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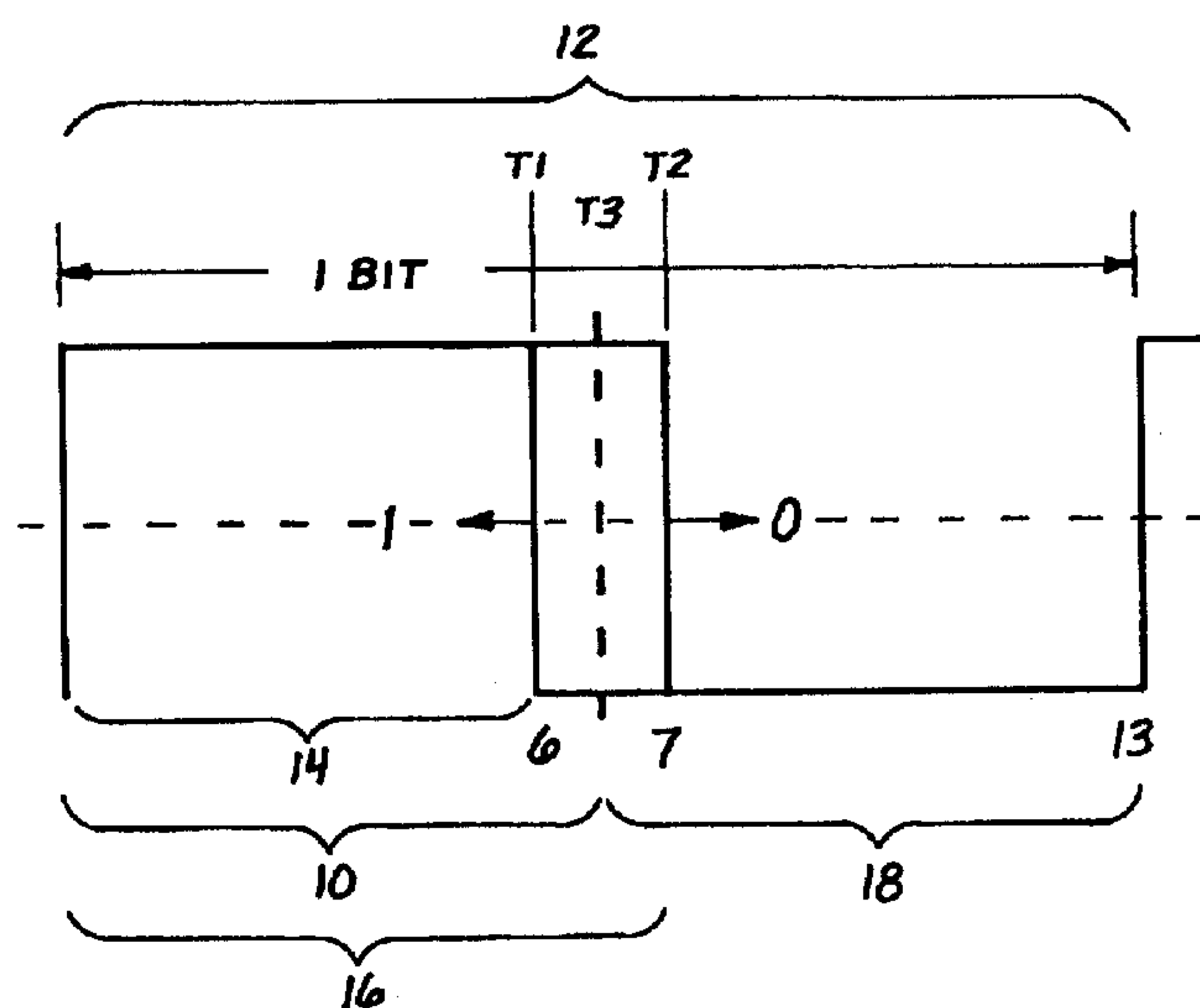
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(54) **MODULATION NUMERIQUE A BANDE LATÉRALE UNIQUE
AVEC SUPPRESSION DE PORTEUSE**

(54) **DIGITAL MODULATION EMPLOYING SINGLE SIDEBAND
WITH SUPPRESSED CARRIER**



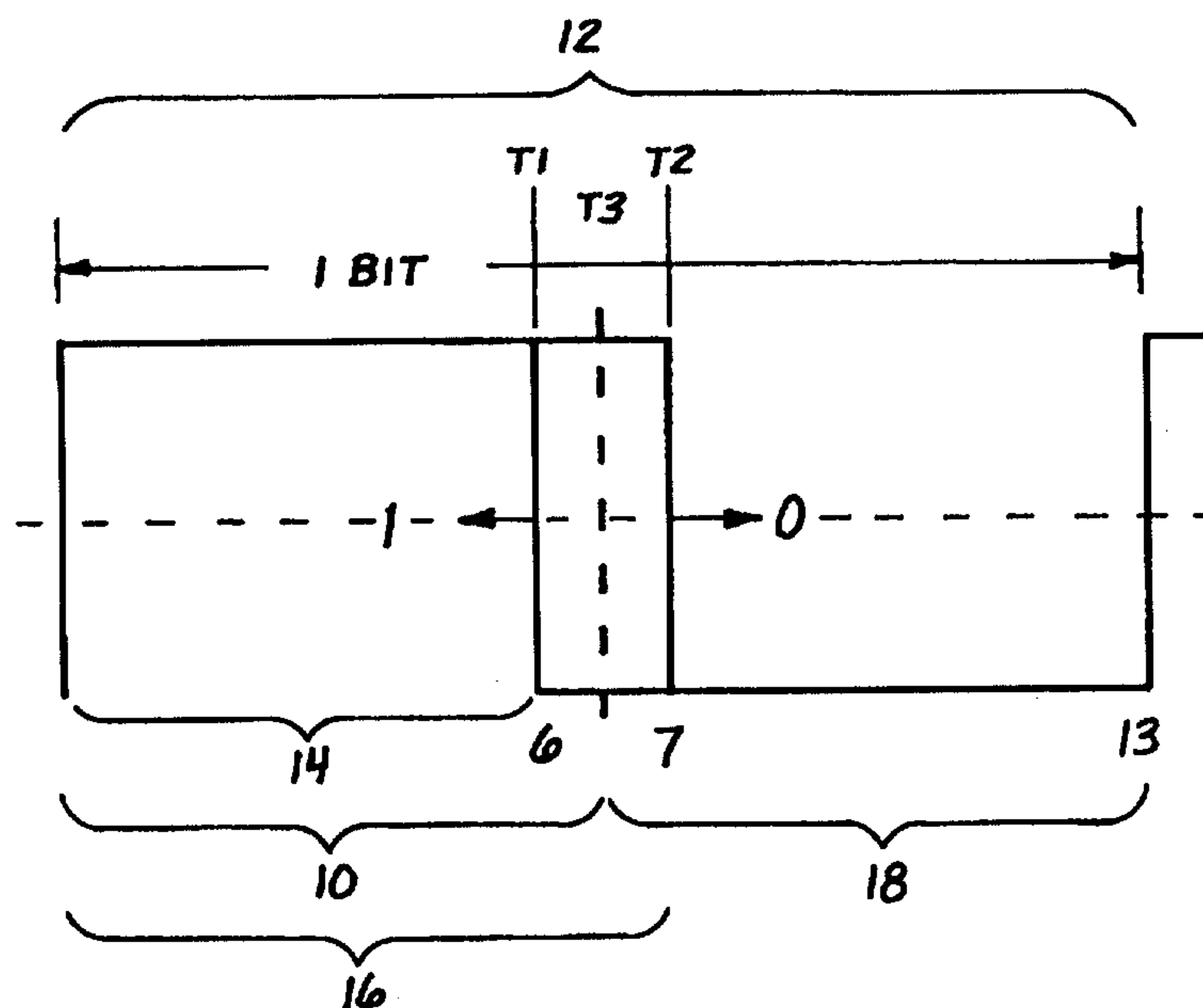
(57) La présente invention concerne un procédé permettant de transmettre des signaux numériques codés "biphase", consistant à recevoir l'un des bits de données d'entrée de la multiplicité, chaque bit étant caractérisé par une longueur (12) de bit de données temporelles, un centre de bit temporel (T3) et un état temporel indicatif du un ou du zéro logique. Le procédé consiste également à générer un signal numérique de bande de base codé, caractérisé par un état temporel indicatif d'un état logique, ce signal modifiant son état temporel si le bit de données d'entrée un modifie son état temporel avant (T1 ou 6) ou après (T2 ou 7) le centre de bit temporel. Avec ce codage, on génère une bande de base unique indépendamment, par synthèse numérique. La détection de la bande de base unique ne requiert pas de réinsertion de la porteuse.

(57) A method of transmitting "biphase" encoded digital signals comprised of the steps of receiving one of a multiplicity of input data bits, wherein each data bit is characterized by a temporal data bit length (12), a temporal bit center (T3) and a temporal state indicative of a logical 1 or 0, and generating an encoded digital baseband signal characterized by a temporal state indicative of a logical state therein the encoded digital baseband signal changes its temporal state according to whether the one input data bit changes its temporal state at a time prior to (T1 or 6) or after (T2 or 7) the temporal bit center (T3). A single sideband with this coding is generated independently by digital synthesis. Detection of the single sideband does not require re-insertion of the carrier.

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(54) Title: DIGITAL MODULATION EMPLOYING SINGLE SIDEBAND WITH SUPPRESSED CARRIER**(57) Abstract**

A method of transmitting "biphase" encoded digital signals comprised of the steps of receiving one of a multiplicity of input data bits, wherein each data bit is characterized by a temporal data bit length (12), a temporal bit center (T3) and a temporal state indicative of a logical 1 or 0, and generating an encoded digital baseband signal characterized by a temporal state indicative of a logical state therein the encoded digital baseband signal changes its temporal state according to whether the one input data bit changes its temporal state at a time prior to (T1 or 6) or after (T2 or 7) the temporal bit center (T3). A single sideband with this coding is generated independently by digital synthesis. Detection of the single sideband does not require re-insertion of the carrier.

DIGITAL MODULATION EMPLOYING SINGLE SIDEBAND WITH SUPPRESSED CARRIER

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1. *Field of Invention*

The field of endeavor of the invention is an apparatus and method for digital, single sideband, transmissions that does not generate or employ a carrier frequency.

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2. *Background of the Invention*

Single sideband transmissions without carriers is well known. These methods have varying bandwidth efficiencies which have been summarized in Applied Microwaves and Wireless Magazine, July/August 1997. In general, most of the existing methods modulate a carrier, which is suppressed in the modulator, then recovered and re-inserted in the receiver.

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While single sideband digital modulation is possible, it has been rarely utilized to date. One method is called Variable Phase Shift Keying (VPSK) as disclosed in U.S. Patents 4,742,532 and 5,185,765. This method utilizes a data encoding method and single sideband transmission to achieve bandwidth efficiencies of as high as 15.3 bits/sec./Hz. A carrier is required to recreate the single sideband suppressed carrier signal and to decode the signal in the receiver. It has been found that proper encoding of the signal at baseband can result in a very narrow frequency band being occupied at baseband and that this can in turn be transmitted single sideband-suppressed carrier (SSB-SC) in a very narrow RF bandwidth. Bandwidth efficiencies as high as 50 bits/second/Hz have been demonstrated in the laboratory with 26-27 bits/sec/Hz being achieved in usable hardware. A comparison of this method,

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which is called Very Minimum Shift Keying (VMSK) with the VPSK method appeared in IEEE Transactions on Broadcasting (January 1997).

Single sideband digital modulation, which involves baseband encoding, a carrier and filtering to remove the unwanted sideband, are unnecessarily
5 complex.

High bandwidth efficiency methods are thought to generally lose power as a result of the bandwidth compression. Carson's Rule and Shannon's Limit are engineering laws which state that there is a proportionality between the signal-to-noise ratio and the bandwidth efficiency.

10 What is needed is some type of design and method which maintains the same signal to noise ratio regardless of bandwidth efficiency.

What is needed is some type of design and method which does not lose power as a result of the bandwidth compression.

15 **Brief Summary of the Invention**

The invention is an encoding method for digital baseband signals comprising the steps of dividing a data bit interval having a bit center into at least two time apertures. Each time aperture in turn is divided into one or more time segments. In the illustrated embodiment there are only two
20 apertures, and the data bit interval is divided into 13 segments of which the first 6 are in the first aperture and the last 7 are in the second aperture. The center of the data bit interval falls exactly between the 6th and 7th segments. A first subset of the plurality of time segments in a first one of the apertures is selected. The first subset of the plurality of time segments ends prior to the bit
25 center. A second subset of the plurality of time segments in a second one of the apertures is selected from the remaining plurality of time segments. The second subset of the plurality of time segments ends after the bit center. The digital baseband signals is encoded to correspond to the first and second

subset of time segments. This method is defined in this specification as "aperture coding".

The method further comprises the step of changing the polarity of said digital baseband signals responsive to the starting time of the data bit interval. then encoding a polarity reversal associated with the data bit based upon time duration of the first and second subset of time segments. In particular the first and second subset of time segments have relative durations representative of digital ones and zeros associated with the digital baseband signal. The digital baseband signal has a rising edge and falling edge. In one embodiment the method comprises using the rising edge as a clock timing to denote the leading edge of the data bit interval. The falling edge is then timed to occur as at a time which is determined depending on whether the digital baseband signal is to be understood as a digital one or zero.

In a second embodiment the role played by the rising and falling edges are reversed. The method comprises the steps of using the falling edge as a clock timing to denote the leading edge of the data bit interval. The rising edge is then timed to occur as at a time which is determined depending on whether digital baseband signal is to be understood as a digital one or zero.

Clearly in both embodiments a sum of the time durations of the first and second time subsets equals the sum of the time durations of all of the plurality of time segments in the two apertures of which, in the illustrated embodiment, the first 6.5 segments are in the first aperture and the last 6.5 in the second aperture.

Instead of phase or time differences between the digital baseband signals depending on whether it is to be understood as a 1 or 0, in another embodiment the first and second subsets of time segments are converted to first and second frequencies generated by a frequency shift means or by a direct digital synthesizer, which produces a modulation index which is less

than approximately 0.1. This frequency conversion alters the nature of the modulated signal from that of a carrier plus two sidebands which would normally occur in frequency modulated signals to that of a narrow biphasic single sideband alone.

5 In another embodiment a third frequency corresponding to a third subset of time segments is obtained by dividing the data bit interval by two, that is by doubling the number of segments in the data bit interval of the digital baseband signal so that there are three times at which a transition can occur for the digital baseband signal. For example, whereas in the illustrated
10 embodiment the data bit interval was divided into 13 time segments with a transition at the 6th and 7th segments denoting a 1 or 0, in another illustrated embodiment the data bit interval is divided into 26 time segments with a transition at the 12th and 14th segments denoting a 1 or 0 and a transition at the 13th segment denoting a repeat of the value of the preceding bit. Again, these
15 time or phase transition events can be converted and expressed as three different frequencies. Each time segment may remain constant in duration so that the data bit interval is doubled in duration and the data bit rate is halved.

The invention is also a digital data transmission method comprising the steps of aperture encoding a digital baseband signal in the which the aperture
20 coding comprises the steps of dividing a data bit interval having a bit center into a plurality of time segments. A first subset of the plurality of time segments is selected. Each of the first subset ends prior to the bit center. A second subset of the plurality of time segments is selected and they are comprised of all remaining ones of the plurality of time segments not included
25 in the first subset of time segments. Each of the second subset ends after the bit center. A signal to be transmitted is phase or frequency modulated using the aperture coded digital data. The phase or frequency shift modulated signal is then transmitted. The transmission may be either at baseband over wire or

fiber, or at radio frequencies in which the aperture coded digital data is used to modulate a carrier frequency.

The method further comprises superimposing additional information other than data upon the modulated transmitted signal using low level
5 amplitude modulation.

The transmitted aperture coded digital data is a single sideband signal with a suppressed carrier. The sideband is located at a frequency equal to the suppressed carrier, but is offset positively or negatively from the carrier by not more than the data rate or in a second embodiment by not more than one half
10 of the data rate. The suppressed carrier can be a multiple of the data rate.

The method may further comprise passing the single sideband through a very narrow bandpass filter, which passes only a center frequency of the sideband plus or minus a first predetermined variance so that a very high bandwidth efficiency and excellent signal-to-noise ratio are obtained.

15 The method still further comprises receiving and decoding the phase or frequency modulated transmitted signal with a phase detecting or discriminating circuit which is responsive to phase or frequency shifts respectively, but does not utilize a carrier or other reference signal.

In another embodiment the method further comprises receiving and
20 decoding the phase or frequency modulated transmitted signal with a narrow bandwidth crystal filter, and a sample and hold or correlator circuitry.

In yet another embodiment the method further comprises receiving and decoding the phase or frequency modulated transmitted signal by differentiating and Manchester decoding the transmitted signal.

25 In yet another embodiment the method further comprises receiving and decoding the phase or frequency modulated transmitted signal by detecting peaks of the transmitted signal as the aperture coded digital data.

The phase or frequency modulated transmitted signal is transmitted without any coherent carrier frequency so that transmission is insensitive to Doppler frequency offsets when transmitting by means of a relay or satellite.

The invention is also defined as an apparatus comprising an encoder
5 for aperture coding a source signal, a transmitter for transmitting the aperture coded source signal, a receiver for detecting and receiving the aperture coded source signal, and a decoder for decoding the aperture coded source signal. Typically, the aperture encoder and transmitter are in one unit at one location and there is one or more receivers and decoders at one or more other
10 locations in other units. Again the aperture coding comprises the steps of dividing a data bit. having a bit center. into a plurality of time segments, and selecting a first subset of time segments from the plurality of time segments. Each of the first subset of time segments end close to the bit center. A second subset of time segments is selected from the remaining plurality of time
15 segments. Each of the second subset of time segments start after the first subset of time segments. The duration of the first and second subsets of time segments depend upon whether the source signal is a digital one or zero.

The encoded digital baseband signal has two polarities. The encoded digital baseband signal comprises changing the polarity of the encoded digital
20 baseband signal responsive to the starting time of the input data bit, then holding that polarity for the duration of the first subset of time segments, and then reversing the polarity of the digital baseband signal for the duration of the second subset of time segments.

The invention now having been briefly summarized, its various
25 embodiments may be visualized in the following drawings where like elements are referenced by like numbers.

BRIEF DESCRIPTION OF DRAWINGS

Fig. 1 is a diagram of the aperture code for a single bit interval.

Fig. 2a is a graph of the frequency spectrum of an aperture coded signal according to Fig. 1 shown at baseband or as a single sideband located above the carrier.

5 Fig. 2b is a graph of the frequency spectrum of an aperture coded signal according to Fig. 1 shown after narrow band filtering.

Figs. 3a - e shows encoding and decoding method for a baseband signal of the prior art.

10 Fig. 4 is a diagram which illustrates a single sideband signal, along with a reinserted carrier f_c , as generated by a direct digital synthesizer or a numerically controlled oscillator..

Fig. 5 is a block diagram of a detector circuit utilizing a ringing coil method of carrier recovery..

15 Fig. 6 is a block diagram of an RF modulator applicable to the present invention.

Fig. 7 is a block diagram of a logic circuit using an alternate method to generate two frequencies in a direct digital synthesizer.

Fig. 8 is a timing diagram of a signal from a quadrature or similar detector which has an "Eye Pattern."

20 Fig. 9 is a diagram of the prior art VPSK encoding method of a bit interval referred to as slip coding.

Fig. 10 is a block circuit diagram of a device for dividing the aperture code by two to yield a three time difference code, which can be converted into three frequencies using a device such as illustrated in Fig. 16.

25 Fig. 11 is a block circuit diagram of a decoder usable with a second embodiment of the invention.

Fig. 12 is a block circuit diagram of a decoder for a Manchester code usable with the present invention.

Fig. 13 is a diagram which illustrates an alternate method of detection which comprises the use of peak detectors and an R/S flip/flop.

Fig. 14 is a phasor plot for a detector usable with the two- and three-time-signal methodologies of the invention for the timing diagrams shown in
5 Fig. 8 and Figs. 15a - 15g.

Figs. 15a - 15h are a timing diagrams for three bit patterns using a three phase encoding which shows the number of phase changes to go from 1 to 0, or 0 to -1.

Fig. 16 is a block circuit diagram of an encoder circuit used to program
10 a direct digital synthesizer to utilize three frequencies in Figs. 15a - 15g.

Fig. 17 is a block circuit diagram of a detector usable according to the phasor diagram of Fig. 14.

The invention and its various embodiments are now set forth in the following detailed disclosure, which are to be understood as illustrating the
15 invention and not limiting it as claimed.

Detailed Description of the Preferred Embodiments

A method for transmitting "biphase" encoded digital signals includes the steps of defining time apertures over the data bit interval and dividing the
20 aperture into a plurality of segments. A first segment is selected from the plurality of segments of the aperture, dependent upon whether the bit is a digital one or zero. A second segment is selected from the plurality of segments of the aperture, so as to take up the remaining aperture of the data bit interval. An encoded signal having phase transitions or frequency
25 components corresponding to the first and second segments of said data bit interval is transmitted. A narrow spectrum results containing no low frequency components. The spectrum is separated from 0 Hz by an amount equal to the data rate or in another embodiment by $\frac{1}{2}$ the data rate. The transmitted

bandwidth is much narrower or higher in terms of bandwidth efficiency in bits/second/Hz, than currently employed methods. The invention has broad utility in radio, microwave, and satellite applications.

More specifically, the encoding method of the invention generates a
5 digital baseband signal derived from a multiplicity of input data bits. Each input data bit is characterized by a temporal data bit length, a temporal bit center and a temporal state indicative of a logical 1 or 0. The encoding method comprises the steps of receiving one of the multiplicity of input data bits, and generating an encoded digital baseband signal characterized by a
10 temporal state indicative of a logical state. which encoded digital baseband signal changes its temporal state according to whether the one input data bit changes its temporal state at a time prior to or after the temporal bit center.

A single sideband with this coding is generated independently by digital synthesis and is transmitted over wire lines or radio frequency channels for
15 detection by a detector circuit. Detection does not require the re-insertion of the carrier. The characteristics of the sideband alone are reproducible by direct digital synthesis. Although the signal can be detected by normal means, that is by reconstructing a carrier and reinserting it in the detector to recover the encoded baseband signal, this is not necessary. There is sufficient
20 information in the sideband alone to be detected without a carrier.

While phase modulation is discussed in the illustrated embodiment, time, frequency and phase are interchangeable, namely

$$T = \Delta\Phi / (2 \pi \Delta f)$$

where T is the time period, $\Delta\Phi$ the phase difference, and Δf the
25 frequency difference in the signals.

The present invention encodes the data into a "biphase" format, which has a spectrum located above and below the carrier by the bit rate, or above zero Hz at baseband. This type of encoding method, which has been

generally named "Aperture Coding", is specifically shown in the case of the invention in Fig. 1. The coding method of the invention when used at baseband has a code width of one bit length. The first half aperture 10 of the bit interval 12 has a specific time duration or length, namely $6/13$ of the total bit width in the case where the bit is divided into 13 segments. A decoding circuit, on detecting a time of duration 14 (the 6th segment of the 13 segments) will automatically designate this bit as a digital one. If the time duration is stretched to cover $7/13$ of the bit interval, duration 16, it will be decoded as a zero. The zero crossing fits into the left half bit aperture 10 if it is a one and into the right half bit aperture 18 if it is a zero.

It is not necessary to divide bit width 12 into 13 parts. Any pair of apertures can be substituted, for example a first aperture of $11/23$ of width 12 and a second aperture of $12/23$ can also be used. Smaller apertures can also be used for decoding. For example, bit interval 12 can be divided into 13 small apertures with the zero crossing having to fit specifically into a $6/13$ or $7/13$ aperture period instead of the larger first half/second half apertures.

This waveform is actually the data clock waveform with a phase distortion at the center. The waveform can be defined by a Fourier series having a base frequency equal the bit rate plus odd harmonics and a varying low frequency amplitude A_0 component depending on the data pattern.

An encoder circuit for generating the aperture codes of Fig. 1 at baseband is shown in block diagram in Fig. 10. An oscillator 20 operating at 13 times the bit rate is used to clock two divide-by-13 counters 22 and 24. Counter 22 provides a clocking signal, DATA CLOCK, for the incoming data on input 26 to clock a flip flop 28 which is set to automatically reset at the end of each bit interval, resulting in a very narrow spike or one shot output on output 30. The automatic reset is delayed by the delay time required for the clock signal to pass through flip flop 28 and results in a very narrow spike or one

shot on output 30, which is the signal, RESET, after each positive incoming clock voltage transition. RESET resets the counter 24 and in a first embodiment clears JK flip flop 32.

Counter 24 has two outputs 34 and 36. Output 34 provides a 6 count
5 and output 36 provides a 7 count. Data, DATA IN, on input 38 is added to the 6 or 7 signal from outputs 34 and 36 respectively in NAND gates 40 and 42 respectively. DATA IN is coupled to NAND gate 42 and DATA IN is inverted by inverter 44 and coupled to NAND gate 40. The outputs of NAND gates 40 and 42 are coupled to the inputs of OR gate 46. The output of OR gate 46 is
10 coupled to the clock input of flip-flop 32. If DATA IN is a one, JK flip flop 32 is set via gates 42 and 46 after a 6 count. If it is a zero, JK flip flop 32 is set after a 7 count via gates 40 and 46. The encoded data is available in complementary form on outputs 48 and 50 of flip-flop 32 as shown in timing lines 49 and 51 respectively.

15 Information encoded in this manner on outputs 48 and 50 can be used to modulate an RF carrier, or it can be used as a baseband signal for transmission over wire lines. In the latter case, the clock is recovered from the rising edge of the signal and the data from the falling edge. As an RF modulated signal the transmission signal consists of a carrier with two
20 sidebands, an upper and a lower sideband, only one of which needs to be transmitted. The sideband has the appearance shown in Fig. 2a.

When used at baseband, the signal can be limited to remove any AM components, such as noise from the phase modulated signal. It is thus possible to superimpose a low level, low frequency, AM signal for control or
25 other purposes on top of the phase modulated data signal. The data rate for the phase modulated signal is generally much higher than that of the AM signal so that the sampling rate eliminates almost all of the phase changes that would be introduced by any such AM modulation.

The frequency spectrums shown in Figs. 2a and 2b consists of a central spike frequency ω_c at the data rate f_b plus the Fourier amplitude products shown as a form of frequency or phase noise at a relatively low level. If the data rate is a submultiple of the modulated carrier, the carrier can be restored from the single sideband frequency by detecting the single frequency spike, dividing it down, then multiplying it back up by the carrier/data rate multiple.

An alternate method to restore the carrier is to use the aperture method described above in Fig. 1 with the method used for VPSK modulation described in *"Improved Data Transmission Using Single Sideband with FM Suppressed Carrier"*, by B. Stryzak and H.R. Walker, Microwave and RF Magazine, Wireless Design Supplement (Nov. 1994). This embodiment of the method uses a coil ringing at 13 times the bit rate assuming a 6,7 (13) code is used as described in Fig. 1. The coil is caused to ring by one shot multivibrator spikes obtained from the zero crossings. The RF frequency must be a multiple of the data rate and the ringing frequency. The circuit using the ring coil method is described below in connection with Fig. 5.

The time periods given in this embodiment can be any pair of time periods. Time is convertible to frequency ($f = 1/t$). Thus instead of having two time periods, two pulses of equal time duration at two corresponding RF frequencies can be used. When mixed with a coherent carrier, the basic Fourier frequency of the aperture encoded waveform appears as the difference frequency with decodable zero crossings. The difference frequency can then be decoded to ones and zeros by squaring the waveform and then determining the aperture time. Thus in the following it must be understood that time and frequency are interchangeable in significance.

The use of the two frequencies of the invention can be compared to the two frequencies used for prior art Gaussian Minimum Shift Keying (GMSK) in that they result in a phase or frequency modulated signal. In the case of

GMSK, the frequencies are $\pm 1/4$ the data rate. The resulting GMSK spectrum results in a central frequency spike similar to that shown in Figs. 2a and 2b bit has two sidebands at 0.24 times the peak level. The modulation index in FM/PM terms is 0.5. The modulation index is defined as $\Delta f / f_m = M$. GMSK can be created by filtering the data input, the modulator output, or by means of a digital synthesizer. GMSK is a double sideband method which does not have a suppressed carrier. Alternate methods of generating GMSK are FM, PM or FSK (frequency shift keying). A modulation index of 0.5 is essential to GMSK operation.

10 In the present method by contrast, the frequency shift or deviation used is much lower than that used for GMSK. In lieu of a modulation index of 0.5, a modulation index between approximately 0.04 and approximately 0.1 is being used in practice. It has been noted that when GMSK is derived using FM, the sidebands are Bessel products, while using FSK generates Fourier products.

15 The significance of this relates to the distribution of the information bearing energy between the carrier and sidebands. This is not be obvious without careful study. Utilizing a digital synthesizer and a modulation index of 0.5, the sidebands appear as expected at the carrier frequency plus or minus the modulation frequency, f_m . It has been noted however that when the

20 modulation index is further reduced, the Bessel products disappear and the remaining signal consists of the central frequency which is now $f_c + f_m$ with Fourier products at a level about - 40 dB below this central frequency. Thus only 1 ten thousandth of the energy lies in the Fourier products. This synthesized signal can be further filtered to remove these Fourier amplitude

25 products leaving only the central spike shown in Fig. 2b plus or minus a small deviation frequency or phase.

The meaning and importance of this is not intuitively obvious. What has happened is that the signal is no longer a carrier with two sidebands, but

becomes a single sideband alone. It has been generated without a carrier ever having been present. This single sideband signal can be transmitted and detected by the normal means described above using a phantom carrier. The fact that the signal appears to be a single frequency line would appear to
5 violate Carson's Rule, which states that $\text{bandwidth} = 2 (f_m + \Delta f)$. This is not the case, however, since inserting the phantom carrier restores the signal to the full Nyquist bandwidth required by Carson's Rule, that is $f_c \pm f_m$. The full modulation or Nyquist noise bandwidth is present, but it has not been transmitted.

10 It is to be noted also that since only a very narrow bandwidth has been transmitted, the noise bandwidth at the receiver is much less than that called for by Carson's Rule and the Nyquist bandwidth. This is of great importance in the mathematical analysis of the signal. These relationships can be seen in the comparative frequency spectrums of Figs. 3a - 3e, which illustrate prior art
15 single sideband transmission. Fig. 3a shows the baseband signal as generated by the aperture code. The information is centered a 1.0 bit data rate from the zero frequency. The signal of Fig. 3a can be used to modulate a carrier resulting in the spectrum seen in Fig. 3b, where two sidebands are created on each side of the carrier frequency, f_c . The signal is then transmitted
20 as a single sideband signal at $f_c + \text{data bit rate}$ with a suppressed carrier as shown in Fig. 3c. At the detector, a coherent carrier, f_c , is reinserted resulting in the restoration of the original baseband signal as shown in Fig. 3d and Fig. 3e.

The methods of Figs. 3a - 3e are unnecessarily complex. Using a direct
25 digital synthesizer as described above, with two frequencies replacing the two time periods of the aperture coding, the signal as generated is that shown in Fig. 3c. No carrier was ever used or needed, but detection can employ a reinserted phantom carrier with the results shown in Fig. 3d and Fig. 3e.

The single sideband signal as generated by the direct digital synthesizer or numerically controlled oscillator is diagrammatically shown in Fig. 4, along with a reinserted carrier f_c which sets the bit width 56. The resulting beat or difference frequency $f_1 + f_2$ or $f_1 - f_2$, has negative going zero crossings to the right or left of the bit width center 54 as shown in Fig., 4. By utilizing frequencies instead of time, the aperture coding pattern of Fig. 1 can be reconstructed.

Fig. 5 is a block diagram of a detector circuit utilizing a ringing coil method of carrier recovery. The two frequency input data is provided on input 62 to detector 60. An oscillator 58 operating at an intermediate frequency is used with coherent detector 60 to obtain a detected waveform as shown in Fig. 4. The detected waveform is squared in a limiting amplifier 64 which is coupled to a bidirectional one shot circuit 66 which in turn is coupled to a ringing coil 68 that obtains energy from one shot circuit 66 to ring at a frequency equal to the number of apertures in the code times the data rate. The aperture number is defined as the segment number of the aperture in the 6, 7 (13) coding in which the zero crossing occurs. Decoder 70 in Fig. 5 has its input coupled to ring coil 68 and contains a divide by N counter. The function of the divide by N counter in decoder 70 is to function as a phase comparator to yield an automatic frequency control (AFC) voltage 72 that causes oscillator 58 to lock in frequency to the correct carrier frequency. To provide a reference for AFC clock oscillator 58 is divided down to obtain a frequency equal to that from ringing coil 68. A phase comparator in decoder 70 creates an AFC voltage on line 72 used to change the frequency of oscillator 58 to match the detected signal. Decoder 70 is able to detect early and late crossings that indicate a zero or one. Digital data is output at output 74.

The data input signal from ring coil 68 is input on line 76 to detector 70 as shown in Figs. 5 is the basic Fourier sequence of a two frequency signal indicative of the digital data based on the aperture coding. A suitable decoder 70 is shown in the block diagram of Fig. 11. The decoder can be used at
5 baseband or RF for aperture coding. At baseband the two-frequency signal is coupled to inverter 82. The same circuit can be used at RF by inputting the signal at 76 and going through the mixer 78 to get the baseband signal. In this case the carrier restoration circuits previously discussed must be used to make the oscillator coherent to the incoming signal.

10 In the case of RF aperture coding the data is mixed in mixer 78 with a beat or carrier frequency from oscillator 80. An oscillator 80 provides a clock frequency which can be several parts per million off from the actual data rate. The beat frequency output is coupled through inverter 82 to the clock input of flip-flop 84, and inverted through inverter 86 and coupled to the clock input of
15 flip-flop 88. The signal is inverted in the inverter 86 and causes spikes on the reverse swing of the incoming data. The incoming baseband signal supplied to inverter 82 is amplified and squared to cause the spiking flip flop 84 to reset both divider 83 and the counter 90. The Q output of flip-flop 84 is coupled to counter 90 which has a 6 count output 92 and a 7 count output 94. 6 count
20 output 92 is coupled to one input of AND gate 96 and a 7 count output 94 is coupled to one input of AND gate 98. The other input of AND gates 96 and 98 is coupled to the Q output of flip-flop 88 which functions as a timing or enable signal for gates 96 and 98. Gates 96 and 98 are activated by the 6 or 7
outputs 92 and 94 respectively of the counter 90 which then pass the received
25 spike on the reverse swing of the incoming data to the RS flip-flop 104 to set its output high or low prior to the mid point of the data clock. Output 100 of AND gate 96 is coupled to the inverted clear input of flip-flop 104. Output 102 of AND gate 98 is coupled to the inverted reset input of flip-flop 104. Flip-flop

104 functions as a sample and hold circuit and thus signifies whether the detected frequency is a 1 or 0. If the output of RS flip flop 104 is high (having been set by a 6 count and pulse), the D input to the D flip flop 108 will be high and a one will be clocked out. If the signal at the D input 106 is low, a zero will
5 be clocked out. Output 106 is coupled to the D input of flip-flop 108 which is clocked by the output 110 of divide by N circuit 83 driven by beat oscillator 80. The Q output 74 of flip-flop 108 is the decoded digital data signal. A data clock for the receiving unit is available through the inverter 107 at point 109. Peak detection without differentiation, as shown in Fig. 13, can also be used.

10 For baseband use, mixer 78 is omitted. For RF it is needed. Oscillator 80 requires a large divide-by-N circuit. The divide-by-N circuit is reset by the positive going spike from Fig. 1. This resetting automatically matches the clock to the signal so that no automatic frequency control is needed. However, a ringing coil attached to the Q output of flip-flop 88 can be used to generate
15 an AFC voltage in Fig. 5.

A reinserted carrier is not actually necessary. There is sufficient information in the central single sideband frequency spike alone to detect the signal with a quadrature, frequency, or phase detector utilizing a narrow band crystal as the phase discriminating element. Thus it is possible to go from the
20 signal in Fig. 3c to Fig. 3e without ever using a carrier at the transmitter or receiver. Detectors capable of going from the signal in Fig. 3c to Fig. 3e without using a carrier are well known in the art and one used in combination with the invention is shown in block diagram in Fig. 17. XOR gate 228 has one input tied high and the other input 236 capacitively coupled to the incoming
25 encoded signal. The output of XOR gate 228 is resistively fed back to input 236 and coupled to one input 246 of XOR gate 230 and input 242 XOR gate 232. Input 238 of XOR gate 230 is tied high. The output 240 of XOR gate 230 is coupled to an LC tuned crystal 234 and thence to the other input 244 of

XOR gate 232, whose RC coupled output 248 is the output shown in waveform 146 of Fig. 8.

Consider now how the signal depicted in Fig. 4 may be generated. Fig. 6 shows an RF modulator or encoder applicable to the present invention for
5 generating a single sideband output signal without use of a carrier. A high frequency oscillator 112 generates a clocking signal that is used by the synthesizers 114 and 116 to synthesize the sideband frequency. Synthesizer 116 determines the clock rate for the incoming data which is coupled to encoder 118 on line 122. Encoder 118 shown in Fig. 6 receives the incoming
10 data on line 120 and converts it to the aperture code or a slip code as described in U.S. Patents 4,742,532 and 5,185,765. Outputs 124 address a read only memory (ROM) 126 that has outputs which program synthesizer 114 to determine what frequency it will synthesize. The synthesized intermediate frequency is provided at the output 128 as the single sideband
15 output. No carrier is present. Synthesizer 116 has a control input 117, LOADING CLOCK, which serves to load a new code from ROM 126 so that direct digital synthesizer 116 outputs a new frequency or phase according to the ROM output.

A simplified circuit is shown in the block diagram of Fig. 7 in which an
20 alternative encoder circuit and method to that shown in Fig. 6 is used. Some digital synthesizers can be toggled between two frequency modes. If the synthesizer is preprogrammed by means of a controlling microprocessor or a ROM sequence, the circuit of Fig. 7 can be used. The data clock to the data source is generated from oscillator 130 through divider 132 and is provided on
25 line 134. The data on line 142 is input to an XOR gate 136 causing synthesizer 138 to toggle at frequency, f_1 , or frequency, f_2 , according to whether the data bit is a one or zero. XOR gate 136 causes polarity reversal in the middle of the bit interval according to the comparison between the clock

polarity and the data polarity. If the data and clock polarities are alike, output 144 is low. If the data and clock polarities are different, output 144 goes high. A change from f_1 to f_2 or vice versa occurs at the center of the bit interval by toggling the synthesizer 138. The circuit including synthesizer 114 may also
5 be modified to operate in a toggled mode.

The detected signal from a quadrature detector or similar detector has an "Eye Pattern" as shown in Fig. 8. This is the pattern for a repeating CC hexadecimal sequence (1100110011 etc.). A digital one is represented by frequency, f_1 , occurring first in the bit period 10 with frequency, f_2 , completing
10 the last half bit period 18. A digital zero has frequency, f_2 , for the first half of the bit period 10 and frequency, f_1 , for the last half 18 as diagrammatically depicted in Fig. 8. When there is a change from 1 to 0, or 0 to 1, the rise or fall has double length or duration equal to a full bit interval. Repeated or
consecutive 1's or 0's have short durations of half bit intervals. The time
15 integral of the rise and fall is detected as a triangular waveshape 146 as shown in Fig. 8. To decode this eye pattern, triangular waveshape 146 is differentiated to obtain a rectangular waveshape 148 that is identical to Manchester code. Rectangular waveshape 148 can then be decoded to obtain ones and zeros. Although the detected result is in the form of a
20 Manchester code, the spectrum employed is not that of Manchester coding at baseband. Manchester coding at baseband would extend from the bit rate f_b down toward 0 Hz, while the spectrum of the present invention is f_b plus or minus a small frequency variation.

A simple decoder for Manchester code applicable to the present
25 invention is given in Fig. 12. The Manchester code is input to D input 150 of flip-flop 152 and to the D input 154 of flip-flop 156. A clock at two times the data rate is provided as the clock input 158 to flip-flops 156, 160 and 162. The Q output 164 of flip-flop 162 is coupled through inverter 166 to the clock input

168 of flip flop 152. The Q output 170 of flip-flop 156 is coupled to both the D input 172 of flip-flop 169 and to XOR gate 174. The other input to XOR gate 174 is coupled to the Q output 176 of flip-flop 160. The output 178 of XOR gate 174 is coupled to the clear input 180 of flip flop 162. Output 164 presents
5 the logically converted Manchester code 148 of Fig. 8 to the binary data output 182 of Fig. 8.

This circuit of Fig. 12 uses a clock at double the bit rate which is divided by 2 in the divider 162 to get the actual clock rate. Manchester coded data from the quadrature detector is input to the D input 150 of the D flip flop 152.
10 A one or zero is output according to whether the first half of the bit is high or low. A shift register is used to obtain a synchronizing pulse for the divide-by-two counter 162. When two half bits alike are in the register, XOR gate 174 has a low output. If two half bits alike are different, the output of XOR gate 174 is high. The divide-by-two counter 162 is reset each time the half bits are
15 different to get the clock edge and data lined up, otherwise they could be off by one half cycle at the clock output 164.

An alternate method and apparatus for detection of the triangular waveform shown in Fig. 8 comprises peak detectors 184 and 186 and an R/S flip/flop 192 shown as a block diagram with the timing diagram of Fig. 13.
20 Utilizing this alternative method there is no need for a differentiator and Manchester decoder. A positive peak detector 184 and negative peak detector 186 are coupled to the triangular waveform input 146 of Fig. 8. The outputs 188 and 190 of positive peak detector 184 and negative peak detector 186 are coupled respectively to the S and R inputs of an RS flip-flop 192. A
25 positive peak sets flip-flop 192 to generate a 1 on output 194 and a negative peaks resets output 194 of flip-flop 192 to a 0 to yield the output waveform 196 shown in Fig. 13. In lieu of differentiating and then detecting a Manchester

code, a peak detector can be used as shown in Fig. 13. The peaks occur at the ends of data bits, hence the output is delayed one data bit respectively.

From Fig. 8 it can be seen that when using frequencies or phase angles of the frequencies the signal need not change at the bit boundaries, but does
5 change at the peaks of waveform 146. Instead, as in Fig. 13, the bit polarity is determined by the rise or fall of triangular waveform 146 and the end of the coded bit is marked by a positive or negative peak. The read out is delayed one bit interval time. It is not necessary to force a transition at the midpoint at the middle of the bit interval.

10 In a second embodiment of the invention the baseband transmission frequency can be cut in half for a given data rate by using three phases or frequencies instead of two together with a sideband synthesizer according to the invention. Fig. 10 shows how the aperture code is generated and then may optionally be divided by two to yield a three time difference code which
15 can be converted into three frequencies. The change from a two frequency to a three frequency code is obtained by the setting of the switch 202. Switch 202 either couples ground or the reset output 30 of flip-flop 28 to the clear input of flip-flop 32. When switch 202 is coupled to reset output 30 the waveforms 51 and 49 in Fig. 10 are obtained on lines 50 and 48 respectively.
20 However, when switch 202 is coupled to ground, reset is ignored and a phase change occurs whenever XOR gate 46 clocks flip-flop 32, which then divides by two, so that waveform 49' in Fig. 10 is obtained on line 48 which is the sum of two time subsets instead, resulting in 12, 13, or 14 aperture intervals, which is a divide-by-two signal. Waveform 49' has three Fourier frequencies which
25 are $13/25$, $13/26$ and $13/27$ times the bit rate. The frequency $13/25$ is assigned to 1, $13/26$ is assigned to the center or hold-the-last-bit value, and $13/27$ is assigned to 0. There is a frequency spread in this case from 0.52 to

0.48 the bit rate or a 0.04 bit rate spread. The total bandwidth indicated is 1/25 of the bit rate or has a bandwidth efficiency of 25 bits/sec/Hz.

The aperture code shown in Fig. 1 has three time points, T1, T2, and a third T3 at the center of the bit which is halfway between T1 and T2. When the aperture divisions are divided by 2, the three aperture points 6, 6.5, and 7 become aperture points 12, 13, and 14, thus making it possible using the positive and negative swing, and peaks at aperture points 12 or 14 to determine polarity (1 or 0) plus a return to the center when a bit is repeated at aperture point 13. The waveform as detected is shown in Figs. 15a - 15h for three examples. The three phase coding is shown in Fig. 14 as including the rest or center point at 200 with the other two phase excursions being plus or minus θ . The three frequency code of the present invention thus resembles the previously patented "slip code", but the encode and decode algorithms differ. This three phase or three frequency method is covered above using the VPSK slip code as an encoding example.

Three code sequences are shown in Figs. 15a - 15h. When the three frequencies or three phases are obtained from a direct digital synthesizer, we see three levels in Figs. 15a - 15h. We will assume that frequencies of a 0.52 bit rate is 1, 0.48 of a bit rate is 0, and 0.50 of a bit rate is the center frequency corresponding to the time period 13/26. Fig. 15g is the data clock. Fig. 15h shows the three frequencies or phases as they are output from the direct digital synthesizer. This can also be the voltage levels used if the modulator is to be a phase or frequency modulator. The upward level represents a one, the negative level a 0. The center line is represents a "hold the last bit". The waveform at Fig. 15h is seen at the phase modulator input and is also the detected output of a phase detector when no bandpass filter is used.

An alternating 10101. . . pattern is shown in Figs. 15a and 15b. A CC hexadecimal pattern, 11001100 . . . is shown in Figs. 15c and 15d, and a

pattern where there are more than two bits of the same polarity in sequence is shown in Figs. 15e and 15f. The data clock is shown in Fig. 15g. The same detector circuitry used for the two frequency or phase method is applicable, but the baseband frequency is cut in half. There are no half bit width changes as shown in Fig. 8. Using a peak detector as shown in Fig. 13, the decoded data pattern would be 1100110011-- or a CC hexadecimal as in Fig. 8. The bits repeat due to the delay between ones and zeros. Alternating ones and zeros without a gap (1010101) result in the pattern shown in Fig. 15a. A long string of ones is shown in Fig. 15e. The R/S flip flop is set by the first one, and will remain at the one position until a negative peak is received.

Fig. 15c shows the signal of Fig. 15h after it has been changed shape by a bandpass filter. A square wave input pulse passing through a narrow bandpass filter will change to a $(\sin x)/x$ pulse which closely resembles a half sine wave. The rectangular wave of Fig. 15h has changed into the triangular wave of Fig. 15c. If the signal of Fig. 15c is peak detected, a CC hexadecimal pattern is detected as shown in Fig. 15d. As shown in Fig. 15c it takes two phase changes (2θ) to go from 1 to 0 or 0 to -1. If the data bit is 1, then the phase is high (θ). If a 1 is again repeated, then according to the encoding of the invention, the phase goes to the rest point 200. When the data then changes to 0, the phase goes low ($-\theta$). If there is a repeated bit, only one phase change need be done, a second repeated bit requires no phase change. A subsequent change then adds the second change of θ degrees. For a 101 or immediate 10 or 01 bit change, the rate is doubled, 2θ per bit change. The phase change $\Delta\theta$ can be equated to frequency.

Fig. 15a shows the integrated phase angle in sloping lines, $((\sin x)/x)^2$ pattern, which is then differentiated or peak detected to obtain the decoded output shown in Fig. 15b. After peak detection the output is a 1 followed by repeated 1's.

Similarly, in the data pattern of Fig. 15c the integrated phase angle in dotted outline, which is then differentiated to obtain the decoded output shown in Fig. 15d. Fig. 15e shows the data pattern according to the invention where a string of 1's are encoded. The integrated phase angle is shown in dotted outline, which is then differentiated to obtain the decoded output shown in Fig. 15f. In all cases the data is shifted one bit width.

Utilizing three phase changes is a phase modulation method. When utilizing three time periods and then integrating, the same result is obtained (integrating PM result in FM). When detecting phase modulation, the triangular waveform of Fig. 8 results. When detecting frequency modulation the rectangular patterns shown in Figs. 1 and 10 result.

The direct digital synthesizer steps in phase angle or frequency, (PSK or FSK) depending on manufacturer and program settings. For example, in Figs. 15a - 15 g the phase angles $-\theta$, 0, and $+\theta$, and in Fig. 8 the upper peak, middle peak, and lower peak each have a different phase or frequency.

Fig. 16 is a block diagram of an encoder circuit capable of programming a direct digital synthesizer to utilize three frequencies. D flip-flops 204 and 206 compare the incoming data in sequence. The data signal is coupled to the D input 208 of flip-flop 204 which has its Q output 210 coupled to the D input of flip-flop 206. The output 212 of flip-flop 206 is the last bit of the data, and the output of flip-flop 204 is the present bit of the data. Both are coupled to the inputs of XOR gate 214. The output 218 XOR gate 214 is coupled to the control input of a direct digital synthesizer shown in Fig. 6. When two consecutive data bits are the same, XOR gate 214 with inverted output 216 is high and the frequency 2 is generated by the direct digital synthesizer.

The last bit on line 212 and current bit on line 210 are also coupled to the inputs of AND gates 220 and 222 respectively. The inverted value of the last bit on line 212 and current bit on line 210 are also coupled to the inputs of

AND gates 222 and 220 respectively. If two consecutive data bits are different, the two AND gates 220 and 222, each having a different input inverted are high either frequency 1 or frequency 3 is coupled from outputs 224 or 226, depending on the state of flip-flop 206, or the value of the last bit.

- 5 The three frequency or phase method utilizes the two frequency algorithm of Fig. 1 divided by two with the third frequency being a midpoint, which is that of an intermediate frequency, indicating there is no change from the last bit.

Fig. 9 shows a prior art VPSK encoding method referred to as "Slip Code." The present method has no slippage. In Fig. 9, the midpoint 228
10 utilizing the present method or bit edge is indicated by the 7 count. A change to 6 or to 8, followed by a return 230 to 7 prevents long term slippage. This is the pattern that would result from three frequencies being used with a reinserted carrier and VPSK encoding.

In Fig. 1 the modulating frequency is equal to the bit rate. In Fig. 9 it is
15 $\frac{1}{2}$ the bit rate. This can result in a theoretical improvement in the signal to noise ratio. The transmission frequency for the waveform shown in Fig. 1 without this return to a midpoint is equal to the phantom carrier plus the bit rate f_b . With the midpoint return obtained by dividing the transmission frequency by two, $f_b/2$. For certain applications, such as FM subcarriers and baseband
20 transmission over power lines, there is an upper frequency limit obtainable for hardware or regulatory reasons. For high data rates the frequency cutoff of the available semiconductors may be too low. The data transmission rate can be doubled utilizing the midpoint return method to overcome these limitations.

The aperture coded waveform centers around the data clock rate f_b
25 located above a phantom carrier, thus at a frequency $f_c + f_b$ leaving the direct digital synthesizer as it creates the sideband alone. This is similar to the "biphase" spectrum. A more desirable spectrum would resemble the "bipolar" spectrum which centers around $f_b/2 = f_m$. This can be accomplished by dividing

the aperture code by 2 as in Fig. 10 and as shown graphically in Figs. 15a - 15g by using a direct digital synthesizer to create the sideband only and by shifting the phase plus or minus an amount $\Delta\theta$ from a center position to obtain ones and zeros. The bipolar modulation pattern results which is shown in 5 Figs. 15a - 15g. This is equivalent to the phase shift of Fig. 1 with a rest position at the center of the bit interval. Fig. 16 shows a three frequency encoder for use with the direct digital synthesizer of Fig. 6. Only one of the three outputs is high at any one time.

Frequency or phase modulation when passed through a narrow band 10 filter appears to integrate the waveform when detected by normal phase detector which results in the waveform shown in Fig. 8. The square wave input from the waveform in Fig. 1 appears as an integrated triangular waveform 146 in Fig. 8 as is apparent from the sloping lines. In Fig. 15b the ones and zeros spread the gap between them. The integrated waveform 146 15 after passing through a narrow band filter is shown by the dotted line.

One of the detection methods used for MSK and MFSK modulation involves the use of separate crystal filters for each frequency transmitted and a correlator or sample and hold detector to indicate which frequency was received. This same detection method is applicable to the present invention, 20 although with greater intersymbol interference. Such detectors are discussed in various texts including Wireless Communications by T.S. Rappaport, (Prentice Hall) and Principles of Communications Systems, by Taub and Schilling, (McGraw Hill).

The modulation index can be determined in terms of frequencies from 25 the relationship:

$$\beta = 2 \Delta f / f_b = \Delta f / f_m$$

Note that $f_m = f_b/2$ or one half the bit rate. To maintain a constant phase shift for comparison of the aperture code with the divide-by-two code, 2

$\Delta f / f_b$ and $\Delta f / f_m$ must be equal. Since $f_b/2 = f_m$, Δf for the return zero phase (bipolar) method, obtained by dividing the first method by 2, is twice as bandwidth efficient as the aperture coded method in Fig. 1. The data rate can be doubled in the same bandwidth, or the filter can be made narrower to
5 improve the signal-to-noise ratio by 2/1.

This method is particularly applicable to data modulation where the highest allowable or obtainable data rate can be doubled over that obtainable from the aperture coded method. For example, the direct digital synthesizer may have an upper frequency limit of 20 MHz when creating the single
10 sideband signal. The data rate can be doubled at that frequency by using the divide-by-two method of Fig. 10 and Figs. 15a - 15g.

The direct generation of the sideband without ever having used a carrier can also be used with the three time period, hence three frequency, prior art VPSK coding method, now referred to as "Slip Coding" referred to above with
15 a different encode/decode algorithm. In the "Slip Coding" method of Fig. 9 there are three possible zero crossings for a bit that has been stretched according to the data pattern. The rules are if the present bit is a repeat of the last bit, the next zero crossing will occur at 6/6 bit width from the last bit for a 6,7,8 code. If there is a change from the last bit from a one to a zero or a zero
20 to a one, then the next zero crossing is delayed 1/6 to 7/6 bit width. If the change count is 2 less than the increment and there is a 101 in the data pipeline, the stretch is to 8/6 bit width. The decoder recognizes this extra stretch and resets the counters, outputs a missed zero plus the next bit, which is always a one, and resets the phase ambiguity. This coding method is
25 described in the issued patents referred to above. Other codes are possible, for example 8,9,10 or 10,11,12.

These three time periods can also be converted to frequencies which can be generated by a digital synthesizer or numerically controlled oscillator in

the same manner as the aperture codes described. The resulting signal is a single sideband signal which has been generated without a carrier ever having been used. It can be detected by means of a phantom carrier utilizing the method described by Stryzak and Walker, or by utilizing the technique
 5 commonly used for MFSK modulation which would employ three narrow band crystal filters and correlation circuitry.

One advantage of the direct sideband modulation method of the present invention is that it requires minimal filtering to meet FCC Regulations. The principal advantage is that no carrier is needed. This can be of great
 10 advantage when used with satellites where there is a frequency offset between up and down links, or when Doppler effects are encountered. The only requirement of this method is that the detected signal fit within the linear filter bandpass.

The coding and modulating method described in this invention can be
 15 analyzed using the formulas developed by R. Best ("Phase Locked Loops", McGraw Hill, 1984, pp 57-59).

For aperture coding:

$$Q = f_b / (2 \Delta f)$$

$$R = f_b / (2 \Delta f)$$

20 For divide-by-two coding:

$$R = f_b / (4 \Delta f)$$

$$SNR = \beta^2 (P_{in}/P_{out}) (Nyquist\ bandwidth/Noise\ bandwidth) E_b/n = \beta^2 Q R E_b/n$$

25 where $\beta^2 = [\pi (\Delta f / f_m)]^2 = [\pi (\Delta f / f_b)]^2$, where E_b is the bit energy in Joules, e.g. signal power/bits/sec, n is the number of symbols used per bit, and where P_{in}/P_{out} is the ratio of the noise bandwidth transmitted full spectrum divided by the narrow bandwidth as actually transmitted and

received. It is also equal to the (Sample Rate)/(Filter bandwidth). Bandwidth efficiency in bits/second per Hz = Q.

The Nyquist bandwidth/Noise bandwidth is a phase noise improvement factor which is independent of the bandwidth efficiency and which is given the designation R. The present invention (VMSK) and the prior art (VPSK) are heavily dependent upon this factor, the biphase encoding methods having been devised to take advantage of it. This factor, R, does not apply to the other commonly used modulation methods such as multiple phase shift keying (MPSK) and quadrature amplitude modulation (QAM), which are NRZ line code methods, not "Biphase" methods.

The values for the formula using aperture coding are:

$$\text{Signal to Noise Ratio} = 2.44 E_b/n$$

The bandwidth efficiency is obtained without using multiple bits per symbol (n = 1 bit per symbol). The data rate and sample rate are the same. When divided by 2 the signal-to-noise ratio is:

$$\text{Signal to Noise Ratio} = 4.9 E_b/n$$

Which is an improvement in signal-to-noise ratio of 2/1.

The Signal to noise ratio is independent of the bandwidth efficiency as indicated above. This is not true of non-biphase encoded modulation methods. It is also to be noted that this ratio is higher than that usually obtained from any other modulation method with the exception of orthogonal MFSK.

Shannon's Limit might be interpreted by some to appear to be very high for this modulation method, being nearly 90 dB for a Q = 26 bits/sec./Hz

bandwidth efficiency. This is a misinterpretation. One bit per symbol is used in the present invention, not Q bits per symbol. The data rate equals the sampling rate and the correct Shannon's Limit is 0 dB as is proven by measurement.

5 Many alterations and modifications may be made by those having ordinary skill in the art without departing from the spirit and scope of the invention. Therefore, it must be understood that the illustrated embodiment has been set forth only for the purposes of example and that it should not be taken as limiting the invention as defined by the following claims.

10 The words used in this specification to describe the invention and its various embodiments are to be understood not only in the sense of their commonly defined meanings, but to include by special definition in this specification structure, material or acts beyond the scope of the commonly defined meanings. Thus if an element can be understood in the context of this
15 specification as including more than one meaning, then its use in a claim must be understood as being generic to all possible meanings supported by the specification and by the word itself.

 The definitions of the words or elements of the following claims are, therefore, defined in this specification to include not only the combination of
20 elements which are literally set forth, but all equivalent structure, material or acts for performing substantially the same function in substantially the same way to obtain substantially the same result. In this sense it is therefore contemplated that an equivalent substitution of two or more elements may be made for any one of the elements in the claims below or that a single element
25 may be substituted for two or more elements in a claim.

 Insubstantial changes from the claimed subject matter as viewed by a person with ordinary skill in the art, now known or later devised, are expressly contemplated as being equivalently within the scope of the claims. Therefore,

obvious substitutions now or later known to one with ordinary skill in the art are defined to be within the scope of the defined elements.

The claims are thus to be understood to include what is specifically illustrated and described above, what is conceptionally equivalent, what can
5 be obviously substituted and also what essentially incorporates the essential idea of the invention.

1 I claim:

1 1. An encoding method for a digital baseband signal comprising:
2 dividing a data bit interval having a bit center into at least two time
3 apertures, each time aperture in turn being divided into one or more time
4 segments;
5 selecting a first subset of said plurality of time segments in a first one of
6 said at least two apertures, said first subset of said plurality of time segments
7 ending prior to said bit center;
8 selecting a second subset of said plurality of time segments in a second
9 one of said at least two apertures from said remaining plurality of time
10 segments, which second subset of said plurality of time segments ends after
11 said bit center; and
12 encoding said digital baseband signal to correspond to said first and
13 second subsets of time segments.

1 2. The method of Claim 1 wherein said data bit interval has a starting
2 time and further comprising changing the polarity of said digital baseband
3 signal responsive to the start of said data bit interval, then encoding a polarity
4 reversal associated with said digital baseband signal based upon time duration
5 of said first and second subset of time segments.

1 3. The method of Claim 1 where in selecting said first and second
2 subset of time segments have relative durations representative of digital ones
3 and zeros associated with said digital baseband signal.

1 4. The method of Claim 1 wherein said digital baseband signal has a
2 rising edge and falling edge, and further comprising selecting between a first
3 and second operational mode, said method comprising in said first operational
4 mode said rising edge as a clock timing and leading edge of said data bit
5 interval, and timing of said falling edge as whether said digital baseband signal
6 is a digital one or zero; and said method comprising in said selected second
7 operational mode said falling edge as a clock timing and leading edge of said
8 data bit interval, and timing of said rising edge as whether said digital
9 baseband signal is a digital one or zero.

1 5. The method of Claim 2 wherein a sum of said time durations of said
2 first and second time subsets equals the sum of said time durations of all of
3 said plurality of time segments in said at least two apertures.

1 6. The method of Claim 2 wherein said first and second subset of time
2 segments are converted to first and second frequencies generated by a
3 frequency shift means or a direct digital synthesizer producing a modulation
4 index which is less than approximately 0.1, which alters the nature of the
5 modulated signal from that of a carrier plus two sidebands to that of a narrow
6 biphase single sideband alone.

1 7. The method of Claim 6 wherein said data bit has a rising edge and
2 falling edge, and further comprising selecting between a first and second
3 operational mode, said method comprising in said first operational mode said
4 rising edge as a clock timing and leading edge of said data bit interval, and
5 timing of said falling edge as whether said data bit is a digital one or zero, and
6 said method comprising in said selected second operational mode said falling

7 edge as a clock timing and leading edge of said data bit interval, and timing of
8 said rising edge as whether said data bit is a digital one or zero; and
9 wherein a third frequency, corresponding to a third subset of time
10 segments, is obtained by dividing the data bit interval in said first operational
11 mode by two.

1 8. A digital data transmission method comprising :
2 aperture encoding a digital baseband signal, said aperture coding
3 comprising:
4 dividing a data bit interval having a bit center into a plurality of
5 time segments;
6 selecting a first subset of said plurality of time segments, each
7 ending prior to said bit center; and
8 selecting a second subset of said plurality of time segments
9 comprised of all remaining ones of said plurality of time segments not
10 included in said first subset of time segments, each of which ends after
11 said bit center;
12 phase or frequency shift modulating a transmitted signal using said
13 aperture coded digital data; and
14 transmitting said phase or frequency modulating a transmitted signal.

1 9. The method of Claim 8 further comprising superimposing additional
2 information upon said modulated transmitted signal using low level amplitude
3 modulation.

1 10. The method of Claim 8 wherein transmitting said phase or
2 frequency modulating a transmitted signal comprises transmitting said aperture
3 coded digital data at baseband over wires.

1 11. The method of Claim 8 wherein transmitting said phase or
2 frequency modulating a transmitted signal comprises transmitting said aperture
3 coded digital data at radio frequencies in which said aperture coded digital
4 data is used to modulate a carrier frequency.

1 12. The method of Claim 11 wherein said transmitting said aperture
2 coded digital data comprises transmitting a single sideband signal with a
3 suppressed carrier, wherein said sideband is located at a frequency equal to
4 said suppressed carrier but offset positively or negatively from said carrier by
5 not more than the data rate.

1 13. The method of Claim 11 wherein said transmitting said aperture
2 coded digital data comprises transmitting a single sideband signal with a
3 suppressed carrier, wherein said sideband is located at a frequency equal to
4 said suppressed carrier but offset positively or negatively from said carrier by
5 not more than one half the data rate.

1 14. The method of Claim 12 wherein said suppressed carrier is a
2 multiple of the data rate.

1 15. The method of Claim 12 further comprising passing said single
2 sideband through a very narrow bandpass filter which passes only a center

3 frequency of said sideband plus or minus a first predetermined variance so
4 that a very high bandwidth efficiency and excellent signal-to-noise ratio are
5 obtained.

1 16. The method of Claim 8 further comprising receiving and decoding
2 said phase or frequency modulated transmitted signal with a phase detecting
3 or discriminating circuit which is responsive to phase or frequency shifts
4 respectively, but does not utilize a carrier or other reference signal.

1 17. The method of Claim 8 further comprising receiving and decoding
2 said phase or frequency modulated transmitted signal with a narrow bandwidth
3 crystal filter, and a sample and hold or correlator circuitry.

1 18. The method of Claim 8 further comprising receiving and decoding
2 said phase or frequency modulated transmitted signal by differentiating and a
3 Manchester decoding said transmitted signal.

1 19. The method of Claim 8 further comprising receiving and decoding
2 said phase or frequency modulated transmitted signal by detecting peaks of
3 said transmitted signal as said aperture coded digital data.

1 20. The method of Claim 8 where in transmitting said phase or
2 frequency modulated transmitted signal no coherent carrier frequency is used
3 so that transmission is insensitive to Doppler frequency offsets when
4 transmitting by means of a relay or satellite.

1 21. An apparatus comprising:
2 an encoder which receives a source signal and aperture encodes it;
3 a transmitter coupled to said encoder;
4 a receiver which communicates with said transmitter and detects said
5 aperture coded source signal; and,
6 a decoder coupled to said receiver which decoder decodes said
7 aperture coded source signal;
8 wherein said aperture coding comprises:
9 dividing a data bit, having a bit center, into a plurality of time
10 segments;
11 selecting a first subset of time segments from said plurality of
12 time segments, each of said first subset of time segments ending close
13 to said bit center; and,
14 selecting a second subset of time segments from said remaining
15 plurality of time segments, each of said second subset of time segments
16 starting after said first subset of time segments;
17 wherein the duration of said first and second subsets of time segments
18 depend upon whether said source signal is a digital one or zero.

1 22. An apparatus comprising:
2 an encoder means for aperture coding a source signal;
3 a transmitter means for transmitting said aperture coded source signal;
4 a receiver means for detecting and receiving said aperture coded
5 source signal; and,
6 a decoder means for decoding said aperture coded source signal;
7 wherein said aperture coding comprises:
8 dividing a data bit, having a bit center, into a plurality of time
9 segments;

10 selecting a first subset of time segments from said plurality of
11 time segments, each of said first subset of time segments ending close
12 to said bit center; and,
13 selecting a second subset of time segments from said remaining
14 plurality of time segments, each of said first subset of time segments
15 starting after said first subset of time segments;
16 wherein the duration of said first and second subsets of time segments
17 depend upon whether said source signal is a digital one or zero.

1 23. An encoding method for generating a digital baseband signal
2 derived from a multiplicity of input data bits, wherein each input data bit is
3 characterized by a temporal data bit length, a temporal bit center and a
4 temporal state indicative of a logical state, said encoding method comprising:
5 receiving one of said multiplicity of input data bits; and
6 generating an encoded digital baseband signal characterized by a
7 temporal state indicative of a logical state, which encoded digital baseband
8 signal changes its temporal state according to whether said one input data bit
9 changes its temporal state at a time prior to or after said temporal bit center.

1 24. The method of Claim 23 wherein said encoded digital baseband
2 signal has two polarities, wherein said one input data bit has a starting time,
3 and wherein generating said encoded digital baseband signal comprises
4 changing said polarity of said encoded digital baseband signal responsive to
5 said starting time of said one input data bit, then changing back said polarity of
6 said digital baseband signal when said one input data bit changes state.

1 25. The method of Claim 23 wherein said encoded digital baseband
2 signal has a first state during a first time period with a first duration and a
3 second state during a second time period with a second duration, wherein
4 generating said encoded digital baseband signal generates said first and
5 second durations to correspond to digital ones and zeros of said one input
6 data bit.

1 26. The method of Claim 23 wherein said encoded digital baseband
2 signal has a rising, leading edge of said digital baseband signal and a falling,
3 trailing edge, and wherein generating said encoded digital baseband signal
4 generates a clock frequency based on said rising, leading edge, and
5 generates said falling, trailing edge at a time dependent upon whether said
6 one input data bit is a digital one or zero.

1 27. The method of Claim 23, wherein said encoded digital baseband
2 signal has a rising, falling edge of said digital baseband signal and a falling,
3 leading edge, and wherein generating said encoded digital baseband signal
4 generates a clock frequency based on said falling, leading edge, and
5 generates said rising, trailing edge at a time dependent upon whether said one
6 input data bit is a digital one or zero.

1 28. The method of Claim 23 wherein generating said encoded digital
2 baseband signal generates said states of said encoded digital baseband signal
3 as two frequencies to produce a modulation index less than approximately 0.1,
4 which alters the nature of said encoded digital baseband signal from that of a
5 carrier plus two sidebands to that of a narrow biphase single sideband alone.
6

1 29. The method of Claim 28 where generating said encoded digital
2 baseband signal generates said encoded digital baseband signal with a
3 temporal duration and further comprises:

4 extending said temporal duration of said encoded digital baseband
5 signal;

6 generating a first and second frequency corresponding to said falling
7 edge depending on whether said one input data bit is a digital one or zero
8 respectively; and

9 generating a third frequency corresponding to a repeat of the digital
10 state of said one input data bit in the next sequential input data bit of said
11 multiplicity of input data bits.

1 30. A method for the transmission of digital data and additional
2 information comprising:

3 encoding digital data using an aperture coding;

4 transmitting the aperture coded digital data utilizing phase or frequency
5 shift modulation; and,

6 receiving and decoding the phase or frequency shift modulated
7 transmitted aperture coded digital data.

1 31. The method of Claim 30 further comprising superimposing said
2 additional information upon said encoded digital data using low level amplitude
3 modulation.

1 32. The method of Claim 30 wherein said multiplicity of input data bits
2 has a data rate and where encoding digital data using said aperture coding
3 comprises:

4 dividing a data bit interval, having a bit center, in time into a plurality of
5 apertures;

6 selecting a first subset of apertures from said plurality of apertures,
7 each of which apertures of said first subset end close to said bit center; and,

8 selecting a second subset of apertures comprised of all of said
9 remaining plurality of apertures not included in said first subset, each of which
10 apertures of said second subset start after said first subset of apertures; and

11 wherein said aperture coding results in a narrow Fourier spectrum
12 containing no low frequency components and is offset from 0 Hz by a quantity
13 equal to said data rate.

1 33. The method of Claim 30 wherein said encoded digital data is
2 transmitted over wires.

1 34. The method of Claim 30 wherein said encoded digital data is
2 transmitted at radio frequencies and said aperture coding is utilized to
3 modulate a carrier frequency within said at radio frequencies.

1 35. An apparatus for use with source signals comprising:
2 means for aperture coding said source signals;
3 means for transmitting said aperture coded source signals;

4 means for receiving and detecting said aperture coded source signals;

5 and

6 means for decoding said aperture coded source signals;

7 wherein said aperture coding comprises:

8 dividing a data bit interval, having a bit center, in time into a

9 plurality of apertures;

10 selecting a first subset of apertures from said plurality of

11 apertures, each of which apertures of said first subset end close to said

12 bit center; and,

13 selecting a second subset of apertures comprised of all of said

14 remaining plurality of apertures not included in said first subset, each of

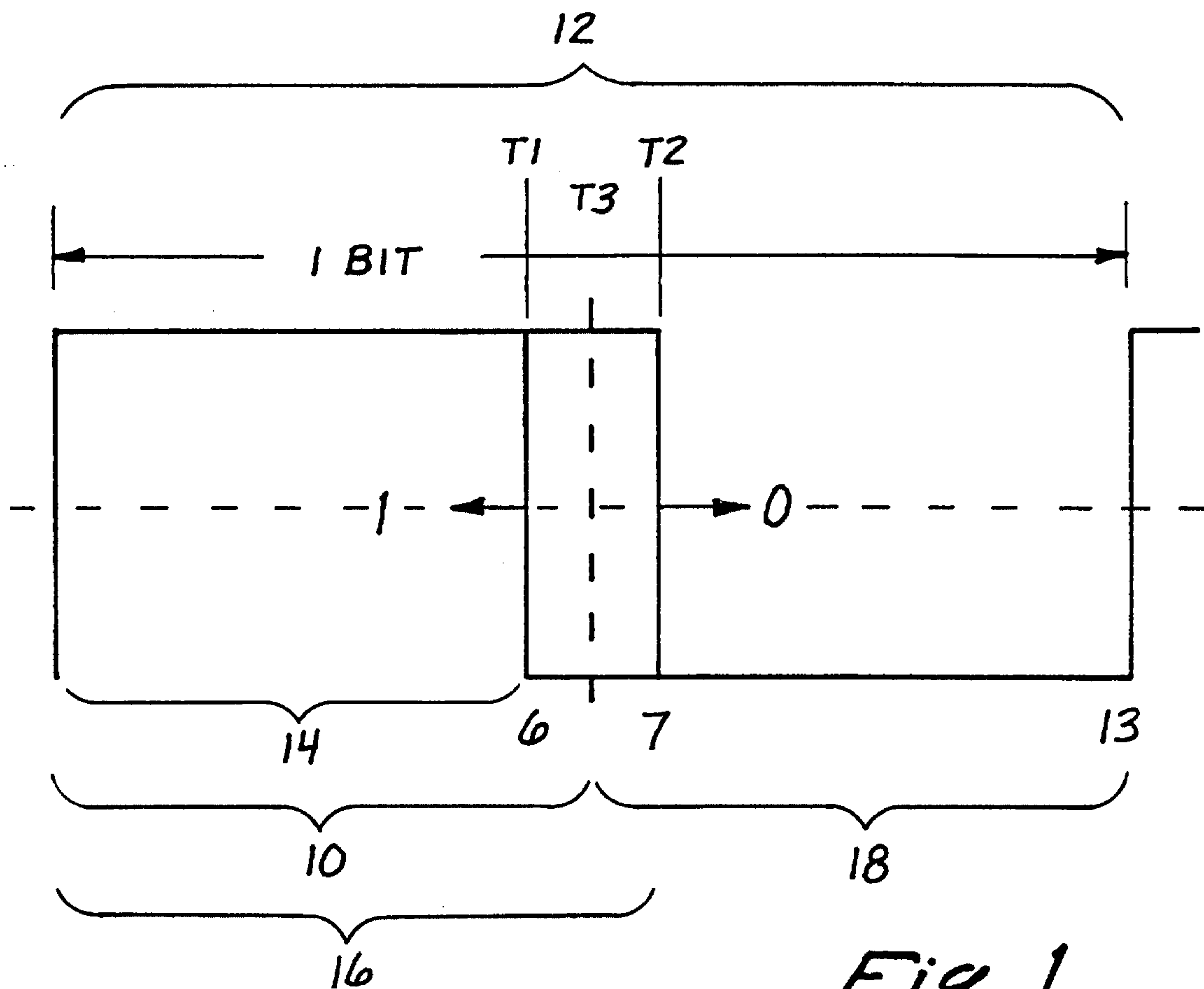
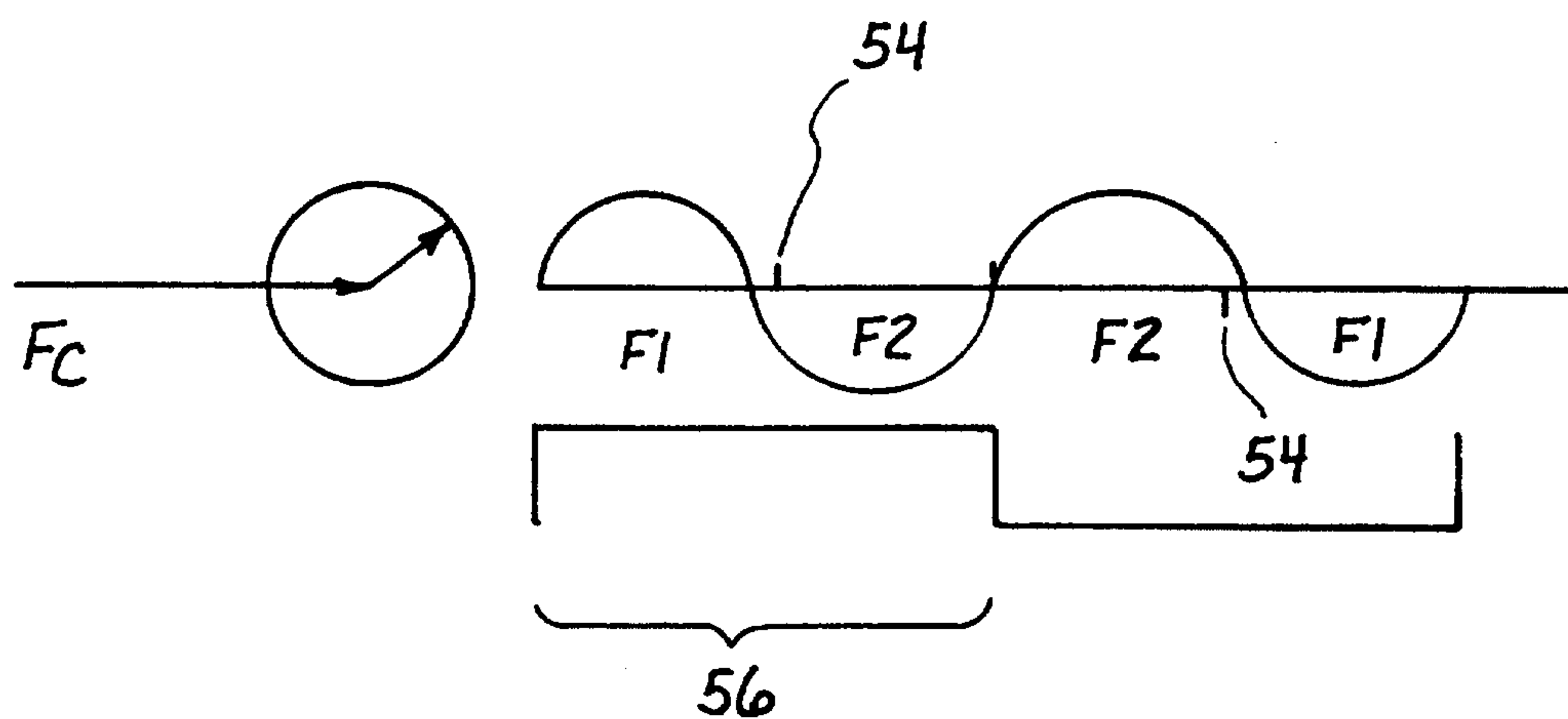
15 which apertures of said second subset start after said first subset of

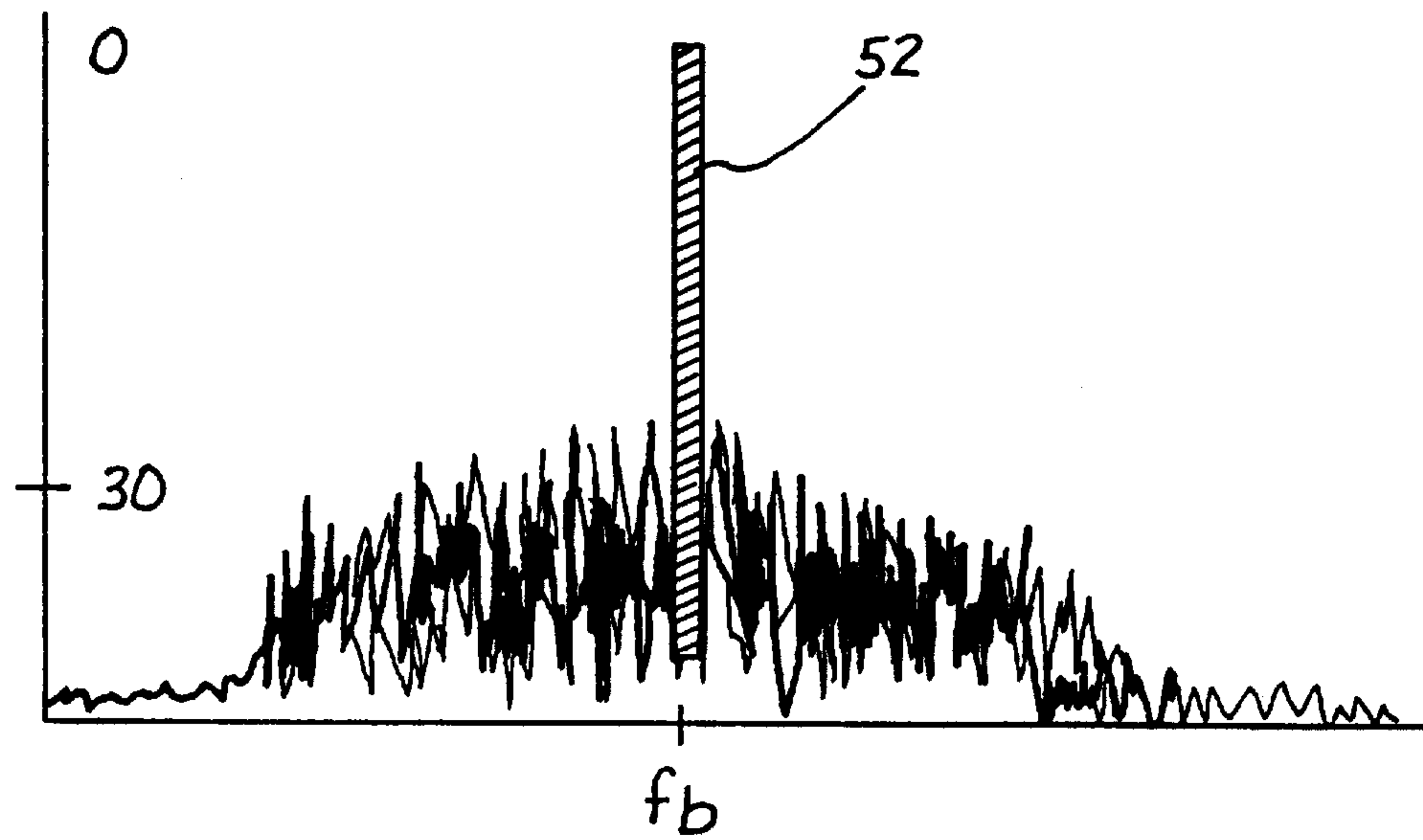
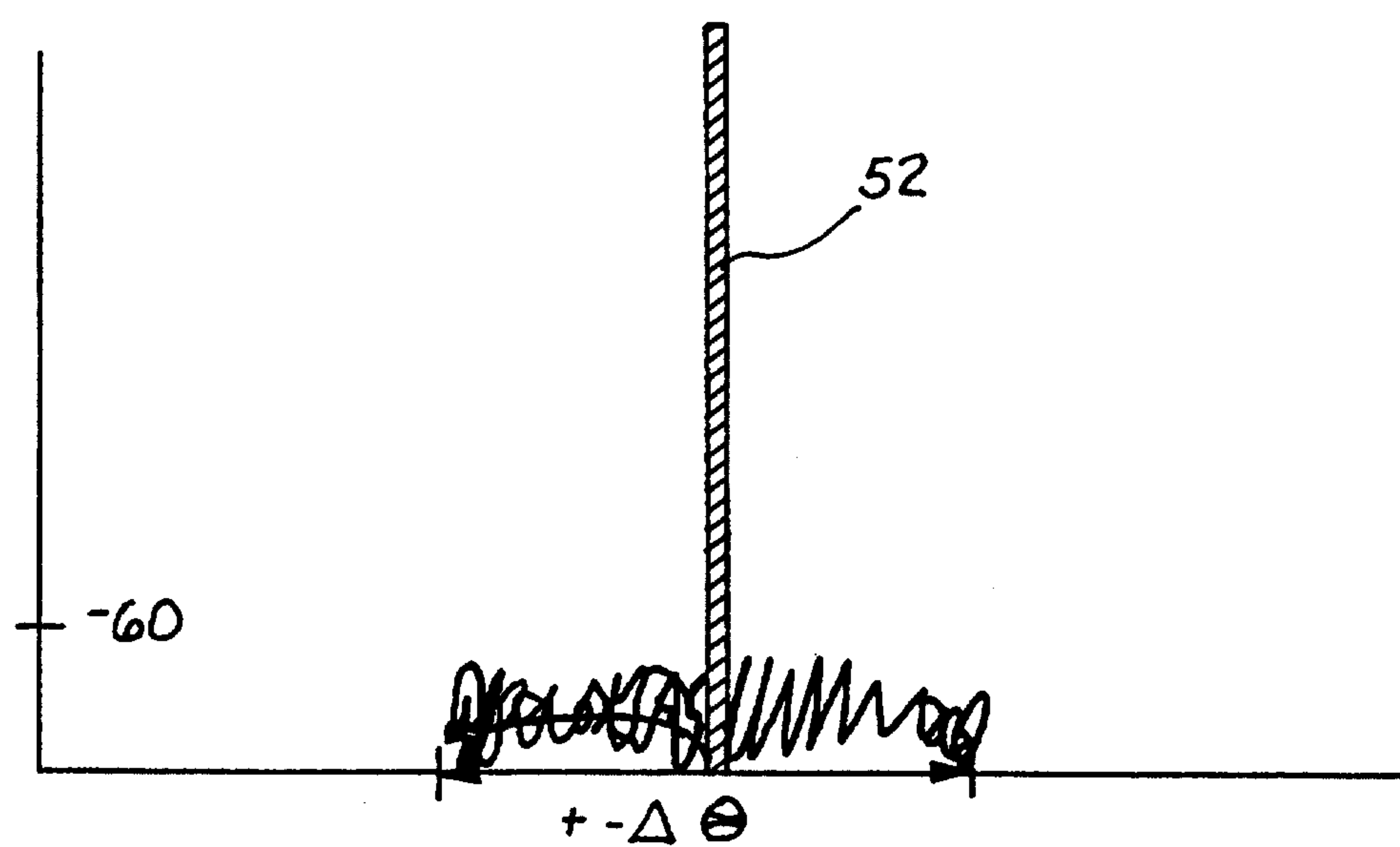
16 apertures; and

17 wherein each of said apertures of said first and second subsets of

18 apertures has a duration which depends on whether a corresponding source

19 signal is a digital one or zero.

*Fig. 1**Fig. 4*

*Fig. 2a**Fig. 2b*

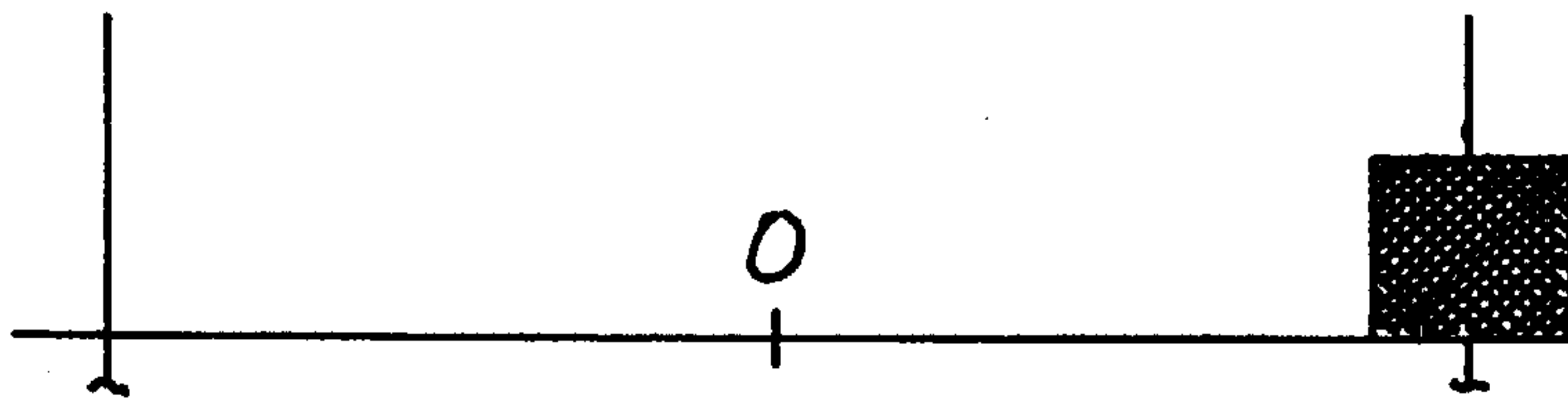


Fig. 3a
PRIOR ART

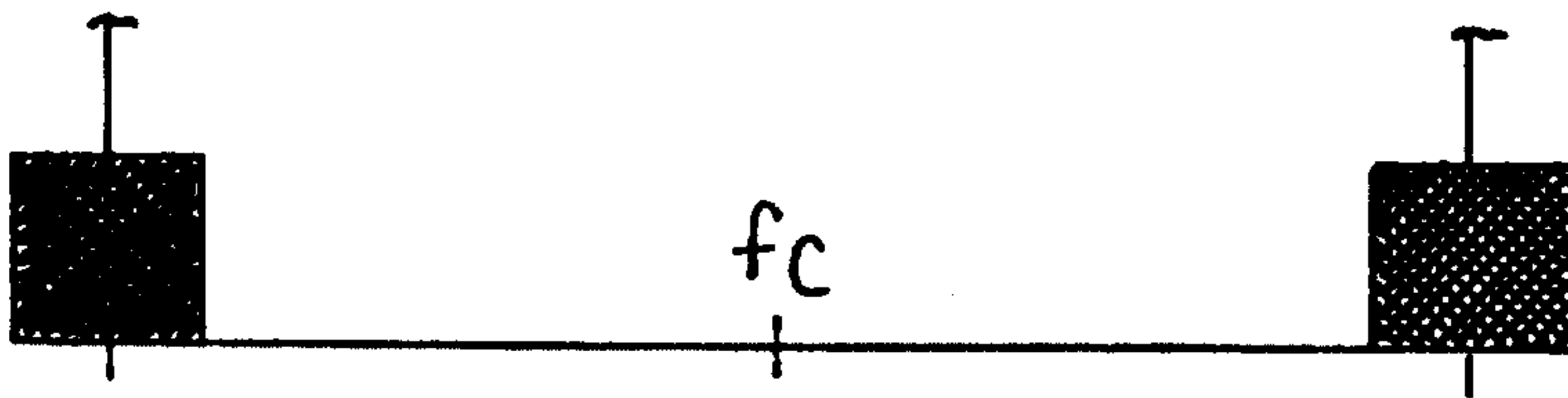


Fig. 3b
PRIOR ART

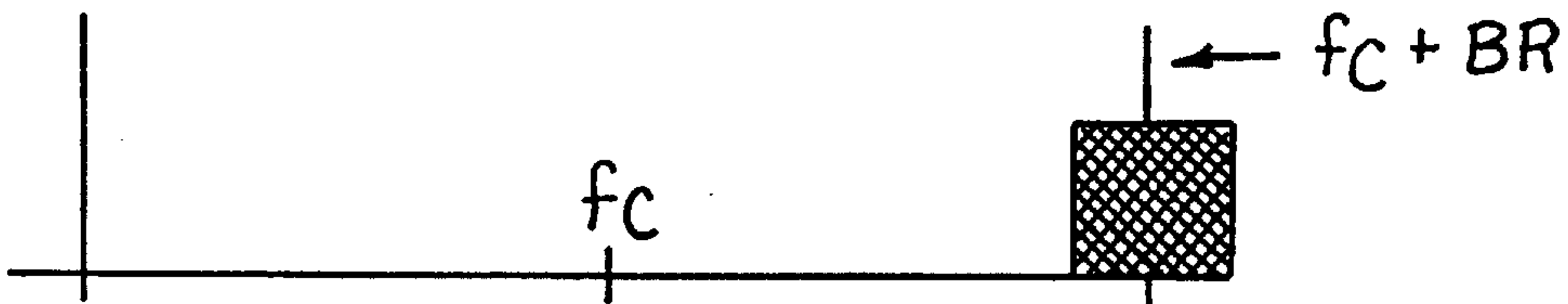


Fig. 3c
PRIOR ART

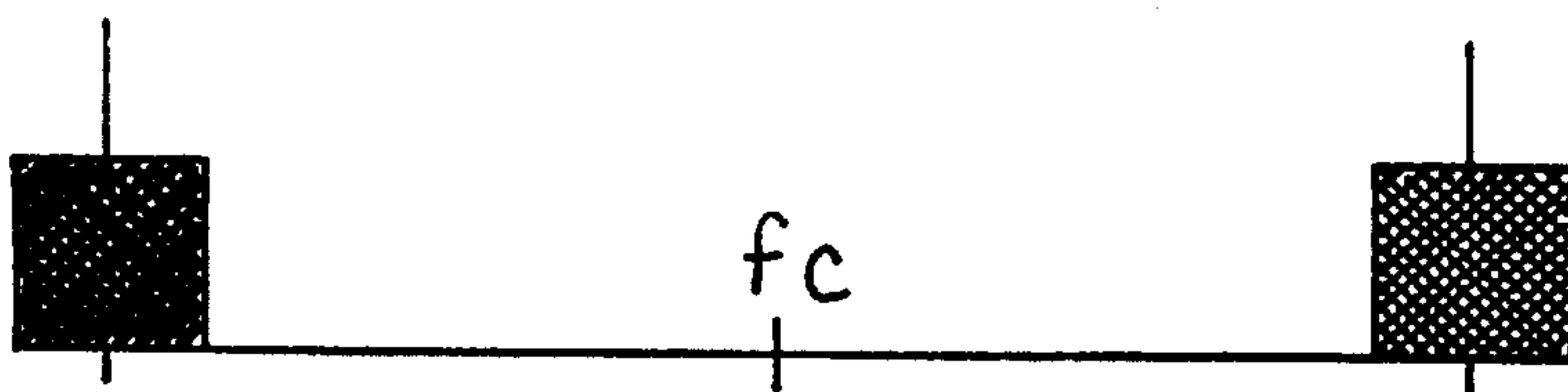


Fig. 3d
PRIOR ART

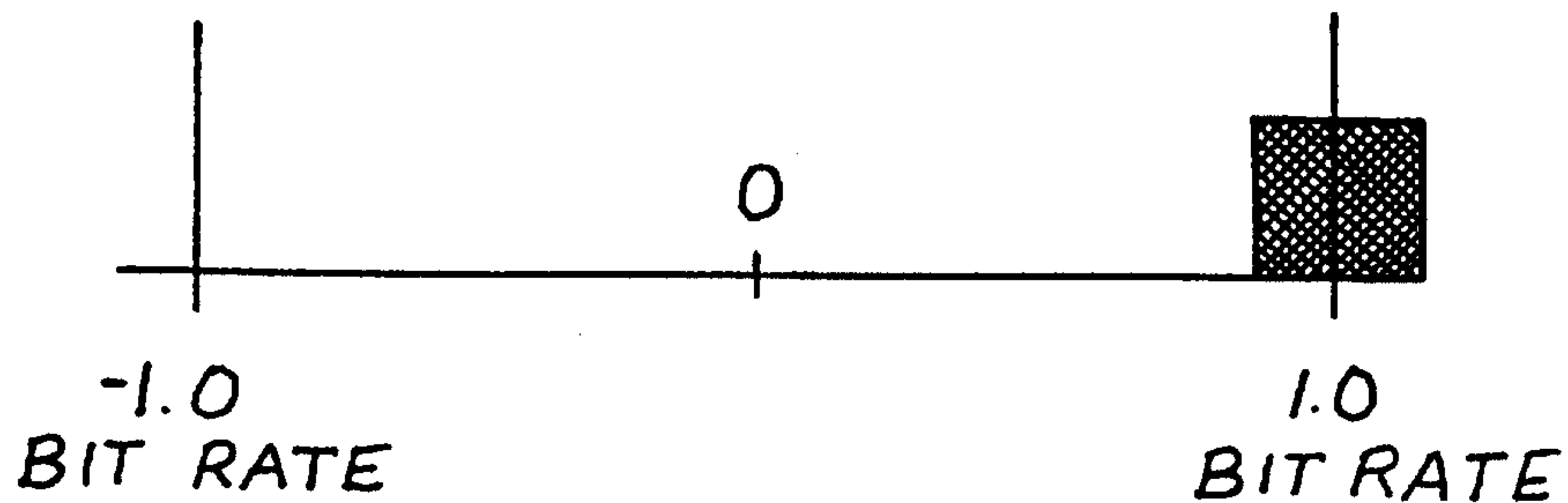
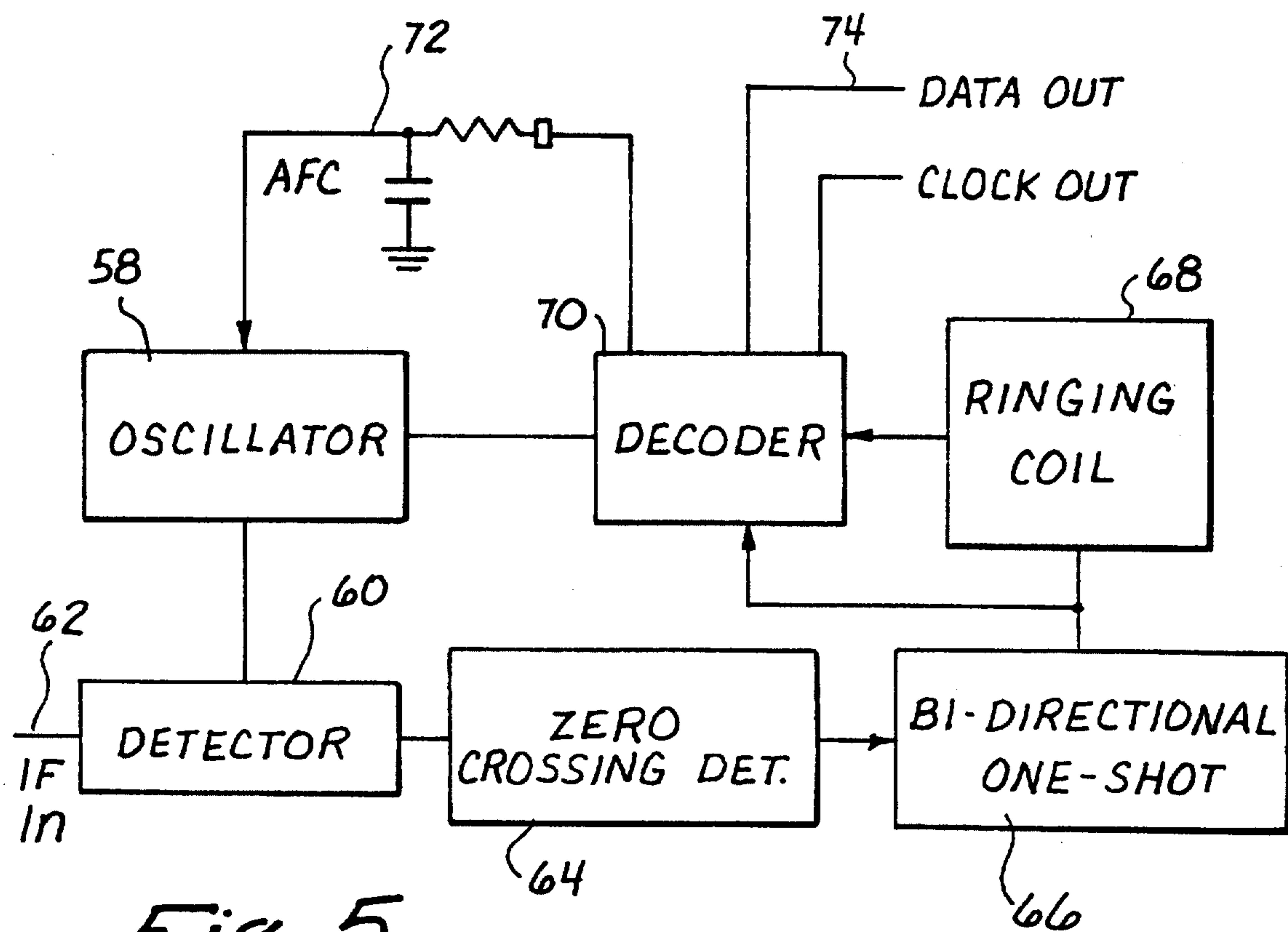
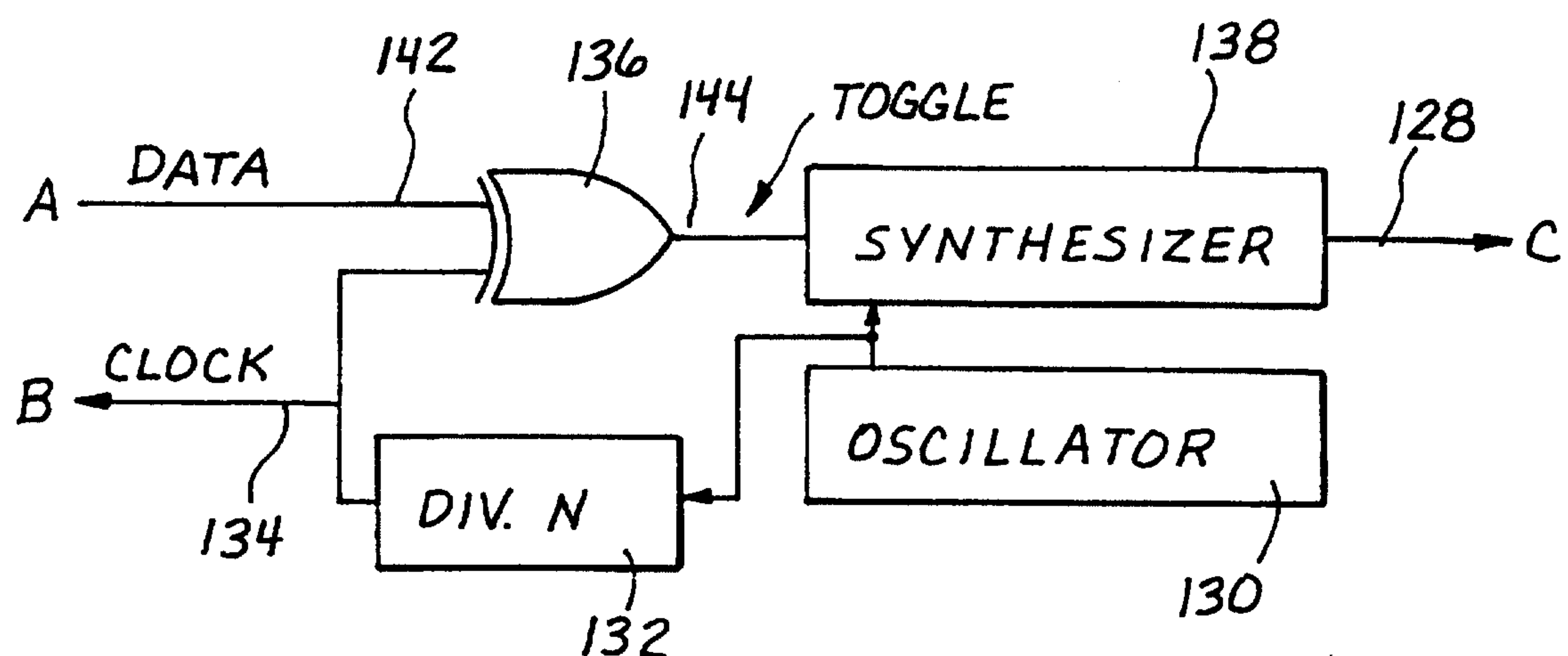
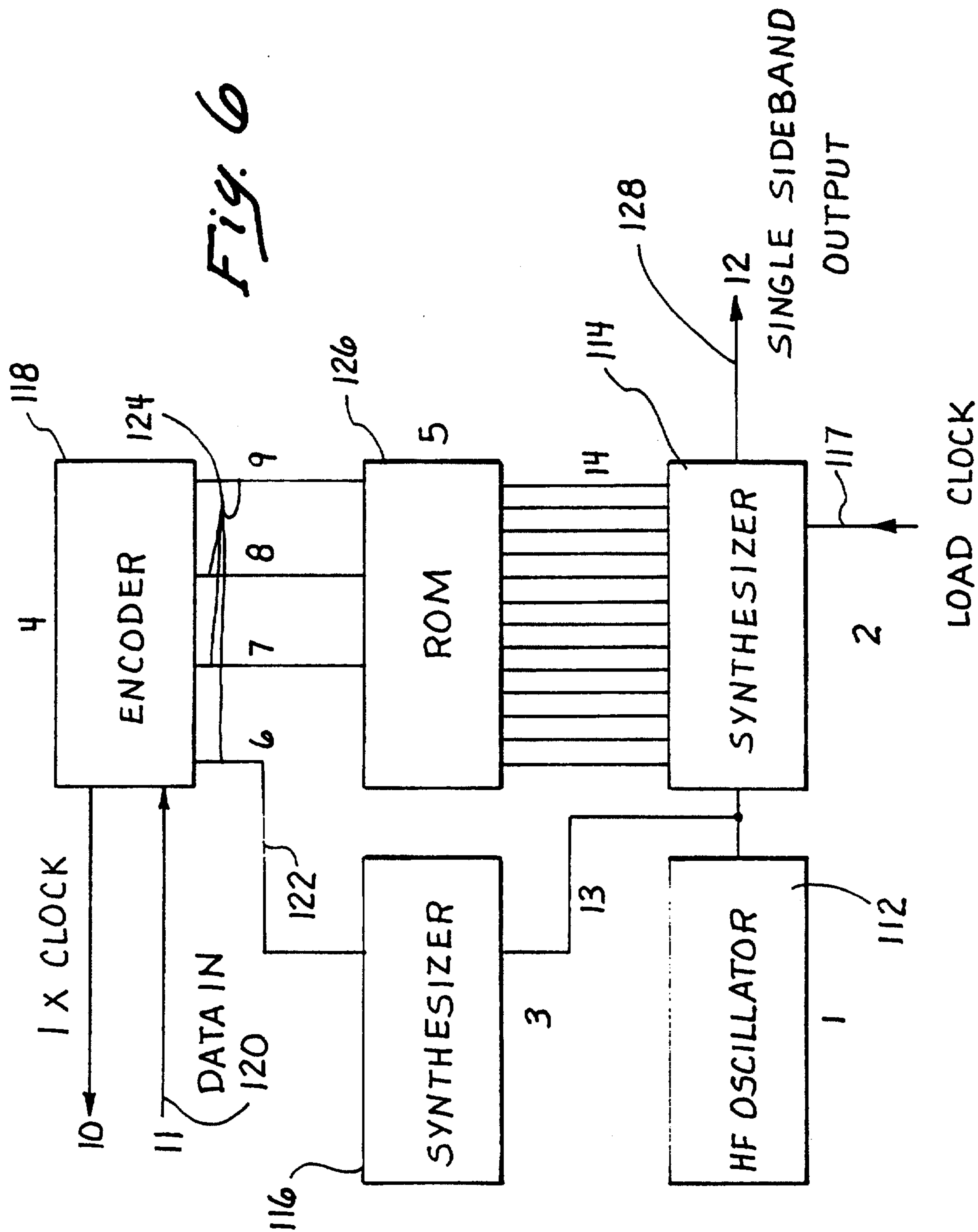
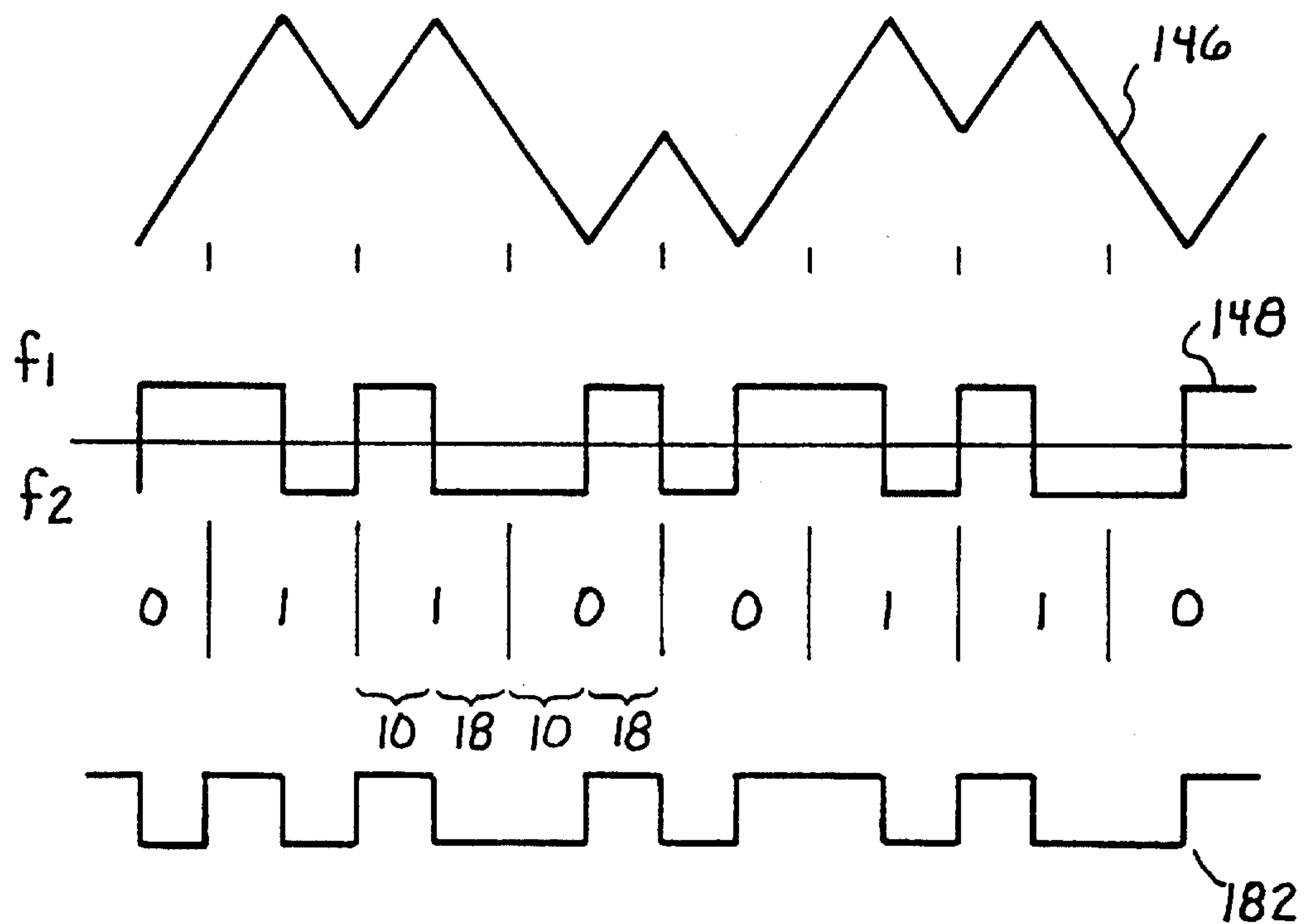
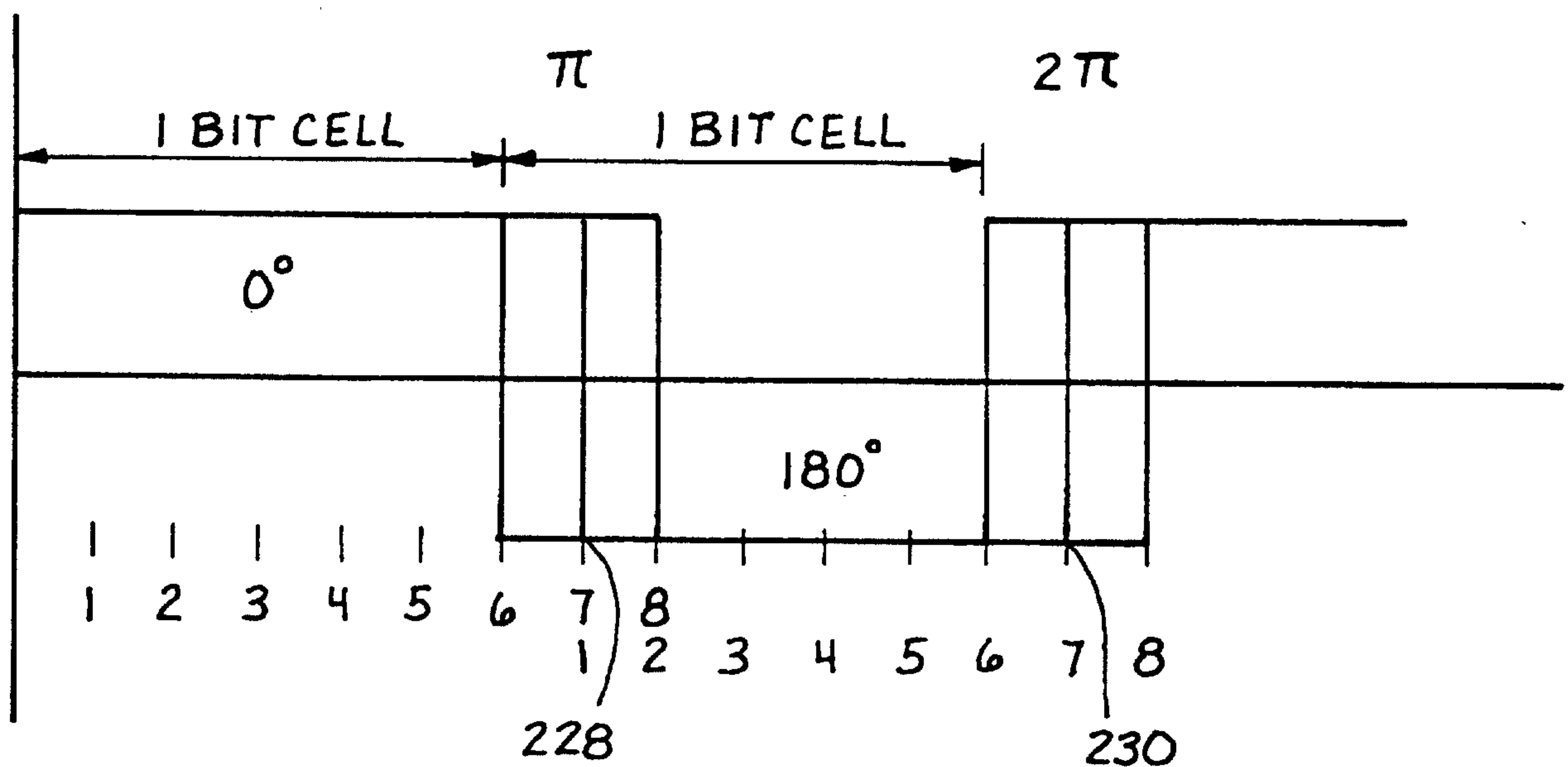


Fig. 3e

*Fig. 5**Fig. 7*



*Fig. 8**Fig. 9*
PRIOR ART

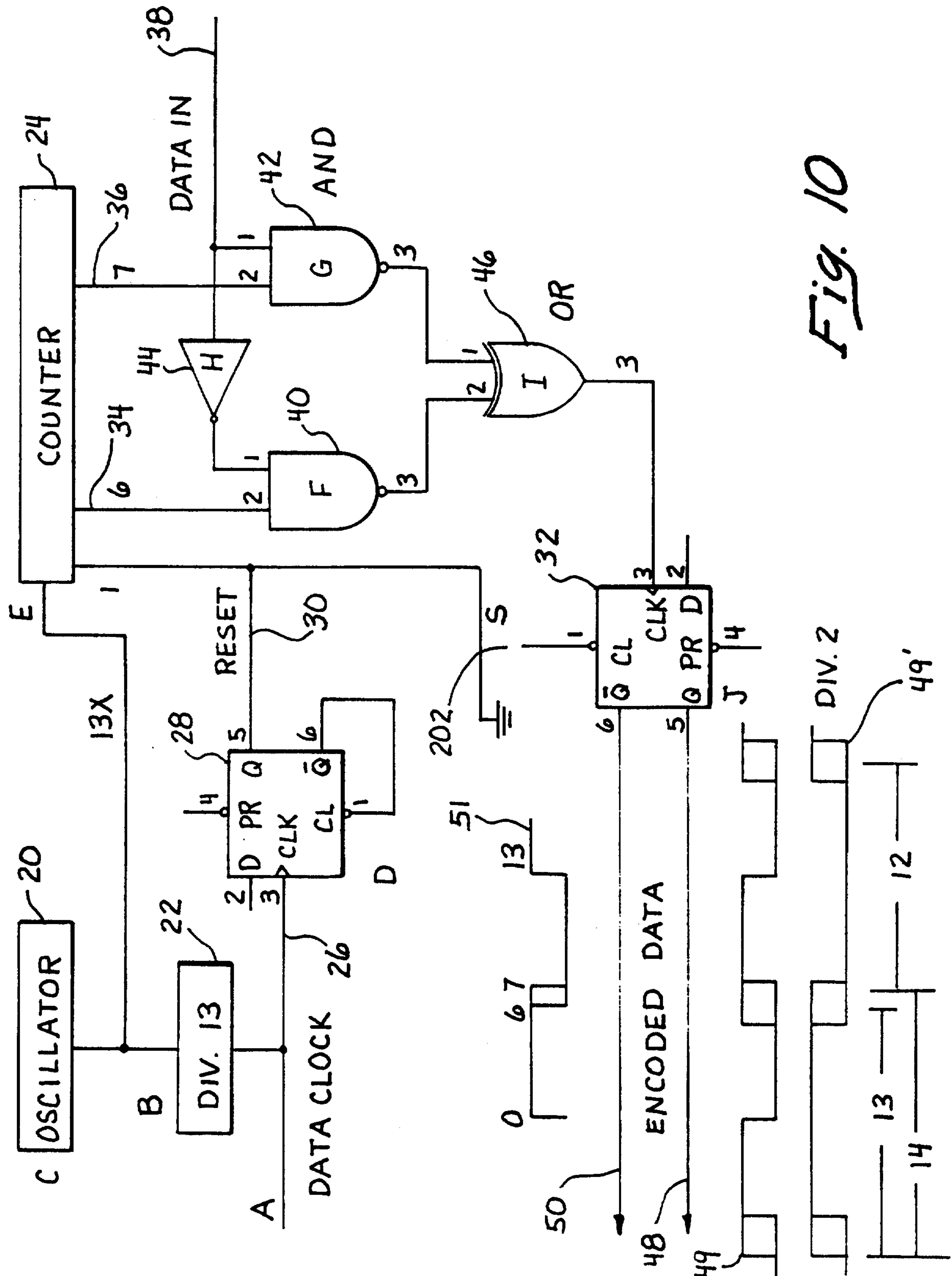


Fig. 10

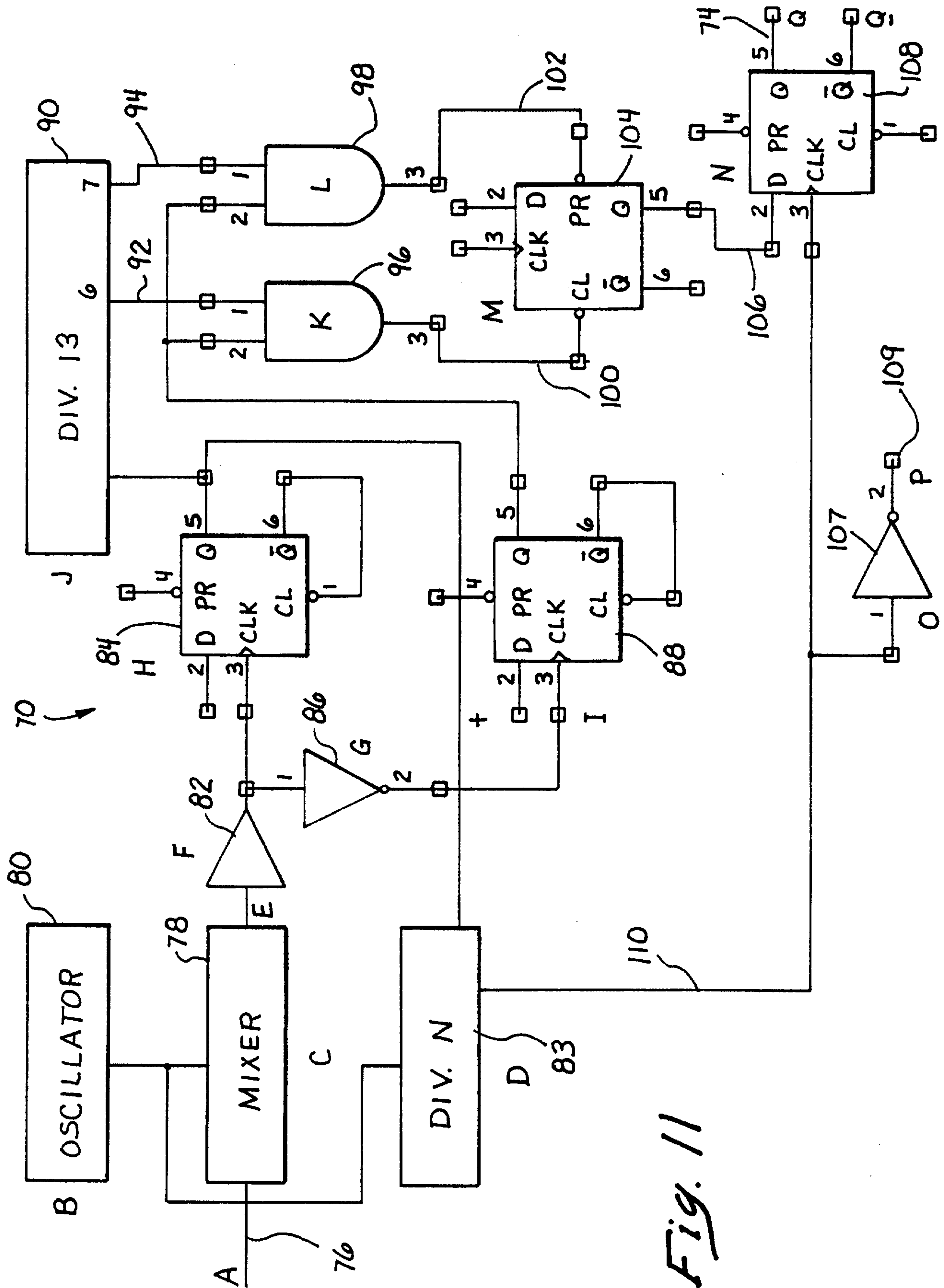


Fig. 11

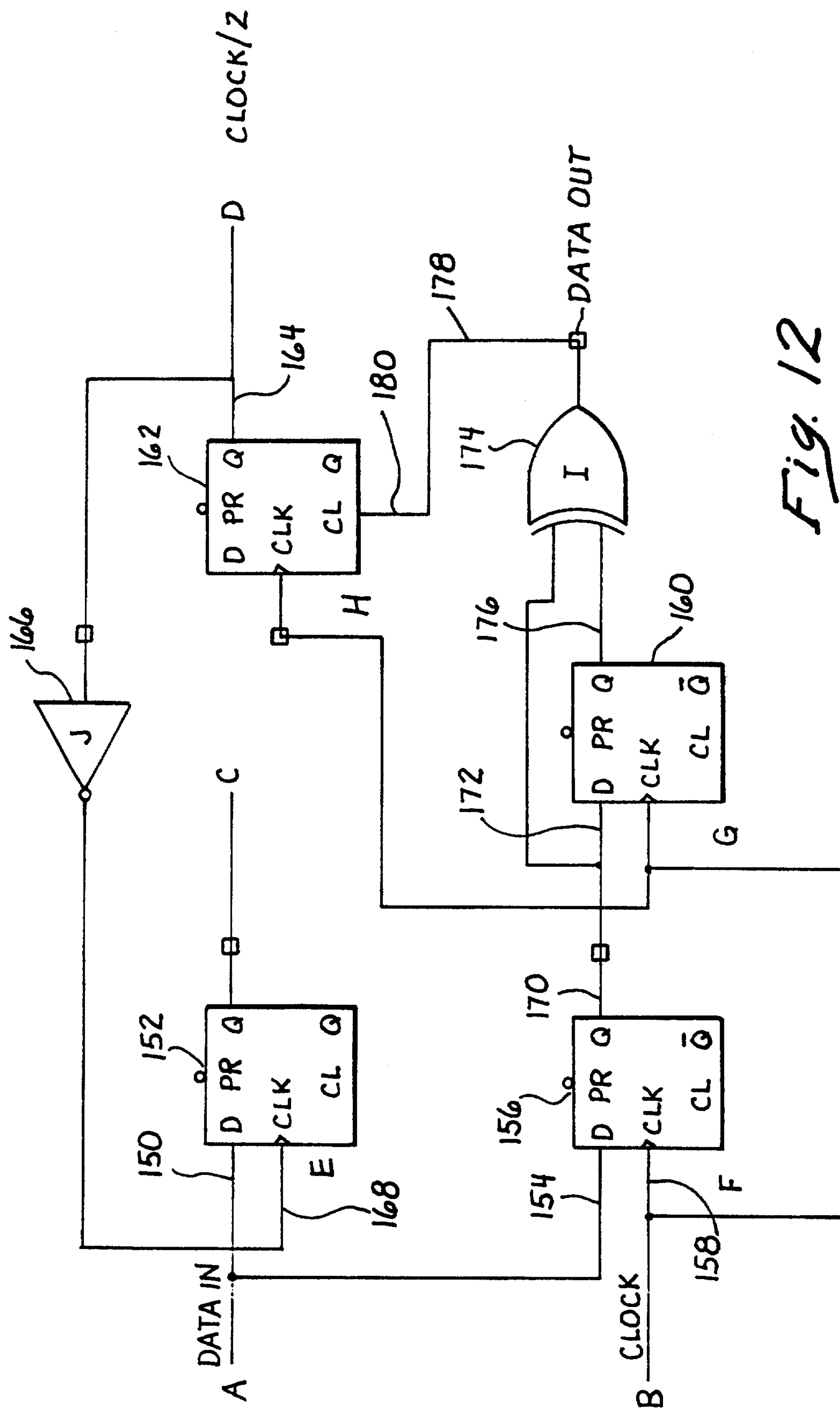
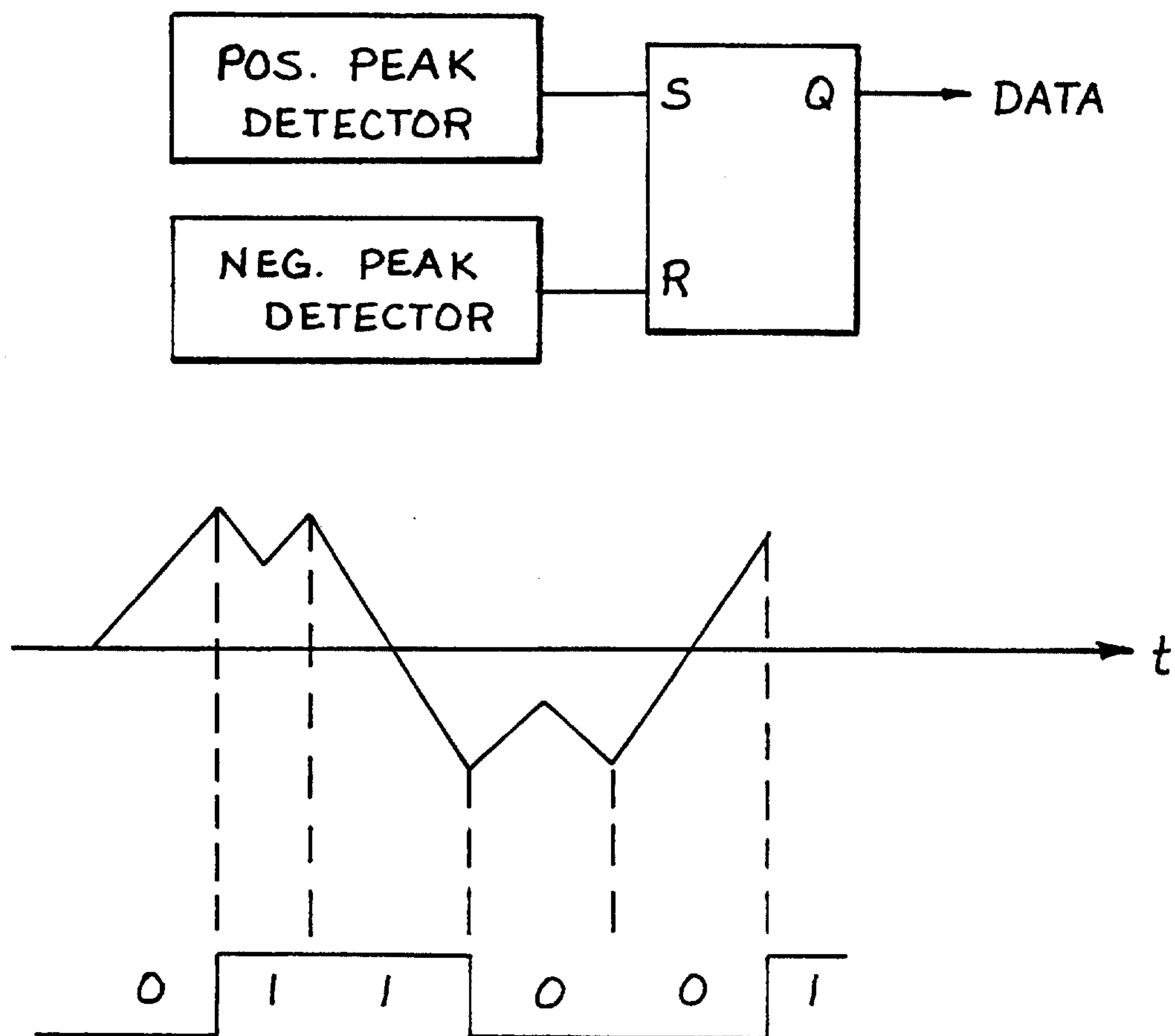
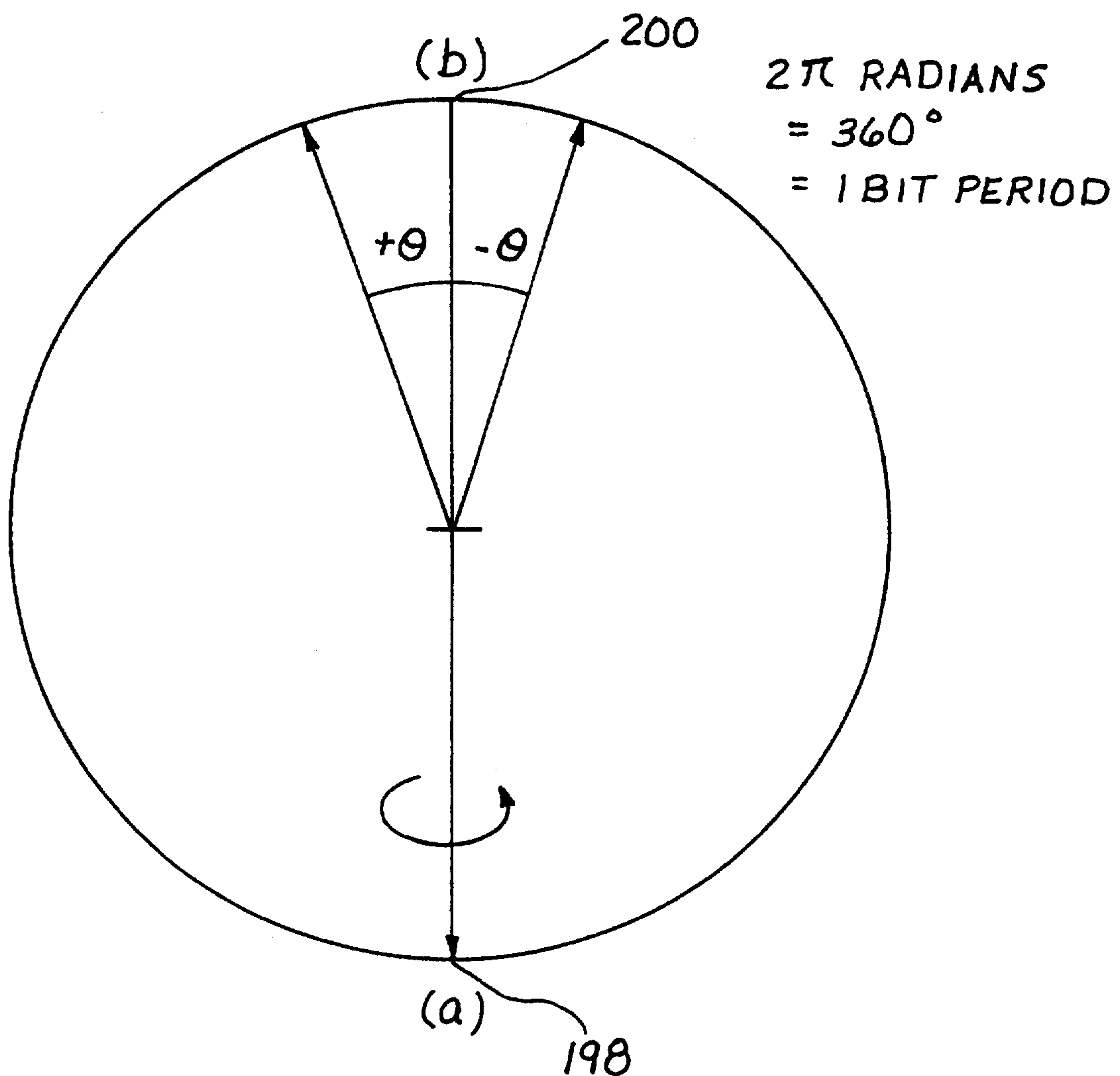


Fig. 12

*Fig. 13*

*Fig. 14*

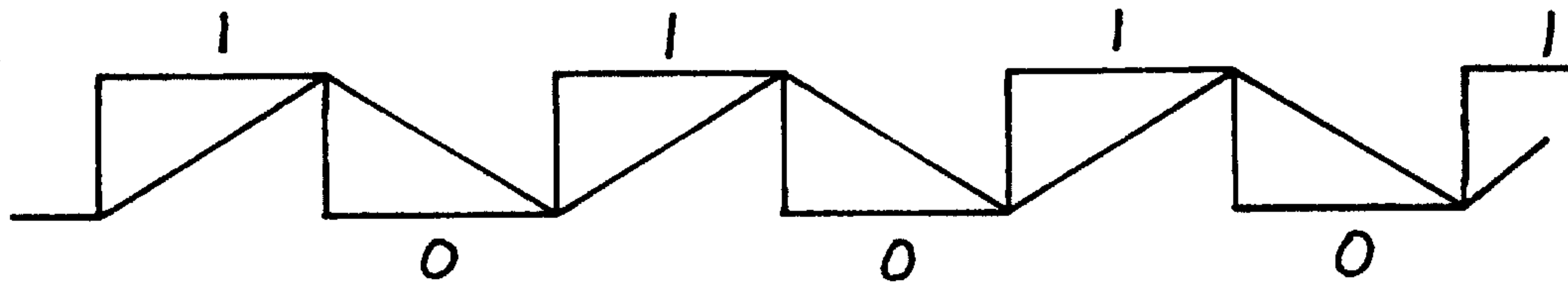
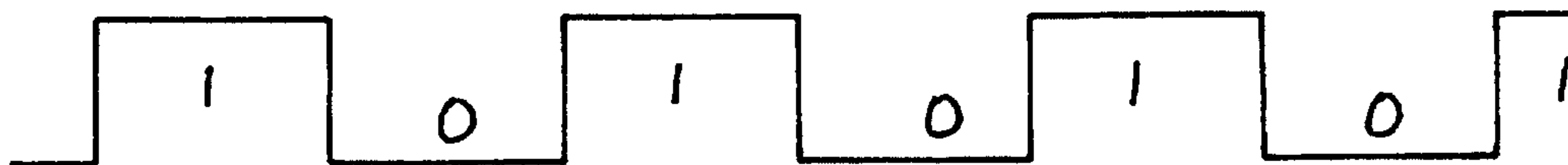
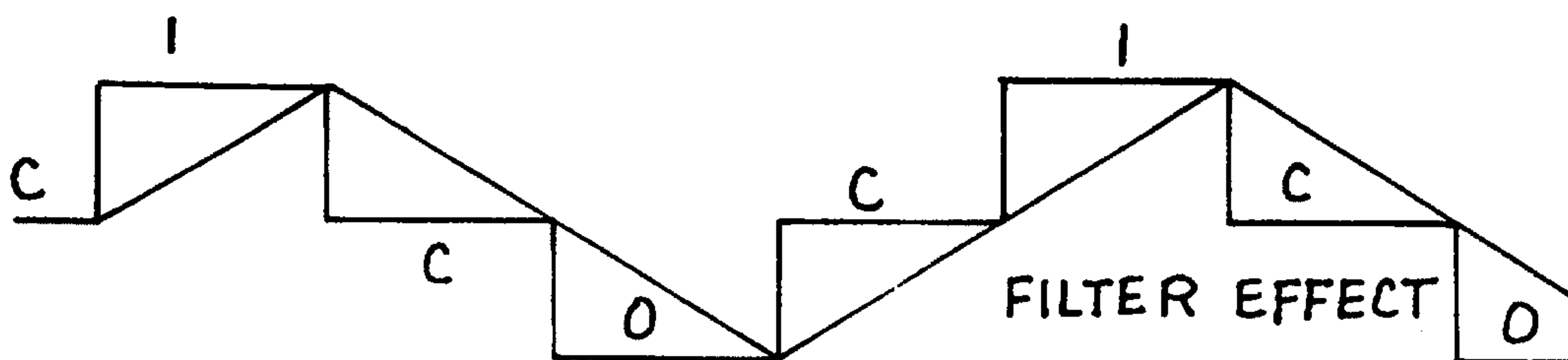
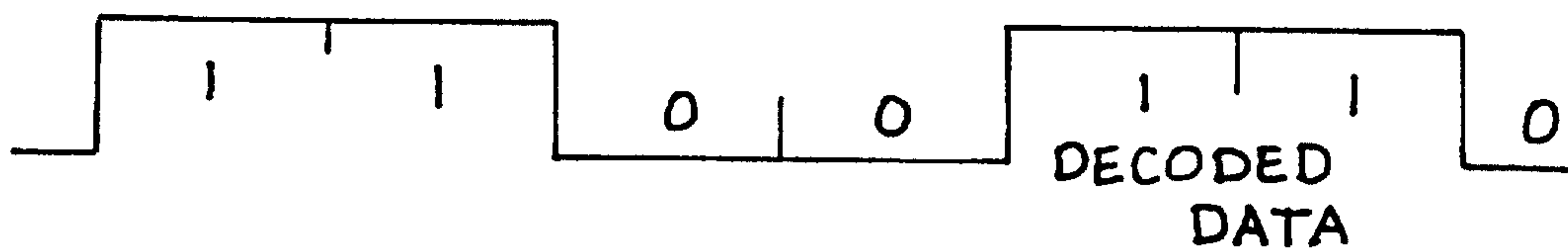
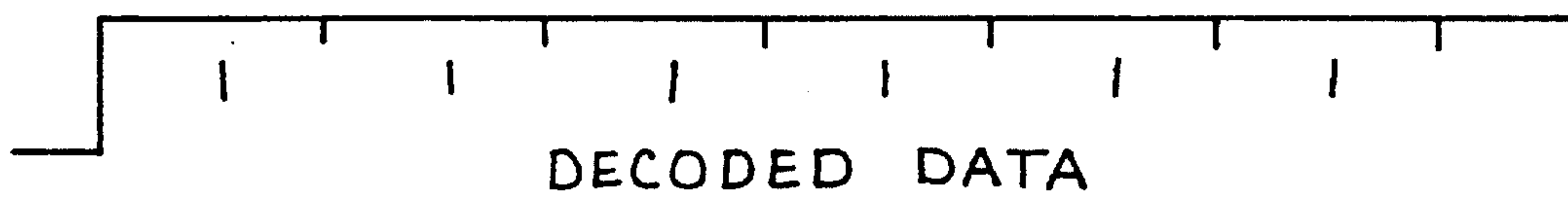
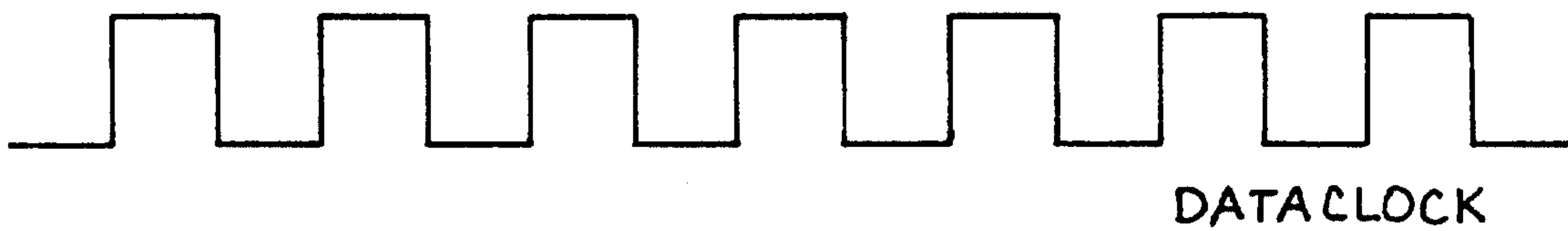
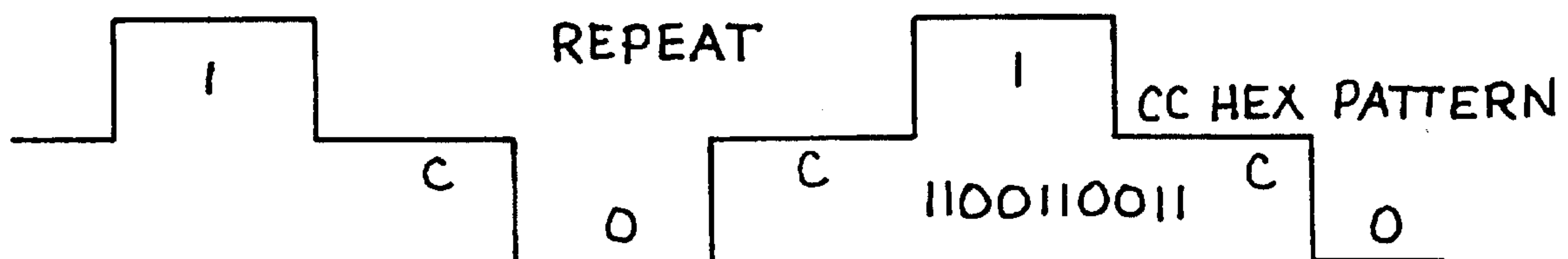
*Fig. 15a**Fig. 15b**Fig. 15c**Fig. 15d*

*Fig. 15e**Fig. 15f**Fig. 15g**Fig. 15h*

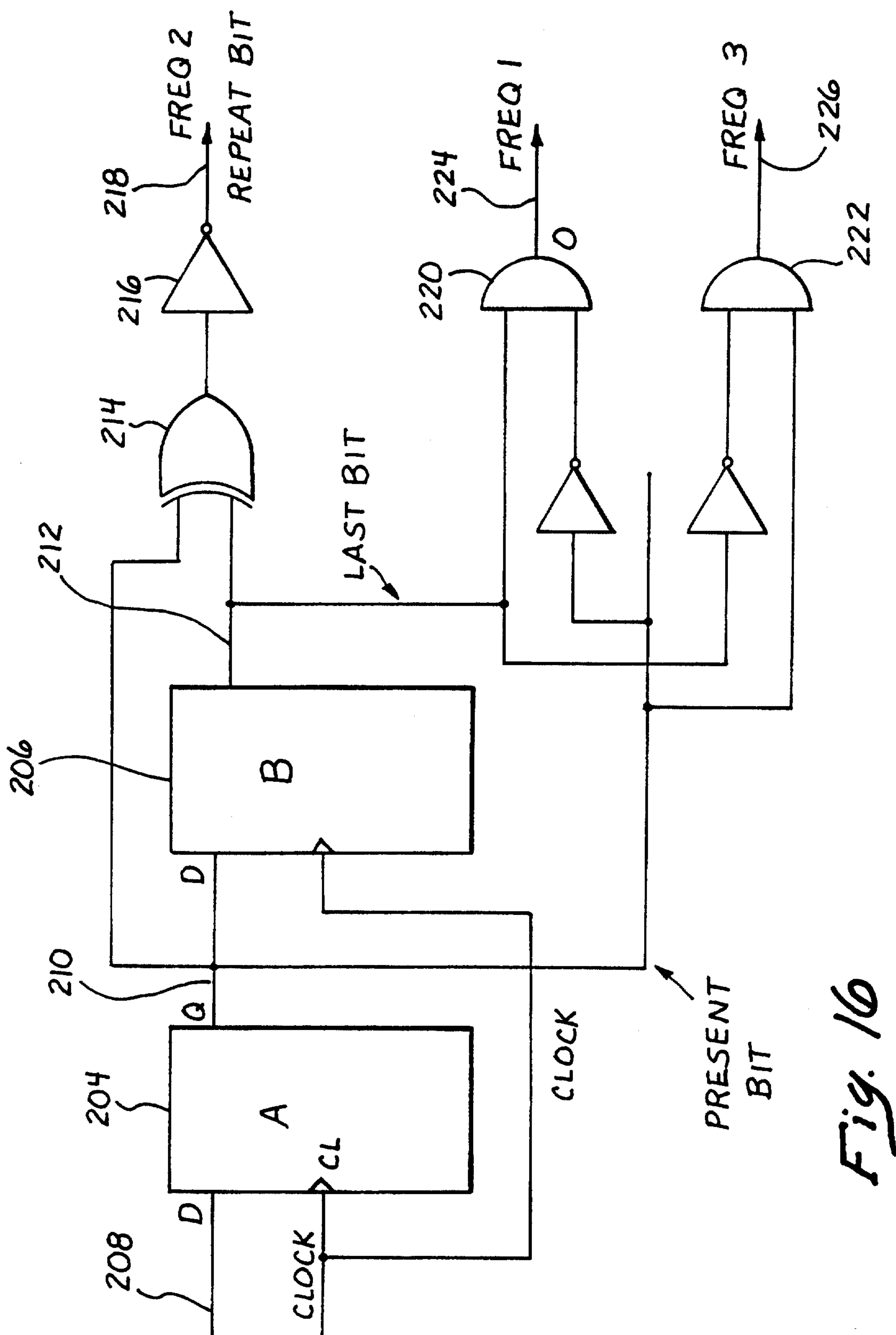


Fig. 16

