL,SIGN,T)
YC=COS(Y)
YS=SIN(Y)
X(1,I)=CMPLX(YC,YS)
2 CONTINUE
DO 3 I=N1,NN
X(1,I)=CMPLX(0.0,0.0)
3 CONTINUE
RETURN
00 Y=DATA(1)
YC=COS(Y)
YS=SIN(Y)
X(1,1)=CMPLX(YC,YS)
DO 4 I=2,MF
Y=DATA(I)
YC=COS(Y)
YS=SIN(Y)
X(1,I)=CMPLX(YC,YS)
X(1,NN+2-I)=X(1,I)
4 CONTINUE
RETURN
END

```

```

SUBROUTINE FFT ( NSTAGE, SIGN)
COMPLEX X, DATANL, W
DOUBLE PRECISION D, DT
DOUBLE PRECISION AF, X0, GX0, GCONST, WCONST
COMMON DT, X(2,2048), DATA(2048), TA(2048), A(101), B(101), MFOUR, MF
COMMON DATANL(2048), DATA(2048)
COMMON AF(10), X0, GX0, GCONST, WCONST, NFLAG, IFLAG, M, MM
INTEGER R
N=2*NSTAGE
N2=N/2
FLTN=N
PHI2N=6.2831853/FLTN
DO 3 J=1, NSTAGE
N2J=N/(2**J)
NP=N2J
NI=(2**J)/2
DO 2 I=1, NI
IN2J=(I-1)*N2J
FLIN2J=IN2J
TEMP=FLIN2J*PHI2N*SIGN
W=CMPLX(COS(TEMP), SIN(TEMP))
DO 5 R=1, NR
ISUR=R+IN2J
ISUR1=R+IN2J*2
ISUR2=ISUR1+N2J
ISUR3=ISUR+N2
X(2, ISUR)=X(1, ISUR1)+W*X(1, ISUR2)
X(2, ISUR3)=X(1, ISUR1)-W*X(1, ISUR2)
5 CONTINUE
2 CONTINUE
DO 6 P=1, N
X(1, P)=X(2, P)
6 CONTINUE
3 CONTINUE
IF (SIGN.GT.0.) RETURN
DO 4 R=1, N
X(2, R)=X(1, R)/FLTN
4 CONTINUE
RETURN
END

```

```

SUBROUTINE TMPROC (NF, D, DEL, NSTAGE)
COMPLEX X, DATANL
DOUBLE PRECISION D, DT
DOUBLE PRECISION AF, X0, GX0, GCONST, WCONST
COMMON DT, X(2,2048), DATA(2048), TA(2048), A(101), B(101), MFOUR, MF
COMMON DATANL(2048), DATA(2048)
COMMON AF(10), X0, GX0, GCONST, WCONST, NFLAG, IFLAG, M, MM
LSTAGE=NSTAGE+1
NN=2*LSTAGE
CALL SAMPLE (NF, +1., D, DEL, NSTAGE)
CALL FFT ( LSTAGE, -1.)
DO 2 K=1, NN
X(1, K)=X(2, K)*DATANL(K)
2 CONTINUE
CALL FFT ( LSTAGE, +1.)
RETURN
END

```

FUNCTION PSI (D,DEL,SIGN,T)

```

COMPLEX X,DATANL
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
DIMENSION CX(101)
PI=3.1415927
Y=2.*PI*T/D
CX1=COS(Y)
CX(1)=CX1
CX(2)=2.*CX1*CX1-1.
DO 2 L=3,MFOUR
CX(L)=2.*CX(L-1)*CX1-CX(L-2)
2 CONTINUE
Z=0.
DO 3 L=1,MFOUR
Z=Z-B(L)*CX(L)/FLOAT(L)
3 CONTINUE
Z=D*Z
E=PI*T*T/D
Z=Z+E
E=2.*PI*DEL*T
G=SIGN*Z
PSI=G+E
RETURN
END

```

SUBROUTINE TSOPLT (N)

```

COMPLEX X,DATANL
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
DATA ISTAR/IH*/
XMAX=DATA(1)
XMIN=DATA(1)
DO 1 J=1,N
IF (DATA(I).GT.XMAX) XMAX=DATA(I)
IF (DATA(I).LT.XMIN) XMIN=DATA(I)
1 CONTINUE
DIFF=XMAX-XMIN
SC=100./DIFF
300 FORMAT (3,300) SC,XMIN,XMAX
WRITE (4X,'SC=',E12.5,' MIN=',E12.5,' MAX=',E12.5,////)
117 FORMAT (1H,9H TIME ,5X,18H AMPLITUDE ,//)
DO 10 I=1,N
XA=ABS(SC*(DATA(I)-XMIN))
IDA=IFIX(XA+1.)
200 FORMAT (1H,F9.3,5X,E12.5,5X,102A1)
10 CONTINUE
RETURN
END

```

```

SUBROUTINE EVLPSI (NSTAGE,N,NN,D)
COMPLEX X,DATA1L
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
DOUBLE PRECISION THIGH,OUT
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
THIGH=D/2.
LSTAGE=NSTAGE+1
C EVALUATION OF THE CONSTANTS.
WRITE (3,1)
1 FORMAT (1H1)
CALL INTGRN (1.0,D 00,THIGH,1.512,0.1D-05,OUT)
IF (NFLAG.GT.0) GO TO 990
GCONST=OUT
CALL INTGRN (2.0,D 00,0.5D 00,1,256,0.1D-05,OUT)
IF (NFLAG.GT.0) GO TO 990
WCONST=OUT
CALL AMPL (THIGH,N,NN)
IF (IFLAG.GT.0.OR.NFLAG.GT.0) RETURN
CALL FFT (LSTAGE,-1.)
DO 2 I=1,MFOUR
B(I)=-2.*AIMAG(X(2,I+1))
2 CONTINUE
DO 10 I=1,MFOUR.2
R(I)=-B(I)
10 CONTINUE
RETURN
990 WRITE (3,8)
8 FORMAT (3X,'AN INTEGRATION IN SUBROUTINE EVLPSI HAS FAILED',//)
RETURN
END

```

```

SUBROUTINE SELECT (NF,VARIABLE,OUTPUT)
COMPLEX X,DATA1L
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
DOUBLE PRECISION VARIABLE,OUTPUT,SGNL,TAPER
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
GO TO (10,20),NF
10 OUTPUT=SGNL(VARIABLE)
RETURN
20 OUTPUT=TAPER(VARIABLE)
RETURN
END

```

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```

SUBROUTINE INTGRN (NF,C,E,N1,NMAX,STOL,S)
COMPLEX X,DATANL
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
DOUBLE PRECISION C,E,STOL,S,DFLOAT,S1,H,HALF,XY,DS,DABS,S2,FC,FE,
1 HALF),C1
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
NFLAG=0
M)=N)
S1=0.D 00
CALL SELECT (NF,C,FC)
CALL SELECT (NF,E,FE)
1 M)=2*M)
IF (M1.GT.NMAX) GO TO 999
H=(F-C)/DFLOAT(M1)
HOV2=H/2.
S=0.D 00
C1=C+HOV2
CALL SELECT (NF,C1,HALF)
NM1=M1-1
DO 2 I=1,NM1
XY=C+DFLOAT(I)*H
CALL SELECT (NF,XY,S2)
S=S+S2
C1=XY+HOV2
CALL SELECT (NF,C1,HALF1)
HALF=HALF+HALF1
2 CONTINUE
S=(H/6.)*(FC+4.*HALF+2.*S+FE)
DS=S-S1
IF (DABS(DS).GT.STOL) GO TO 3
RETURN
3 S)=S
GO TO 1
999 NFLAG=1
RETURN
END

```

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```

SUBROUTINE AMPLE (THIGH,N,NN)
COMPLEX X,DATANL
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
DOUBLE PRECISION DFLOAT, T,T1,GTINT,GTINT1,Y,RL,THIGH
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATANL(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
X0=0.D 00
GX0=0.D 00
GTINT=0.D 00
DT=2.*THIGH/DFLOAT(NN-1)
T=0.D 00
T1=DT/2.
C LOOP FOR THE EVALUATION OF THE DERIV. OF THE PHASE PSI (T,ALPHA).
DO 1 I=1,N
CALL INTGRN (1,T,T1,1.512,0.1D-05,GTINT1)
IF (NFLAG.GT.0) GO TO 990
GTINT=GTINT+GTINT1
Y=WCONST*GTINT/GCONST
CALL INVERT (2,0.1D-05,0.1D-07,500,Y)
IF (NFLAG.GT.0) GO TO 990
IF (IFLAG.GT.0) RETURN
DATA(N+1)=X0
T=T1
T1=T+DT
1 CONTINUE
DO 2 I=1,N
RL=(DFLOAT(I)-0.5)*DT/(2.*THIGH)
R=RL
R=DATA(N+I)-R
X(1,N+I)=CMPLX(R,0.0)
X(1,N-I+1)=CMPLX(-R,0.0)
2 CONTINUE
RETURN
990 WRITE (3,3)
3 FORMAT (3X,51H AN INTEGRATION IN THE SUBROUTINE SAMPLE HAS FAILED)
1 RETURN
END

```

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```

SUBROUTINE INVERT (NF,XTOL,FTOL,NTOL,Y)
COMPLEX X,DATA1L
DOUBLE PRECISION D,DT
DOUBLE PRECISION AF,X0,GX0,GCONST,WCONST
DOUBLE PRECISION FX0,X1,GX1,XTOL,FTOL,Y,DABS,DERIV,DELTA
COMMON DT,X(2,2048),DATA(2048),TA(2048),A(101),B(101),MFOUR,MF
COMMON DATA1L(2048),DATA(2048)
COMMON AF(10),X0,GX0,GCONST,WCONST,NFLAG,IFLAG,M,MM
IFLAG=0
DO 1 I=1,NTOL
FX0=GX0-Y
IF (DABS(FX0).LT.FTOL) RETURN
CALL SELECT (NF,X0,DERIV)
IF (DERIV.EQ.0.D 00) GO TO 999
DELTA=FX0/DERIV
X1=X0
X0=X0-DELTA
CALL INTGRN (NF,X1,X0,1.256,0.1D-05,GX1)
IF (NFLAG.GT.0) GO TO 989
GX0=GX0+GX1
IF (DABS(DELTA).LT.XTOL) RETURN
1 CONTINUE
IF (DABS(DELTA).GT.XTOL) GO TO 990
RETURN
989 WRITE (3,2)
2 FORMAT (3X,49H THE INTEGRATION IN SUBROUTINE INVERT HAS FAILED ,
1 //)
RETURN
999 WRITE (3,3)
3 FORMAT (3X,45H THE DERIVATIVE IN SUBROUTINE INVERT IS ZERO ,//)
IFLAG=2
990 WRITE (3,4)
4 FORMAT (3X,50H DELTA IN SUBROUTINE INVERT IS GREATER THAN XTOL
1 //)
IFLAG=1
RETURN
END

```

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FUNCTION TAPER (PHI)

```
COMPLEX X, DATANL
DOUBLE PRECISION D, DT
DOUBLE PRECISION PI2, CX, Y, PHI, CX1, Z, DLOG, DEXP, TAPER, DCOS
DOUBLE PRECISION AF, X0, GX0, GCONST, WCONST
COMMON DT, X(2, 2048), DATA(2048), TA(2048), A(101), B(101), MFOUR, MF
COMMON DATANL(2048), DATA(2048)
COMMON AF(10), X0, GX0, GCONST, WCONST, NFLAG, IFLAG, M, MM
DIMENSION CX(10)
PI2=6.283185307179
Y=PI2*PHI
CX1=DCOS(Y)
CX(1)=CX1
CX(2)=2.*CX1*CX1-1.
DO 2 L=3,5
CX(L)=2.*CX(L-1)*CX1-CX(L-2)
2 CONTINUE
Z=1.D 00
DO 3 L=1,5
Z=Z+2.*AF(L)*CX(L)
3 CONTINUE
TAPER=Z
RETURN
END
```

FUNCTION SGNL (T)

```
COMPLEX X, DATANL
DOUBLE PRECISION D, DT
DOUBLE PRECISION AF, X0, GX0, GCONST, WCONST
DOUBLE PRECISION T, SGNL
COMMON DT, X(2, 2048), DATA(2048), TA(2048), A(101), B(101), MFOUR, MF
COMMON DATANL(2048), DATA(2048)
COMMON AF(10), X0, GX0, GCONST, WCONST, NFLAG, IFLAG, M, MM
SGNL=1.D 00
RETURN
END
```

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14.

KEY WORDS

Pulse Compression  
Nonlinear FM  
Fast Fourier Transform  
Digital Signal Processor  
Doppler Sensitivity

LINK A		LINK B		LINK C	
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