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Inventor William M. Waters

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Inventor: Waters
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COMMUNICATIONS SYSTEM USING A SHARPLY BANDLIMITED KEYING WAVEFORM

Background of the Invention

Field of the Invention

This invention in general describes a communications system, specifically a communications system having sharply bandlimited waveforms for sampled data communications.

Description of the Related Art

Proliferation of communications, radar, and other systems making use of electromagnetic wave propagation has resulted in overcrowding in the useful portions of the electromagnetic spectrum. Even line-of-sight communications links seldom utilize frequencies above the X-band because of the absorption and scattering by rain and other constituents of the troposphere. Consequently, it becomes more important for transmitters to radiate signals with well confined spectra so limited spectral space is used more efficiently.

In order to communicate a sequence of sampled digital data, r(tn), over a band limited communications channel, the sequence of values must be transformed into an analog signal which is spectrally confined to the band available. Traditionally, confinement of radiated signals to assigned bands has been accomplished by frequency or phase modulation (FSK or PSK) followed by analog filtering operations at intermediate frequency and final amplifier stages of the transmitter. At the receiver, this signal is filtered from adjacent channels after appropriate down-conversion by utilizing analog filters to select the desired signal for subsequent conversion to digital form.

It is well known to those practicing the art that the spectrum of a signal formed from appropriate combinations of cardinal functions will have zero energy outside the designated frequency band width. For this to be exact, it is required that the cardinal functions be of infinite length. Theoretically, this causes no intersymbol interference since cardinal functions are zero at all sample points except one, t=0. Thus the analog/digital converter, appropriately synchronized and in the absence of noise, reads values of the transmitted sequence.

Functions of infinite length are not physically realizable. However, studies have shown that, by appropriately modifying the cardinal function, one can achieve excellent interpolation accuracies and resultant phase accuracies in direct sampling applications even if the function is truncated to just a few sample intervals.

In the field of electromagnetic transmission of digital data there is a need for an alternative to FSK and various forms of PSK signals which would improve the confinement of the signal spectrum to an assigned frequency band in transmitters and receivers. It is well known that bi-phase code modulation consists of a sequence of equal-length intervals, often called "chips" or samples. Within each chip a sinusoid at the carrier frequency is bi-phase modulated; more specifically, the carrier phase within a chip is constant but may jump from chip-to-chip by 180 degrees in accordance with modulating binary data. A drawback to the application of this waveform is the 180° phase "jump" which causes spreading of its spectrum. However, after a bi-phase signal is modified by the process which is the subject of this invention, its spectrum is well-defined to the desired band.

Further, it is important that adjacent communication channel interference and mutual interference among radar, navigation, communications and other systems which depend upon electromagnetic radiation be prevented.

SUMMARY OF THE INVENTION

It is the object of this invention to transform a communications sequence in a communications system into an analog signal which is spectrally confined to the band available for the transmission within a band limited communications channel without compromising communication accuracy.

Another object of this invention is to improve spectral confinement thereby allowing closer channel spacing and better electromagnetic compatibility among radar, navigation, and communications systems.

These and other objects are achieved by a communication system using sharply bandlimited signals for sampled data communications wherein a sharply bandlimited waveform is computed for each sample value in a sequence of data samples. Each waveform is centered at a sample point, is weighted by the corresponding data value, and is truncated outside an appropriate time interval. Resulting data-weighted waveforms are summed in the computer for all data samples of the sequence to be communicated, and the result is converted to analog form, up-converted to the desired carrier frequency, and transmitted. Because the signal spectrum is sharply bandlimited, communication channels may be spaced more closely, permitting more channels in the same frequency band.

At the receiver, signal-plus-noise is down-converted, data is sampled and converted to digital form, de-multiplexed (if necessary), and delivered to corresponding users.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 shows the error for low-pass interpolation using cardinal functions and self-truncating cardinal functions for B/W = 3/4 and 1/2 (33% and 100% over-sampling, respectively) as a function of the truncation length as measured by N, the number of samples within the truncated interval.

Figure 2 is a schematic of a transmission system utilizing a sharply bandlimited communications filter.

Figure 3 depicts a test configuration for testing a sharply bandlimited communications filter.

Figure 4 shows (a) the measured spectrum of the output waveform of the waveform generator with interpolation and (b) the measured spectrum of the output waveform of the waveform generator without interpolation.

Figure 5a shows the computed interpolation waveform.

Figure 5b shows the interpolation waveform spectrum.

Figure 5c shows the I component after processing.

Figure 5d shows the Q component after processing.

Figure 5e shows the correlated magnitude $(I^2 + Q^2)^{1/2}$ of the I and Q components.

DESCRIPTION OF THE PREFERRED EMBODIMENT

In order to communicate a sequence of sampled digital data over a bandlimited communications channel in a format other than the conventional analog format, a bi-phase code modulation utilizing a sharply bandlimited keying waveform may used.

A sharply bandlimited keying waveform is computed for each sample value in a sequence of data samples and centered at a sample point, weighted by a corresponding data value, and truncated outside an appropriate time interval. For a predetermined carrier frequency, f_o, it is well known in the art that sampling theory requires that the signal must be sampled at a rate $f_s = 4f_o/(2M-1)$ which is greater than or equal to twice the signal bandwidth. Where M is an integer. Defining the bandwidth to be $1/t_c$, where t_c is the spacing between "chips" (elements of a phase code) or chip width, then the minimum sampling rate is $f_s = 2/t_c$.

A signal with no spectral energy outside the band from -W to +W is completely specified by samples taken at uniform intervals less than or equal to 1/2W. The original signal may be exactly reconstructed from these sampled values, $r(t_n)$, by the interpolation

$$r(t) = \sum_{-\infty}^{+\infty} r(t_n)\, s(t-t_n) \tag{1}$$

where $t_n = n/2W$, n being the number of the samples, and the interpolation function, or sampling function, is

$$s(t) = \frac{(\sin 2\pi Wt)}{(2\pi Wt)} \tag{2}$$

A bandpass signal with no spectral energy outside the bands -MW to -(M-1)W and (M-1)W to MW may also be completely specified by samples taken at 1/2W (or shorter) intervals. The original function may be reconstructed exactly from these sampled values by the interpolation defined by Eq.(1).

Eq.(1) holds exactly only if s(t) as defined by Eq. (2) for the bandpass case is defined for all values of t from $-\infty$ to $+\infty$. The interpolation error (magnitude of the difference between right and left sides of Eq. (1)) grows as s(t) is truncated to shorter lengths, and as the width of the spectrum of the sampled signal, B, approaches 1/2 the sampling rate f_s + 2W.

Figure 1 shows the error for low-pass interpolation (Eqs. (1) and (2)) for B/W = 3/4 and 1/2 (33% and 100% over-sampling, respectively) as a function of the truncation length as measured by N, the number of samples within the truncated interval. **Figure 1** also shows how the interpolation error shrinks dramatically (for large N) as a self-truncating "taper" is applied to the interpolation function. This tapered function is given for the low-pass case where q=1-B/W and m depends on both q and N.

$$s(t)=[\sin(2\pi qW/m)\,t/(2\pi qW/m)\,t]^{m}\sin 2\pi Wt/2\pi Wt \quad (3)$$

where q=1-B/W.

Since interpolation error is dramatically reduced using the "self-truncating" interpolation function, it is logically concluded that the spectrum of s(t) from Eq. (3) is better confined to the band -W to +W than of s(t) from Eq. (2). The same logic applies to the bandpass case since the taper of Eq. (3) (the bracketed factor raised to the power, m) may be used in the bandpass case with similar results.

The interpolation waveforms have the property that the zeros are at the right points so as not to cause intersymbol interference. Normally the keying waveform is truncated to a shorter time interval resulting in poor spectral confinement.

In the preferred embodiment, referring to **Figure 2**, sampled digital data in a sequence of data samples **12** are scaled according to the value $r(t_n)$ and input to the system **10** where a waveform generator **16** generates an interpolation waveform for each data sample **12**. The sampled input data **12** can be voice, television signals, pagers, or any other signal sampled as a function of time.

The sequence of data-weighted waveforms are summed in a computer **14** for all data samples of the sequence to be communicated, and the result is converted to analog form by a digital-to-analog (D/A) converter (not shown), filtered in a low-pass filter **22**, up-converted to the desired carrier frequency in a up-converter **18**, and the signal is transmitted by a conventional method for transmission of electromagnetic waves, a transmitter **24**.

The computer **14** receives a data bit stream from the sampled input data **12** and performs three functions. First, the computer **14** controls a waveform generator **16**, secondly it weighs each waveform with the data and, thirdly, it forms the sum of weighted waveforms to form the output signal in digital form. The waveform generator 16 takes the digital data bits containing the output signal and converts it to analog form by the use of an internal digital-to-analog (D/A) converter (not shown). The mathematical mechanism by which a waveform generator 16 synthesizes the bandlimited signal is as described above.

The output of the waveform generator 16 is passed through an integral or external low-pass filter 22 and applied to the up-converter 18 where the signal is up-converted to the desired carrier frequency which can then be multiplexed in a multiplexer 21, if required. A number of data channels may be multiplexed consistent with the group bandwidth. Final separation of data channels may then be accomplished according to the multiplexing method used. Analog low-pass filters 22 are well known to those practicing the art and will not be further discussed. The analog low-pass filter 22 filters out any harmonics generated and applies the filtered analog signal to the intermediate frequency (IF) stage of a transmitter 24 for generation of sampled data signal output as radio frequency (RF) energy.

A number of data channels may be grouped as sub-bands and a digital-to-analog (D/A) converter (not shown) would sample at a rate consistent with the group bandwidth. Final separation of data channels may then be accomplished by digital filtering.

At a receiving location, the transmitted signal, plus any induced noise, is demultiplexed by a demultiplexer 28, down-converted by a low-noise amplifier filter down-converter 32, data is sampled and converted to digital form by an analog-to-digital (A/D) converter 34 , separated into separate channels by a digital channel separation filter 36, and applied to an audio or digital display device 38.

Any standard receiver 28 is capable of receiving the transmitted signal. The analog signal-plus-noise is down-converted on a down-converter 32 and converted from an analog to a digital format in an analog-to-digital (A/D) converter 34 centered around the 5 MHz bandwidth. The digital signal can then be de-multiplexed by a de-multiplexer 28, if required, filtered by a digital channel separation filter 36, and delivered to the using equipment 38.

In the formation and transmittal of the sharply bandlimited signals for sampled data communications, the waveform generator 16 receiving the sample data delivers accurate data to the receiver 28 unless the remainder of the transmitter 24 distorts the radiated waveform. The up-converter 18 and transmitter 24 power amplifier (not shown) must exhibit a flat amplitude and linear phase characteristic across the signal band. Since the signal is bandlimited, this is more easily accomplished by analog low-pass filters **22**. The better spectral confinement achieved through the use of the device set forth above leads to closer channel spacing and better electromagnetic compatibility among radar, navigation, and communications channels.

Operational data link interpolation may be performed by a finite impulse response (FIR) filter (not shown) implemented by a digital signal processor (DSP) microchip, such as a Part no. DSP96002, manufactured by Motorola Corp. of Phoenix, AZ. The preferred DSP should have a speed and digital precision to provide the same processing as a computer **14**, but requiring only a single microchip plus requisite drivers and power supplies. D/A conversion may also be performed by a much smaller unit than a waveform generator **16**, such as a Model 1010-60-860, manufactured by KOR Industries of Garden Grove, CA. digital-to-analog converter.

The better spectral confinement provided by this invention leads to closer channel spacing and better electromagnetic compatibility among radar, navigation, and communications systems.

Figure 3 depicts a sharply bandlimited keying waveform test configuration **40** of the preferred embodiment under laboratory conditions to verify that the output data is identical to the input data prior to operational data link interpolation. See, Sharply Bandlimited Signals for Sampled Data Communications, NRL/MR/5330--94--7481, Jun 94, which is hereby incorporated by reference.

Data simulating direct samples, f_s, **42** from a pseudo-random bi-phase coded sinusoid (not shown) are selected by a computer **43** and entered into a waveform generator **52** synthesizer memory module **44**. These data are used to weight the interpolation function previously described incorporating the computer program set forth in **Table 1**. **Table 1** specifies mathematically the mechanism by which the waveform generator synthesizes the bandlimited signal from a sequence of data signals. Samples **42** are taken from the sinusoid at intervals, t_s, of 0.144 μs and have a width of biphase modulation chips, t_c, equal to 0.288 μs. Weighted and summed interpolation functions are computed at 0.008 μs intervals corresponding to a 125 MHz rate of the first memory **44** of the waveform generator **52** and read into a fast digital-to-analog (D/A) converter **46**.

Within the D/A converter **46**, the digital values from first memory **44** are converted to analog form and low-pass filtered by a low-pass filter **48** (0 to 50 MHz) to form the bandlimited analog signal centered at f_0 output of the waveform generator **52**, **Figure 3**. The waveform generator **52** in the test installation functioned not only as a waveform generator but also as a D/A converter **46** and a low-pass filter **48**. The spectrum, as shown in **Figure 4a**, of the bandlimited analog signal of the waveform generator **52** is measured by a spectrum analyzer **54**. **Figure 4b** shows the spectrum of the same phase coded signal but without the weighting of Eq. (3).

To analyze the sharply bandlimited keying waveform signal, to determine fidelity with the digital sample **42** from the bi-phase modulated sinusoid, the I and Q components **66** and **64**, respectively, of the output from the waveform generator **52** with the measured spectrum from the spectrum analyzer **54** are split out in an I/Q detector **56** when a reference signal **58** is heterodyned against the signal having the sharply bandlimited keying waveform in phase quadrature in order to obtain phase and amplitude information. Phase accuracy, where the phase, ϕ, is defined as $\text{Tan}^{-1}Q/I$, is dependent upon careful matching of these separate quadrature channels. Accordingly, phase errors limit the performance achievable from subsequent digital processing.

The method of forming two quadrature channels requires two separate frequency translations in parallel. The base-band mixers utilized in these frequency translations must track each other in order to obtain the desired frequency accuracy. Conversion of the I and Q components **62** and **64**, respectively, from an analog to a digital format is accomplished in analog-to-digital (A/D) converters **66** and **68** to which f_s, the sampling rate **72**, is applied. The digital outputs of the A/D converters **66** and **68** are applied to a computer **74** where the I and Q components are correlated with the I and Q components corresponding to the original input data samples **42** in the correlator **76**, and an envelop detector **78** computes the square root of the sum of the squares which is applied to a plotter **82** for display of the test data.

In the test configuration **40**, the particular values of t_s and t_c were chosen to be integer multiples of the waveform generators **52** fast memory read interval of 0.008 μs. The cycle period of the output signal carrier frequency was also chosen to be an integer multiple of 0.008 μs (1/15.625 = 0.064 and 1/31.25 = 0.032). The results of this test are shown in **Figure 5a** and **5b**. The curves in **Figure 5a** depicts I and Q components of the received signal; **Figure 5b** depicts the interpolation waveform spectrum; **Figures 5c** and **5d** show the I and Q components after processing and **Figure 5e** depicts the correlated magnitude $(I^2 + Q^2)^{1/2}$ from **Figures 5c** and **5d**.

It will be understood by those skilled in the art that still other variations and modifications are possible and can be affected without detracting from the scope of the invention

TABLE 1

```
10!  RE-STORE "W_256BIPH1:,1400,1"
20  ! PROGRAM TO GENERATE A DIRECT SAMPLED MAXIMAL
LENGTH BIPHASE WAVEFORM WITH WEIGHTED CHIPS
30  ! FOR OUTPUT TO THE HP8770 WAVEFORM SYNTHESIZER
40      PRINTER IS CRT
50      ! GENERATE MAXIMAL LENGTH SHIFT REGISTER
SEQUENCE
60      OPTION BASE 1
70      SEPARATE ALPHA FROM GRAPHICS
80      GCLEAR
90      INTEGER X(8), Xs(255)
100     INTEGER I,J,K,N,Pn,Ix,Ic,Xn,Ns,Nz,Nc,
Chip,Chs,Che
110     DIM F1$|60|
120     DIM Aa(25), La(25), Ra(25)
130     DIM W$|32767|
140     DIM Comm$|96|
141     GOTO 1250
150     Ix=67 !INITIAL CONDITION
160     N=8 !NUMBER OF STAGES
170     Ic=Ix
180     FOR I=N TO 1 STEP -1   ! SET INITIAL CONDITION
190       X(I)=0
200       Pn=2"(I-1)
210       IF Ic>=(Pn) THEN X(I)=1
220       IF Ic>=(Pn) THEN Ic=Ic-Pn
230     NEXT I
240     Pn=2"N=1   !  NUMBER OF ELEMENTS IN SEQUENCE
250     FOR I=1 TO Pn
260       Xn=(X(8)+X(4)+X(3)+X(2)) MOD 2! FEEDBACK
CONNECTION
270       Xs(I)=X(8)
280       S=0
290       FOR K=1 TO N
300         S=S+X(K)*2"(K-1)
310       NEXT K
320       PRINT USING ""(D),3X,DDD";X(8),X(7),X(6),
X(5),X(4),X(3),X(2), X(1),S
330       FOR J=N TO 2 STEP -1
340         X(J)=X(J-1)
350       NEXT J
360       X(1)=Xn
370     NEXT I
380     RAD
390     DISP "ENTER CHIP WIDTH IN MICROSECONDS";
```

```
400     INPUT Tc
410     DISP "ENTER THE CARRIER FREQUENCY";
420     INPUT F
430     W=125.
440   ! DISP "ENTER THE SAMPLING RATE OF THE HP8770";
450   ! INPUT W
400     DISP "ENTER NAME OF FILE WHERE DATA IS TO BE
STORED";
470     INPUT Nam$
480     DISP " ENTER A COMMENT DESCRIBING WAVEFORM";
490     LINPUT Comm$
500     Nc=T*W!NUMBER OF SAMPLES IN CHIP
510     Ns=Nc*Pn
520     PRINT "NUMBER OF SAMPLES IS ";Ns
530     PRINT "LENGTH OF PULSE IS ";Ns/W
540     Nz=(1-FRACT Ns/8))*8
550     PRINT Nz;" ZEROS WILL BE ADDED TO MAKE SAMPLES A
MULTIPLE OF 8"
560     Tp=(Nz+Ns)/W
570     PRINT " TOTAL PULSE WILL BE ";Tp
580     PRINT
590     PRINT  "              +- 1        12-BIT    WEIGHT"
600     VIEWPOINT 10,150,15,90
610     WINDOW 0,(Ns/w),-2048,2048
620     WINDOW 0,(Ns/W)/16,-1200,1200
630     FRAME
640     CLIP OFF
650     L=1
660     ! INTERPOLATION FUNCTION PARAMETERS
670     M1=13 ! FOR Λ .4 CHIP LENGTH       !5 ! FOR Λ
.144 CHIP LENGTH
680     M2=2
690     U=.1
700     Fo=F
710     Ws=2*Fo/(2*M1 -1)
720     Ic=0
721     Tw=Tc/2
722     Tw2=Tw/2
730     FOR T=0 TO (Ns/W) STEP 1/W !    STEP THROUGH THE
ENTIRE PULSE
740        Chip=(T*W)/Nc+.5
750        Chs=Chip-5
760        Che=Chip+5
770     ! TAPER START AND END
780        IF Chip<=5 THEN Chs=5
790        IF Chip>=Pn-4 THEN Che=Pn-4
800        FOR N=Chs TO Che
801          FOR M=1 TO 2
810            Y=2*PI*Ws*(T-Tc*N-M*Tw+Tw2)
```

```
820          IF Y=0 THEN
830            S3=1
840            GOTO 910
850          ELSE
860            GOTO 880
870          END IF
880          S1=(SIN(M1*Y)-SIN((M1-1)*Y))/Y
890          S2=(SIN(U*Y)/(U*Y))"M2
900          S3=S1*S2
910          S1=((-1)"N)*(Xs(N)-.5)*2*S3
920          V=V+S4
921         NEXT M
930        NEXT N
940        ! LIMIT AMPLITUDE TO 1 V PEAK
950        V=V*.4
960        Mag=ABS(V)
970        IF Mag>=1 THEN V=V/Mag*.99
980        Ic=Ic+1
990        Z=INT(V*2047+.5)
1000       PLOT T,Z
1010       W$|L,L|=CHR$(BINAND(255,BINAND(Z,-256)/256))
1020       W$|L+1,L+1|I=CHR$(BINAND(Z,255))
1030       L=L+2
1040!PRINT USING "DDDDD,4X,DD.DDDD,4X,DDDDDD,4X,
D.DDD";I,X,Z,Y
1050       V=0
1060     NEXT T
1070     Nz=(1-FRACT(Ic/8)*8
1080     PRINT Nz;" ZEROS WILL BE ADDED TO MAKE SAMPLES A
MULTIPLE OF 8"
1090     FOR I=1 TO Nz
1100       Ic=Ic+1
1110       W$|L,L|=CHR$(0)
1120       W$|L+1,L+1|=CHR$(0)
1130       L=L+2
1140     NEXT I
1150     PRINT "TOTAL SAMPLES=";Ic
1160     B=0
1170     Cdir=0
1180     CREATE BDAT Nam$,1,L+200
1190     ASSIGN @Wf TO Nam$
1200     OUTPUT @Wf;L-1,W$|I1,L-1|,Cdir,F,B,Tp,W,Comm$
1210     ASSIGN @Wf TO *
1220!    PRINT "=****** DONE *******"
1230!
1240!
1250!
1260!    PROGRAM TO READ WAVEFORMS AND OUTPUT IN AN
ARBITRARY SEQUENCE
```

```
1270!  CONVERTED TO SERIES 300 3/30/88  MOD 4/14/88
1280!  PRINTER IS CRT
1290   GCLEAR
1300   CLEAR SCREEN
1310   PRINT "              PROGRAM TO SEND WAVEFORMS TO HP8770"
1320   PRINT
1330   PRINT "     AS A MINIMUM, FUNCTION KEYS 1 AND 2 NEED TO BE USED
1340   PRINT "     1 TO LOAD WAVEFORMS FROM DISC AND STORE IN THE 8770 MEMORY"
1350   PRINT "     2 TO SET UP THE SEQUENCING OF THE WAVEFORM DATA"
1360   PRINT
1370   PRINT "A MODE EXISTS VIA KEY 5 TO CHANGE TIME DELAY BY ENTRY OF TIME IN MICROSECONDS"
1380   ON KEY 1 LABEL "LOAD WAV" GOSUB 1480
1390   ON KEY 2 LABEL "SEQ SET" GOSUB 1820
1400   ON KEY 3 LABEL "SEQ CHG" GOSUB2180
1410   ON KEY 4 LABEL "STATUS" GOSUB 2030
1420   ON KEY 5 LABEL "DEL TIME" GOSUB 2250
1430   ON KEY 6 LABEL "CLOCK" GOSUB  2570
1440   ON KEY 7 LABEL "CAT" GOSUB 2690
1450   ON KEY 8 LABEL "RESTART" GOTO 1280
1460   KEY LABELS ON
1470   GOTO 1470
1480   F1$=""
1490   OUTPUT 719;"*RST"
1500   OUTPUT 719;"PURGE BOTH"
1510   OUTPUT 719;"FORMAT SIGN"
1520   GOSUB 2570 !SELECT CLOCK SOURCE
1530   A=0
1540   S$=VAL$(A)! START FIRST WAVEFORM SEGMENT AND PACKET AT ZERO
1550   DISP "ENTER THE NUMBER OF WAVEFORM FILES";
1560   INPUT J
1570   FOR I=1 TO J
1580     DISP "ENTER FILE NAME";
1590     INPUT F$
1600     F1$=F1$&F$&"*"
1610   NEXT I
1620   K=1
1630   PRINT "    WAVEFORM MEMORY CONTENTS"
1640   PRINT " NAME       LENGTH IN SAMPLES   ADDRESS PULSE W at CLOCK  BANDWIDTH START FREQ"
1650   FOR I=1 TO J
1660     M=POS(F1$|K|,"*")
1670     F$=F1$|K,K+M-2|
1680     ASSIGN @W1 TO F$
```

```
1690      K=K+M
1700      ENTER @W1,1;L,W$,Cdir,Fs,B,T,W,Comm$! READ
NUMBER OF BYTES IN, AND DATA
1710      Nn(I)=L/2
1720      N$=VAL$(Nn(I))! NUMBER OF WAVE SEGMENTS
1730      PRINT USING "10A,4X,DDDDD,11X,DDDDD,6X,DDD.D,
4A,DDD,5X,DDD.DDD,6X,DD.DDD";F$,Nn(I),Λ,T," at ",W,B,Fs
1740      Λ1(I)=Λ
1750      U$|1,1|=CHR$(BINAND(255,BINAND(L,-256)/256))
1760      U$|2,2|=CHRS$(BINAND(L,255))
1770      OUTPUT 719;"WMEM "&S$&",#Λ"z&U$&W$|1,L|
1780      Λ=Λ+Nn(I)
1790      S$=VAL$(Λ) NEXT START ADDRESS
1800    NEXT I
1810    RETURN
1820    DISP "ENTER NUMBER OF WAVEFORM SEGMENTS";
1830    INPUT J2
1840    Y=1
1850    J1=J2
1860    FOR I=Y TO J1
1870      DISP "ENTER SEGMENT STARTING ADDRESS";
1880      INPUT Aa(I)
1890      DISP "ENTER LENGTH OF SEGMENT";
1900      INPUT La(I)
1910      DISP "ENTER NUMBER OF TIMES TO REPEAT
SEGMENT";
1920      INPUT Ra(I)
1930    NEXT I
1940    OUTPUT 719;"PURGE SEQ"
1950    FOR I=1 TO J2
1960      S$=VAL$(Λa(I))
1970      N$=VAL$(La(I))
1980      R$=VAL$(Ra(I))
1990      OUTPUT 719;"PACLIT "&S$&","&N$&","&R$&", AUTO"
2000    NEXT I
2010    OUTPUT 719;"GO"
2020    RETURN
2030    PRINT "           SEQUENCE STATUS"
2040    PRINT "ADDRESS    LENGTH    REPEATS    TIME"
2050    FOR I=1 TO J2
2060      PRINT USING "DDDDD,3X,DDDDD,5X,DDDDD,4X,
DDDD.DDD";Λa(I),La(I),Ra(I),Ra(I)/F*LA(I)
2070    NEXT I
2080    PRINT "   WAVEFORM MEMORY CONTENTS"
2090    PRINT "   NAME       LENGTH IN SAMPLES    ADDRESS'
2100    K=1
2110    FOR I=1 TO J
2120      M=POS(F1$|K|,"*")
2130      F$=F1$|K,K+M-2|
```

```
2140      K=K+M
2150      PRINT USING
"10A,4X,DDDDD,12X,DDDDD,";F$,Nn(I),
A1(I)
2160    NEXT I
2170    RETURN
2180    CLEAR SCREEN
2190    GOSUB 2080
2200    DISP "TO CHANGE ONE SEGMENT ENTER SEGMENT
NUMBER";
2210    INPUT Y
2220    J1=Y
2230    GOTO 1860
2240    RETURN
2250!          GENERATING WAVEFORM SEQUENCES OF DESIRED
TIME DURATION
2260    J2=J2+1
2270    PRINT "AT START THERE MUST BE ONLY ONE ZERO
SEGMENT IN SEQUENCE"
2280    DISP "ENTER ADDRESS OF ZERO WAVEFORM SEGMENT";
2290    INPUT Z
2300    DISP "ENTER SEQUENCE POSITION OF ZEROS";
2310    INPUT I2
2320    FOR I=J2 TO I2+2 STEP -1
2330      Aa(I)=Aa(I-1)
2340      La(I)=La(I-1)
2350      Ra(I)=Ra(I-1)
2360    NEXT I
2370    DISP "ENTER DESIRED TIME";
2380    INPUT T
2390    IF T<6.1 THEN T=6.1
2400    L=INT(F/8+.5)*8! SEQUENCE MUST BE MULTIPLE OF 8
2410    R1=INT((T-3)/L*F)! REPETITIONS OF SEQUENCE L
2420    T2=T-R1*L/F
2430    L2=INT(T2*F/8+.5)*8
2440    R2=1
2450    Aa(I2)=Z
2460    La(I2)=L
2470    Ra(I2)=R1
2480    Aa(I2+1)=Z
2490    La(I2+1)=L2
2500    Ra(I2+1)=R2
2510    GOSUB 1940
2520    GOSUB 2030
2530    DISP "TO EXIT, ENTER AN E -- TO CONTINUE,
ANYTHING ELSE";
2540    INPUT E$
2550    IF E$="E" THEN RETURN
2560    GOTO 2370
```

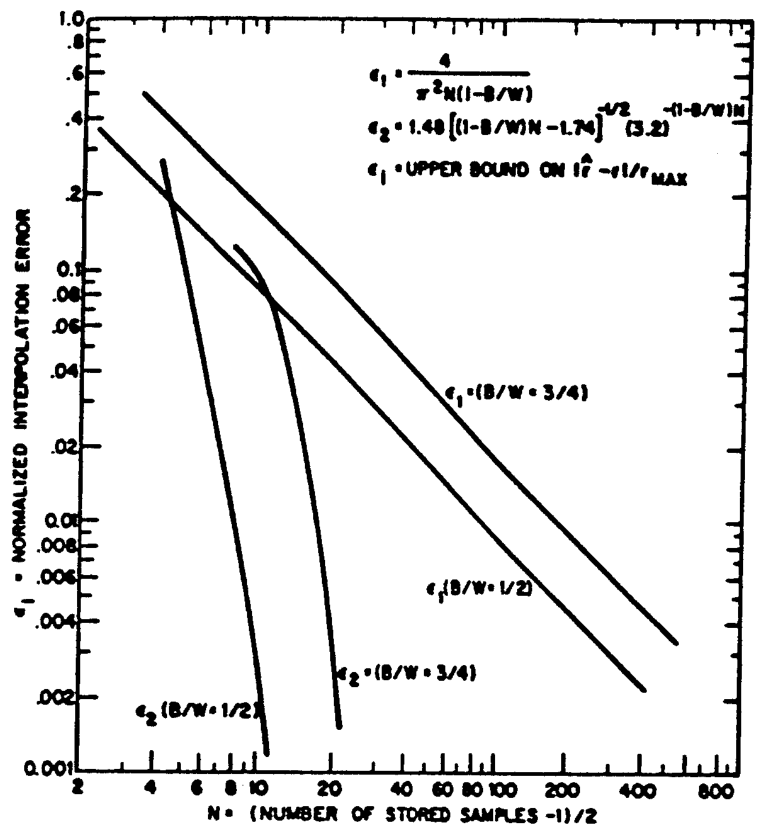
```
2570  ! ROUTINE TO MODIFY CLOCK SELECTION
2580   DISP "enter an EXT     for external clock
otherwise the internal 125MHz is assumed";
2590   INPUT Cls$
2600   IF Cls$="EXT" THEN 2650
2610   F=125
2620   PRINT "INTERNAL CLOCK SELECTED     FREQUENCY
ASSUMED ";F
2630   OUTPUT 719;"CLKSEL INT"
2640   RETURN
2650   F=120
2660   PRINT "EXTERNAL CLOCK SELECTED     FREQUENCY
ASSUMED ";F
2670   OUTPUT 719;"CLKSEL EXT"
2680   RETURN
2690  !GET A CATALOG OF DISC
2700   CAT
2710   RETURN
2720   END
```

Inventor: Waters
Serial No.:

PATENT APPLICATION
Navy Case No.: 75,573

ABSTRACT

This invention is a communication system utilizing sharply bandlimited waveforms for sampled data communications computed for each sample value in a sequence of data samples. The computed waveform is centered at a sample point, weighted by a corresponding data value, and truncated outside an appropriate time interval. Sampled digital data in a sequence of data samples is received by a computer and scaled according to the value $r(t_n)$. A waveform generator controlled by a computer which generates an interpolation waveform for each data sample and weights each waveform with the data and forms the sum of weighted waveforms to form the output signal in digital form which is converted to analog form. The analog output of the waveform generator is up-converted to produce a desired carrier frequency which can then be multiplexed, if required, and passed through a low-pass filter to filter out any harmonics generated. The filtered analog signal is applied to the intermediate frequency (IF) stage of a transmitter for generation of sampled data signal output as radio frequency (RF) energy which can be received by any standard receiver.



$\epsilon_1 = \frac{4}{\pi^2 N(1-B/W)}$
$\epsilon_2 = 1.48\left[(1-B/W)N - 1.74\right]^{-1/2}(3.2)^{-(1-B/W)N}$
ϵ_1 = UPPER BOUND ON $|\hat{r} - r|/r_{MAX}$
ϵ_1 = (B/W = 3/4)
ϵ_1(B/W = 1/2)
ϵ_2 = (B/W = 3/4)
ϵ_2 (B/W = 1/2)
ϵ_1 = NORMALIZED INTERPOLATION ERROR
N = (NUMBER OF STORED SAMPLES -1)/2

FIG. 1

9

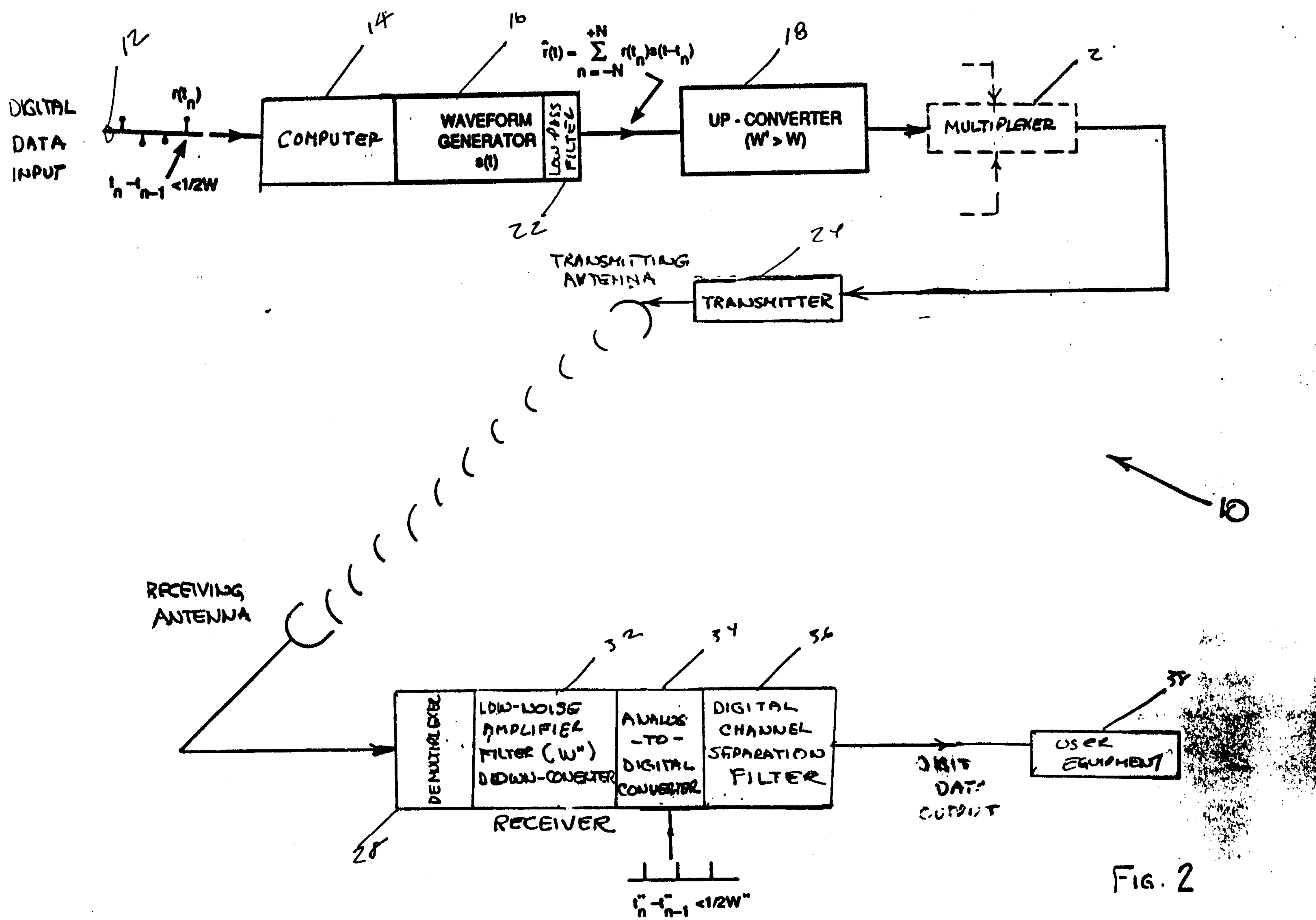

DIGITAL DATA INPUT
12
$r(t_n)$
$t_n - t_{n-1} < 1/2W$
14
COMPUTER
16
WAVEFORM GENERATOR s(t)
LOW PASS FILTER
22
$\hat{r}(t) = \sum_{n=-N}^{+N} r(t_n)s(t-t_n)$
18
UP-CONVERTER (W' > W)
2
MULTIPLEXER
24
TRANSMITTER
TRANSMITTING ANTENNA
10
RECEIVING ANTENNA
DEMULTIPLEXER
32
LOW-NOISE AMPLIFIER FILTER (W") DOWN-CONVERTER
34
ANALOG -TO- DIGITAL CONVERTER
36
DIGITAL CHANNEL SEPARATION FILTER
RECEIVER
28
$t''_n - t''_{n-1} < 1/2W''$
DATA OUTPUT
38
USER EQUIPMENT

FIG. 2

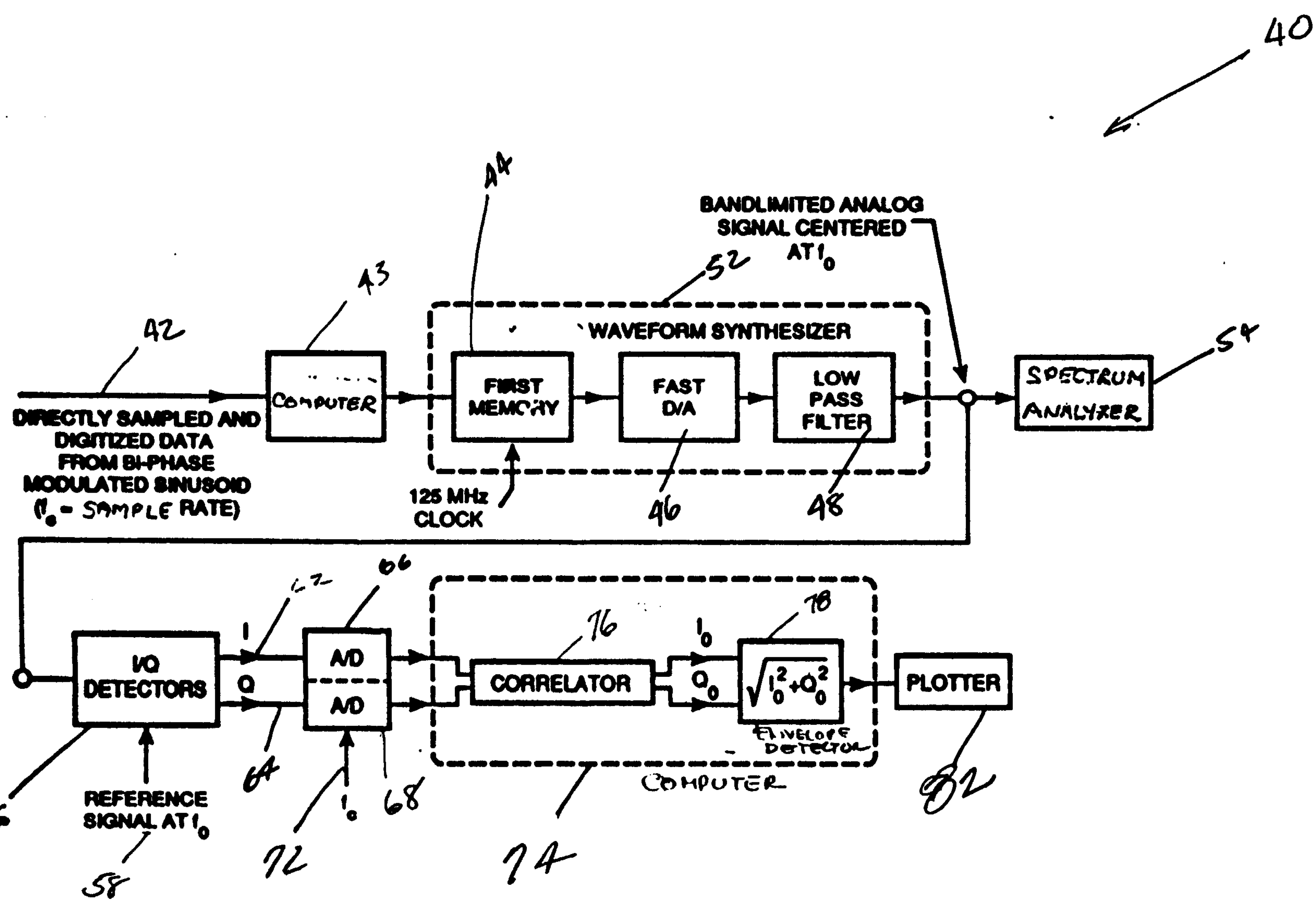
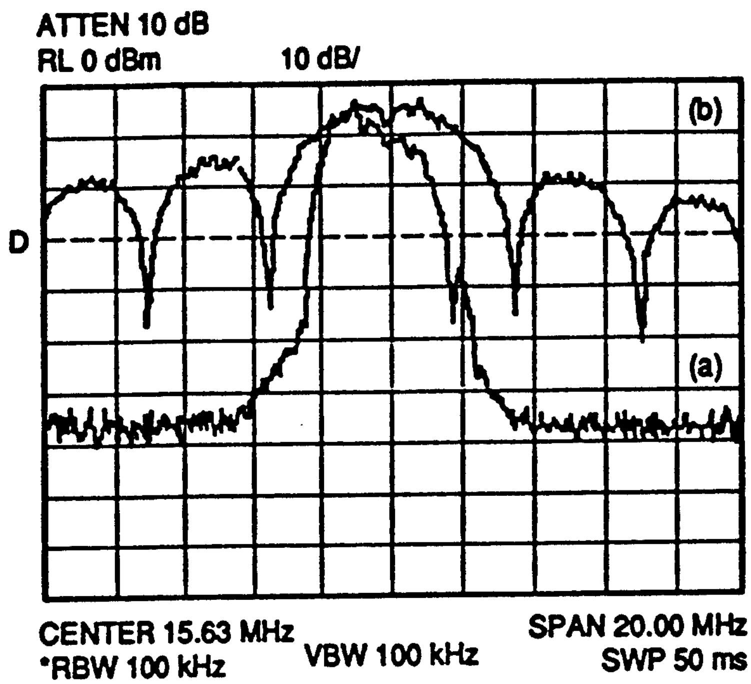

40
42
43
44
52
54
BANDLIMITED ANALOG SIGNAL CENTERED AT f_0
WAVEFORM SYNTHESIZER
COMPUTER
FIRST MEMORY
FAST D/A
LOW PASS FILTER
SPECTRUM ANALYZER
DIRECTLY SAMPLED AND DIGITIZED DATA FROM BI-PHASE MODULATED SINUSOID (f_0 = SAMPLE RATE)
125 MHz CLOCK
46
48
66
78
76
I
Q
I_0
Q_0
I/Q DETECTORS
A/D
A/D
CORRELATOR
$\sqrt{I_0^2+Q_0^2}$
PLOTTER
ENVELOPE DETECTOR
COMPUTER
56
64
68
82
REFERENCE SIGNAL AT f_0
f_0
58
72
74

Fig 3



ATTEN 10 dB
RL 0 dBm
10 dB/
(b)
D
(a)
CENTER 15.63 MHz
*RBW 100 kHz
VBW 100 kHz
SPAN 20.00 MHz
SWP 50 ms

FIG. 4

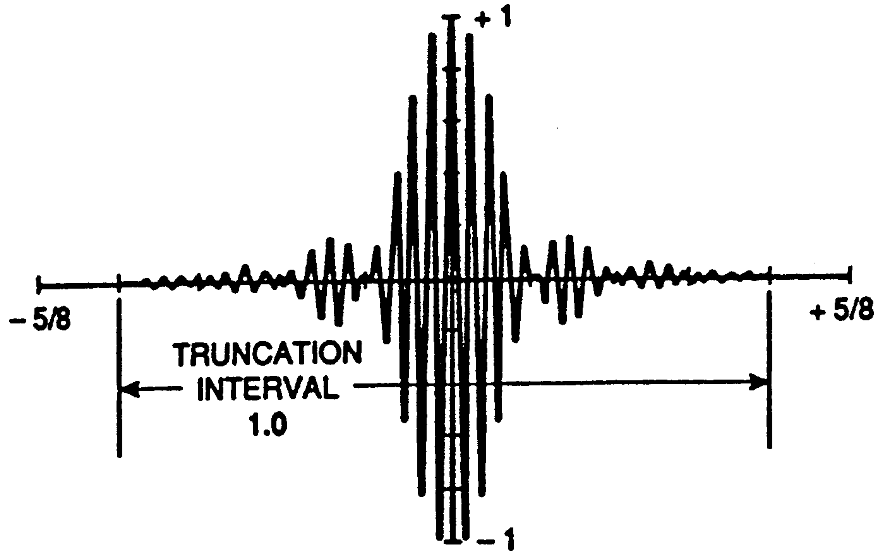

+ 1
– 1
– 5/8
+ 5/8
TRUNCATION
INTERVAL
1.0

Fig 5a

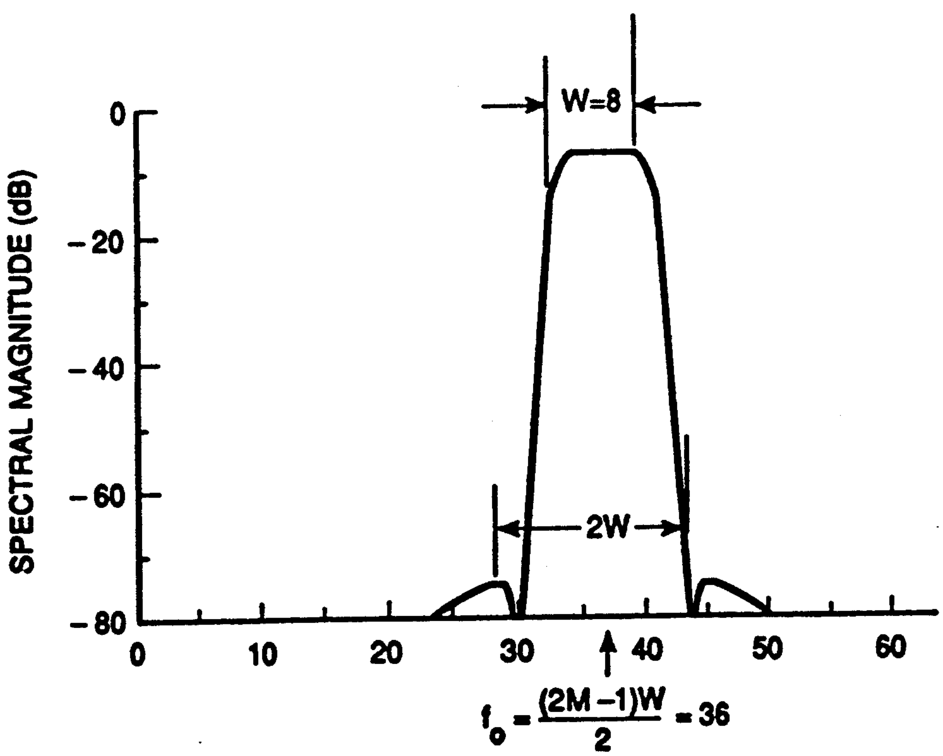

W=8
2W
SPECTRAL MAGNITUDE (dB)
0
– 20
– 40
– 60
– 80
0
10
20
30
40
50
60
f_o = (2M – 1)W / 2 = 36

Fig 5b

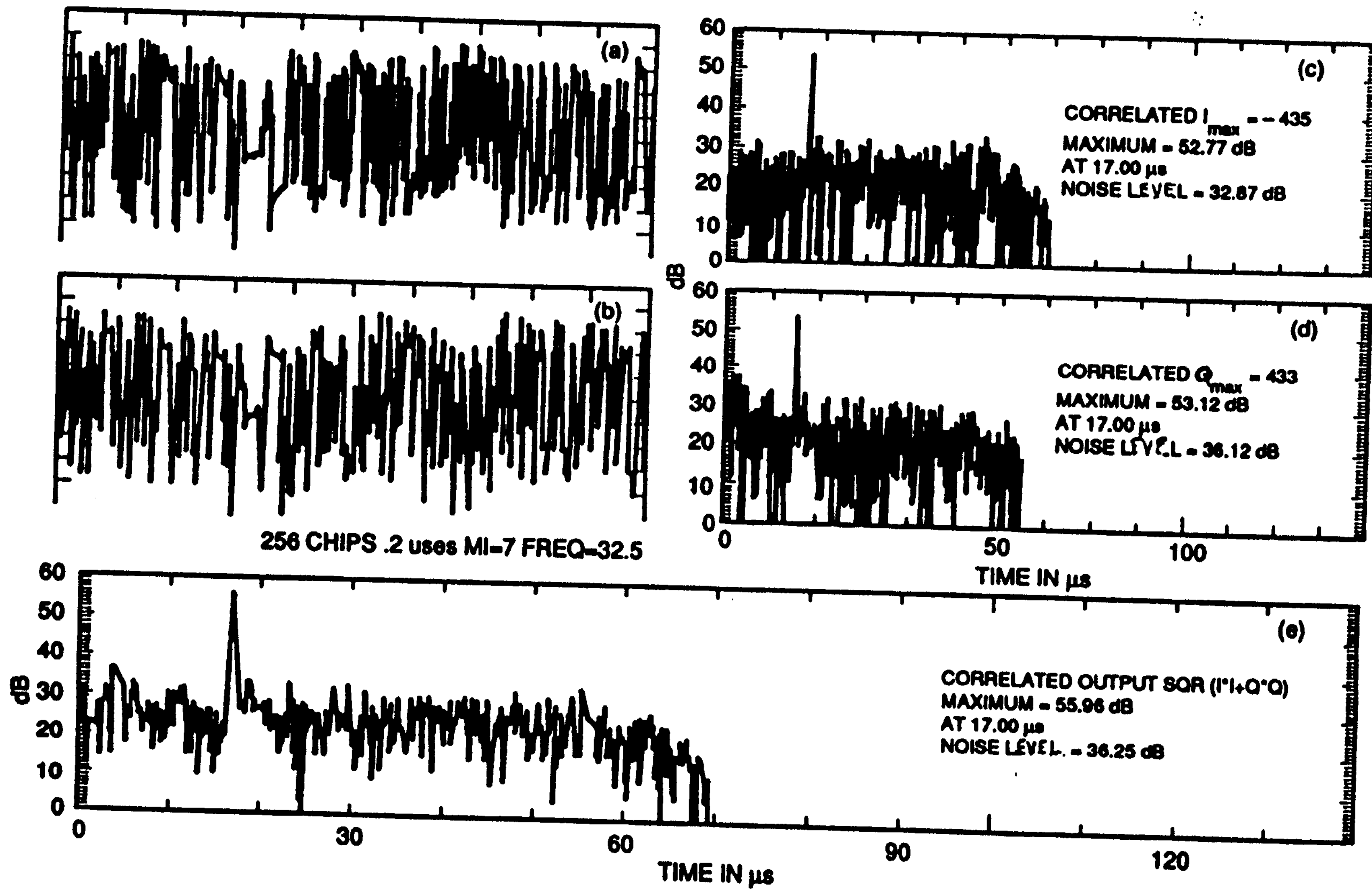


Fig 5c

Fig 5d

Fig 5e