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(71) Applicant (for all designated States except US): **BAE SYSTEMS PLC** [GB/GB]; 6 Carlton Gardens, London SW1Y 5AD (GB).

(72) Inventor; and

(75) Inventor/Applicant (for US only): **GOULD, Dale Martin** [GB/GB]; Bae Systems, ATC Great Baddow, Chelmsford, Essex CM2 8HN (GB).

(74) Agent: **BAE SYSTEMS PLC, GROUP IP DEPT**; PO Box 87, Farnborough Aerospace Centre, Farnborough, Hampshire GU14 6YU (GB).

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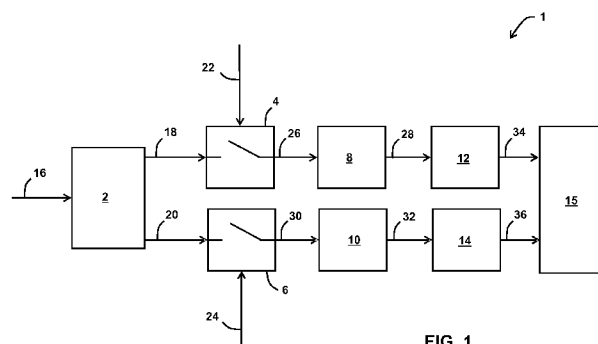


FIG. 1

(57) Abstract: A method and apparatus for performing sampling, comprising: sampling an input signal (16) using a first sampling signal (22) to provide a first sampled signal (26); sampling the input signal (16) using a second sampling signal (24) to provide a second sampled signal (30); measuring a frequency of one of the first sampled signal (26) and second sampled signal (30); measuring a phase difference between the first and the second sampled signal (26, 30); and determining an input signal frequency using the frequency and the phase difference; wherein the second sampling signal (24) has substantially equal frequency to the first sampling signal (22); and a waveform of the second sampling signal (24) is shifted in time with respect to a waveform of the first sampling signal (22). Measuring a frequency of the first or second sampled signal may be performed within a Nyquist zone of an analogue-to-digital converter (12, 14).

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SAMPLING

FIELD OF THE INVENTION

The present invention relates to sampling and apparatus for performing
5 sampling.

BACKGROUND

Conventionally, when sampling signals that have a frequency outside the
Nyquist bandwidth of an analogue-to-digital converter (ADC) being used to
10 perform the sampling of the signal, more than one such ADC is required to
achieve the sampling rate required by the Nyquist sampling theorem. For
example, if a sampling rate of 8GHz is required, and the fastest sampling rate
that can be achieved by an ADC is 1GHz, then eight such ADCs are required to
achieve the required sampling rate.

15 However, although the required sampling rate may be achieved, the
response of the ADCs, i.e. the bandwidth that the ADCs can respond to, may be
too low for the signal being sampled.

More details regarding background art are as follows.

When sampling RF signals an ADC is usually limited to a signal within its
20 first Nyquist band. For a sample rate of f_s the Nyquist limit is given as $f_s/2$. To
receive signals beyond $f_s/2$ it is usually necessary to either:

- a) sample faster; or
- b) to interleave several (M) ADCs offset in time to achieve a higher
effective sampling rate.

25 Approach a) is usually limited by current ADC technology and fabrication
techniques. Thus approach b) is usually adopted. This approach, i.e. prior art
digitization, is shown in Figure 7. In the Figure 7 prior art arrangement, each
clock is sampling at the rate f_s . The timing difference between clocks is
 $1/(M \cdot f_s)$, as shown in Figure 8 which schematically shows a prior art clock
30 waveforms example. The processor takes the first sample of data from

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Memory1 sample 1, then the first sample from Memory2 sample 2, then the first sample from Memory3 sample 3, etc up to MemoryM. In this way the effective sampling rate is increased to $M \cdot f_s$. This new sampling system has a Nyquist limit of $M \cdot f_s / 2$.

5 There are a number of limitations to this approach:

c) The input bandwidth limit of the individual ADCs may well be less than $M \cdot f_s / 2$ particularly when M is large. Thus limiting the maximum input frequency of the system.

10 d) Each ADC's input or the splitter/buffer output feeding it can introduce an amplitude and a phase error.

e) The time offset of $1/(M \cdot f_s)$ between adjacent clocks is hard to get perfect.

15 A consequence of d) and e) is that errors are introduced onto the interleaved signal (interleaving errors), which limit the dynamic range of the sampling system.

Consequently many interleaved systems require error correction.

Examples of such arrangements are described in the following publications: US patent 5,294,926 by Corcoran; International Patent WO 2004/004130 by Wood; US Patent 4,763,105 by Jenq; European Patent EP1 720 259 by Fernandez;
20 International patent WO 2008/082835 by Fang.

All these arrangements are for systems that 'calibrate' M interleaved ADCs to reduce the errors introduced by d) and e) above. The whole purpose of these systems is to achieve a higher sample rate $M \cdot f_s$ with successive samples separated in time by $1/(M \cdot f_s)$.

25 Different algorithms are used but each makes use of the sampled data from each ADC and often require frequency domain information obtained by a time to frequency domain transformation (e.g. Discrete Fourier Transform) and phase information to perform the error reduction. Some measure errors and attempt to correct for timing delays by the control of external delay elements.

30 Others calculate a correction matrix and multiply the input signals by the correction matrix to obtain the corrected output.

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As time delays, amplitudes and phases can drift overtime, usually due to changes in temperature (but can also be due to aging effects), it is usually necessary to repeat the calibration phase periodically.

In the above mentioned prior art arrangements, once the corrections are
5 made the new interleaved ADC system can instrument signals up to $M \cdot f_s/2$, the Nyquist limit for M interleaved ADCs provided the input bandwidth of the individual ADCs extends to $M \cdot f_s/2$. For large M this tends not to be true.

Prior art interleaved ADC systems such as those mentioned above can, after calibration, identify signals correctly up to $M \cdot f_s/2$ provided the ADCs' input
10 bandwidth can pass the signal. However, all such interleaved ADC systems will incorrectly identify the frequency of input signals above $M \cdot f_s/2$ due to aliasing.

In the above prior art arrangements, as far as the first two ADCs are concerned of the M interleaved ADCs, one is delayed relative to the other. For these M interleaved ADCs the whole purpose of this delay is for all the M ADCs
15 in the sampling system to be sampling equally spaced in time.

WO 2008/039528 (Fang) describes a further prior art arrangement. In the described arrangement the maximum frequency that the system can instrument is dictated by the frequency response of the ADCs in use.

20 SUMMARY OF THE INVENTION

In a first aspect the present invention provides a method of performing sampling, the method comprising sampling an input signal using a first sampling signal to provide a first sampled signal, sampling the input signal using a second sampling signal to provide a second sampled signal, measuring a
25 frequency of at least one of the first sampled signal and the second sampled signal, measuring a phase difference between the first sampled signal and the second sampled signal, and determining a frequency of the input signal using the measured frequency and the measured phase difference, wherein the second sampling signal has substantially equal frequency to the first sampling
30 signal, and a waveform of the second sampling signal is shifted in time with respect to a waveform of the first sampling signal.

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The steps of sampling an input signal using a first sampling signal to provide a first sampled signal and sampling the input signal using a second sampling signal to provide a second sampled signal may comprise using external sampling gates that have offset clocks.

- 5 The first sampled signal and the second sampled signal may be the only two sampled signals provided.

Narrow input gates may be arranged to mix the input signal down into the bandwidth of ADCs.

- 10 The measured frequency of at least one of the first sampled signal and the second sampled signal may be a measured alias frequency of the true frequency of at least one of the first sampled signal and the second sampled signal.

The waveform of the second sampling signal may be substantially equal to the waveform of the first sampling signal.

- 15 The method may further comprise the step of determining the amplitude of the input signal using the waveforms and magnitudes of the sampling signals, values of the frequency of the waveforms of the sampling signals, and the determined input signal frequency.

- 20 The step of measuring a frequency of at least one of the first sampled signal and the second sampled signal may be performed within a Nyquist zone of an analogue-to-digital converter.

- 25 The step of determining a frequency of the input signal may further comprise using a value of a frequency of a sampling signal, a value of the time shift that the second sampling signal is shifted in time relative to the first sampling signal, and a value of the time period of a sampling signal.

The frequency of the input signal may be determined using the following formula:

$$f_{\text{sig}} = -1 * \text{Spectral\_index} * f_s + \text{Measured\_frequency}$$

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where: f_{sig} is the frequency of the input signal;

f_s is the frequency of a sampling signal;

Measured_frequency is a measured value of a frequency of at
5 least one of the first sampled signal and the second sampled signal; and

$$\text{Spectral\_index} = \frac{\text{Phase\_difference}}{\text{Phase\_step\_for\_first\_indexed\_spectral\_line}}$$

where: Phase_difference is the measured phase difference; and

Phase_step_for_first_indexed_spectral_line is a value of a phase
10 step of a first spectral line in a frequency spectrum of the sampling signals.

The frequency of the input signal may be determined using the following formula:

$$f_{\text{sig}} = -1 * \text{Spectral\_index} * f_s + \text{Measured\_frequency}$$

15

where: f_{sig} is the frequency of the input signal;

f_s is the frequency of a sampling signal;

Measured_frequency is a measured value of a frequency of at
least one of the first sampled signal and the second sampled signal; and

$$20 \quad \text{Spectral\_index} = \text{Round} \left[\frac{\text{Phase\_difference}}{\text{Phase\_step\_for\_first\_indexed\_spectral\_line}} \right]$$

where: Phase_difference is the measured phase difference; and

Phase_step_for_first_indexed_spectral_line is a value of a phase
step of a first spectral line in a frequency spectrum of the sampling signals; and

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Round[X] is the value of X rounded to the nearest integer.

Phase_step_for_first_indexed_spectral_line may be given by the following equation:

$$\text{Phase\_step\_for\_first\_indexed\_spectral\_line} = -2 \cdot \pi \cdot \frac{t_{sep}}{T}$$

- 5 where: t_{sep} is a value of the time shift that the waveform of the second sampling signal is shifted by with respect to the waveform of the first sampling signal; and

T is a value of a time period of a sampling signal.

- The step of measuring a frequency of at least one of the first sampled
 10 signal and the second sampled signal may comprise filtering the first sampled signal such that it lies within a Nyquist zone of a first analogue-to-digital converter, using the first analogue-to-digital converter, converting to a discrete signal the filtered first sampled signal, filtering the second sampled signal such that it lies within a Nyquist zone of a second analogue-to-digital converter, using
 15 the second analogue-to-digital converter, converting to a discrete signal the filtered second sampled signal, and measuring a value of the frequency of signal converted by the first analogue-to-digital converter and/or a value of the frequency of signal converted by the second analogue-to-digital converter.

- In a further aspect the present invention provides an apparatus for
 20 performing sampling, the apparatus comprising a first sampler adapted to sample an input signal using a first sampling signal to provide a first sampled signal, a second sampler adapted to sample the input signal using a second sampling signal to provide a second sampled signal, means for measuring a frequency of at least one of the first sampled signal and the second sampled
 25 signal, means for measuring a phase difference between the first sampled signal and the second sampled signal, and a processor adapted to determine a frequency of the input signal using the measured frequency and the measured phase difference, wherein the second sampling signal has substantially equal

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frequency to the first sampling signal, and a waveform of the second sampling signal is shifted in time with respect to a waveform of the first sampling signal.

In a further aspect the present invention provides computer program or plurality of computer programs arranged such that when executed by a computer system it/they cause the computer system to operate in accordance with the method of any of the above aspects.

In a further aspect the present invention provides a machine readable storage medium storing a computer program or at least one of the plurality of computer programs according to the above aspect.

10

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a schematic block diagram of an apparatus used in an embodiment of an offset sampling process;

Figure 2 is a process flow chart showing certain steps of an embodiment of the offset sampling process;

Figure 3 is a schematic illustration showing a first sampling signal and a second sampling signal in the time domain;

Figure 4 is a schematic illustration showing the first sampling signal in the frequency domain;

Figure 5 is a schematic illustration of the frequency spectrum of an output signal;

Figure 6 is a schematic illustration of the frequency spectrum of a filtered output signal;

Figure 7 is a schematic illustration of a prior art digitization approach;

Figure 8 schematically shows a prior art clock waveforms example;

Figure 9A shows, for the prior art approach, the effective sampling clock for M interleaved ADCs; and

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Figure 9B shows, by way of comparison with Figure 9A, a schematic representation (not to scale) of an example of the effective sampling clock of embodiments of an offset sampling approach.

5 DETAILED DESCRIPTION

Figure 1 is a schematic block diagram of an apparatus used in an embodiment of an offset sampling process described below with reference to Figure 2. The apparatus shown in Figure 1 is hereinafter referred to as the “offset sampler 1”. The term “offset” is used herein because the sampling
10 signals used in the below described apparatus and process are offset in time relative to each other, as described in more detail later below with reference to Figure 3.

The offset sampler 1 comprises a splitter 2, a first sampler 4, a second sampler 6, a first Nyquist filter 8, a second Nyquist filter 10, a first analogue-to-
15 digital converter (ADC) 12, a second ADC 14, and a processor 15.

The splitter 2 receives an input signal 16. The input signal 16 is a signal from a system being sampled by the offset sampler 1. In this embodiment, the input signal 16 is a continuous wave signal. The splitter 2 splits the input signal 16 into two substantially identical copies of the input signal 16, hereinafter
20 called the “first input copy 18” and “the second input copy 20”.

The first input copy 18 is sent from the splitter 2 to the first sampler 4.

The first sampler 4 receives the first input copy 18 and a signal corresponding to a first sampling function, hereinafter referred to as the “first sampling signal 22”. The first sampling signal 22 is described in more detail
25 later below with reference to Figures 3 and 4. The first sampler 4 multiplies (in the time domain) the received first input copy 18 and first sampling signal 22 to produce a first output signal 26. The first output signal 26 is sent from the first sampler 4 to the first Nyquist filter 8.

The first Nyquist filter 8 filters the received first output signal 26 as
30 described in more detail later below with reference to Figure 5. The filtered first

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output signal, hereinafter referred to as the “first filtered signal 28” is sent from the first Nyquist filter 8 to the first ADC 12.

The first ADC 12 converts the received continuous first filtered signal 28 into a discrete signal, hereinafter referred to as the “first discrete signal 34”.

5 The first discrete signal 34 is sent to the processor 15 from the first ADC 12.

The second input copy 20 is sent from the splitter 2 to the second sampler 6.

10 The second sampler 6 receives the second input copy 20 and a signal corresponding to a second sampling function, hereinafter referred to as the “second sampling signal 24”. The second sampling signal 24 is described in more detail later below with reference to Figures 3 and 4. The second sampler 6 multiplies (in the time domain) the received second input copy 20 and second sampling signal 24 to produce a second output signal 30. The second output
15 signal 30 is sent from the second multiplier 6 to the second Nyquist filter 10.

The second Nyquist filter 10 filters the received second output signal 30 as described in more detail later below with reference to Figure 5. The filtered second output signal, hereinafter referred to as the “second filtered signal 32” is sent from the second Nyquist filter 10 to the second ADC 14.

20 The second ADC 14 converts the received continuous second filtered signal 32 into a discrete signal, hereinafter referred to as the “second discrete signal 36”.

The second discrete signal 36 is sent to the processor 15 from the second ADC 14.

25 The first and second discrete signals 34, 36 are processed by the processor 15 to determine the frequency of the input signal 16.

Figure 2 is a process flow chart showing certain steps of an embodiment of an offset sampling process. The offset sampler 1 of Figure 1 is used to perform this offset sampling process.

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At step s2, the input signal 16 is received at the splitter 2. The input signal 16 is a continuous wave signal from a system being sampled by the offset sampler 1. In this embodiment, for clarity and ease of understanding the frequency of the input signal f_{sig} is 7.1GHz. In this embodiment, we assume that
 5 the frequency of the input signal 16 is known. However, in other embodiments, this frequency is unknown, but may be determined by implementation of the offset sampling process using the offset sampler 1.

The input signal 16 is a real input. In this embodiment, the input signal 16 is given by the following formula:

$$10 \quad \text{Input\_signal} = B \cdot \cos(\omega_{sig} \cdot t) = B \cdot \frac{e^{j \cdot \omega_{sig} \cdot t} + e^{-j \cdot \omega_{sig} \cdot t}}{2}$$

where: B is the maximum amplitude of the input signal; and

ω_{sig} is the angular frequency of the input signal.

The angular frequency of the input signal ω_{sig} is given by the following formula:

$$15 \quad \omega_{sig} = 2\pi f_{sig}.$$

Thus, the input signal 16 is real and consists of two contra rotating

$$\text{vectors: } \frac{B}{2} \cdot e^{j \cdot \omega_{sig} \cdot t} \quad \text{and} \quad \frac{B}{2} \cdot e^{-j \cdot \omega_{sig} \cdot t}.$$

At step s4, the splitter 2 splits the input signal 16 into the first input copy 18 and the second input copy 20. The first and second input copies 18, 20 are
 20 sent to the first and second multipliers 4, 6 respectively.

At step s6, the first input copy 18 is multiplied by the first sampling signal 22 by the first sampler 4. Also, the second input copy 20 is multiplied by the second sampling signal 24 by the second sampler 6.

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Figure 3 is a schematic illustration (not to scale) showing the first sampling signal 22 and the second sampling signal 24 in the time domain. The sampling signals 22, 24 are shown on a common time axis.

The first sampling signal 22 comprises a pulse having a repetition period
5 T, a pulse width t_0 , and amplitude A.

In this embodiment, the repetition period is one nanosecond ($T = 1\text{ns}$). Also, in this embodiment the pulse width is a sixteenth of a nanosecond ($t_0 = 62.5\text{ps}$).

The second sampling signal 24 is substantially the same as the first
10 sampling signal 22, but is shifted in time relative to the first sampling signal 22 by a separation time t_{sep} . In other words, the second sampling signal 24 comprises a pulse having a repetition period T, a pulse width t_0 , and amplitude A. Also, the centre of a pulse of the second sampling signal 24 is separated in time by a separation time t_{sep} from the centre of the corresponding pulse of the
15 first sampling signal. The same characters are used herein to indicate the same characteristics of the first and second sampling signals 22, 24, i.e. T is used to indicate the repetition periods of the first and second sampling signals 22, 24, t_0 is used to indicate the pulse widths of the first and second sampling signals 22, 24, and A is used to indicate the amplitudes of the first and second sampling
20 signals 22, 24.

In this embodiment, the separation time t_{sep} is a sixteenth of a nanosecond ($t_{\text{sep}} = 62.5\text{ps}$), i.e. $t_{\text{sep}} = t_0$.

Figure 4 is a schematic illustration (not to scale) showing the first sampling signal 22 in the frequency domain. The second sampling signal 24
25 has the same frequency distribution as the first sampling signal 22.

The frequency spectrum of the first sampling signal 22 comprises a plurality of spectral lines. The spectral lines are spaced by a frequency amount known as the sampling frequency f_s . The relationship between the sampling frequency f_s , the repetition period T, and a wavelength of the first sampling
30 signal ω_0 is given by:

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$$T = \frac{1}{f_s} = \frac{2 \cdot \pi}{\omega_0}$$

Thus, the sampling frequency is equal to one gigahertz ($f_s = 1\text{GHz}$).

The spectral lines have a $|\sin(x)/x|$ shaped distribution. An envelope for this $|\sin(x)/x|$ shaped distribution is shown in Figure 4 by a dotted line which is indicated by the reference numeral 38. The value for t_0 effectively is used to define the bandwidth of interest as $1/t_0$ defines the first null in the $|\sin(x)/x|$ envelope. In this embodiment, $t_0 = t_{\text{sep}} = 62.5\text{ps}$ and T is 1 ns. Thus, the $|\sin(x)/x|$ spectrum for the sampling function has a first null at 16GHz and the lines are spectral spaced by 1GHz. The value of the repetition period T defines the sampling rate which has to be matched by the ADC.

The Fourier series of first sampling signal 22 is given by the formula:

$$f_1(t) = \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j \cdot n \cdot \omega_o \cdot t}$$

where: α_n is given by the formula:

$$\alpha_n = A \cdot \frac{t_0}{T} \cdot \frac{\sin\left(\frac{n \cdot \omega_o \cdot t_0}{2}\right)}{\frac{n \cdot \omega_o \cdot t_0}{2}}$$

15

The Fourier series of the second sampling signal 24 is given by the following formula:

$$f_2(t) = \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j \cdot n \cdot \omega_o \cdot (t - t_{\text{sep}})}$$

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Since $T = \frac{1}{f_s} = \frac{2 \cdot \pi}{\omega_0}$ the Fourier series of the second sampling signal 24

may be expressed as:

$$f_2(t) = \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j \cdot n \cdot \omega_0 \cdot (t - t_{sep})} = \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j(n \cdot \omega_0 \cdot t)} \cdot e^{-j \left(\frac{n \cdot 2 \cdot \pi \cdot t_{sep}}{T} \right)}$$

- 5 In other words, the Fourier transform of the second sampling signal 24 is a linearly phase shifted version of Fourier series of first sampling signal 22. The

term $e^{-j \left(\frac{n \cdot 2 \cdot \pi \cdot t_{sep}}{T} \right)}$ contains the phase shift $\frac{n \cdot 2 \cdot \pi \cdot t_{sep}}{T}$ incurred on the spectral line $n \cdot \omega_0$ for the second sampling signal 24 due to the delay relative to first sampling signal 22.

- 10 The product of the input signal 16 and the first sampling signal 22 is the first output signal 26, which is given by the following formula:

$$Output_1(t) = B \cdot \frac{e^{j \cdot \omega_{sig} \cdot t} + e^{-j \cdot \omega_{sig} \cdot t}}{2} \cdot \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j \cdot n \cdot \omega_0 \cdot t}$$

- 15 The product of the input signal 16 and the second sampling signal 24 is the second output signal 30, which is given by the following formula:

$$Output_2(t) = B \cdot \frac{e^{j \cdot \omega_{sig} \cdot t} + e^{-j \cdot \omega_{sig} \cdot t}}{2} \cdot \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j \cdot n \cdot \omega_0 \cdot t} \cdot e^{-j \cdot n \cdot 2 \cdot \pi \cdot t_{sep} / T}$$

20

An output signal 26, 30 is equal to the sampling spectrum shifted by $\pm \omega_{sig}$.

Figure 5 is a schematic illustration (not to scale) of the frequency spectrum of an output signal 26, 30. The frequency of the input signal f_{sig} is

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7.1GHz, thus the output spectrum has peaks at ± 7.1 GHz. For clarity, an envelope of the product of a sampling signal and the -7.1GHz peak of the input signal is shown in Figure 5 as a solid line indicated by the reference numeral 100. Also, for clarity an envelope of the product of a sampling signal and the
5 +7.1GHz peak of the input signal is shown in Figure 5 as a dotted line indicated by the reference numeral 101.

As shown schematically in Figure 5, a pair of spectral lines is present at each of the following output locations: 0GHz, 1GHz, 2 GHz, 3GHz, ..., nGHz. The spectral lines of each respective spectral line pair are positioned either side
10 of the respective output locations. Because the frequency of the input signal f_{sig} is 7.1GHz, spectral lines appear at ± 100 MHz, ± 900 MHz, ± 1.1 GHz, ± 1.9 GHz, ± 2.1 GHz, ± 2.9 GHz etc. In this embodiment, the outputs are generated at a 1GHz data rate (because $T = 1$ ns).

At step s8, the first output signal 26 is filtered by the first Nyquist filter 8
15 to produce the first filtered signal 28. Also, the second output signal 30 is filtered by the second Nyquist filter 10 to produce the second filtered signal 32.

In the offset sampler 1, the Nyquist filters 8, 10 perform multiple tasks, namely: sampler product filtering, pulse interpolation filtering, and ADC anti-aliasing filtering. The Nyquist filters 8, 10 ensure that the signals the ADCs 12,
20 14 are to digitise (at step s10 below) are signals within the Nyquist capabilities of the ADCs 12, 14.

The Nyquist frequency is equal to half of the frequency of a sampling signal, i.e. in this embodiment the Nyquist frequency is $f_s/2 = 500$ MHz.

In the description of this embodiment, to simplify the explanation, the
25 Nyquist filters 8, 10 are treated as idealised or so-called "brick wall filters" that are effective at the Nyquist frequency of 500MHz, in other words, the Nyquist filters 8, 10 block all frequencies above 500 MHz and let all frequencies between 0MHz and 500MHz pass.

Figure 6 is a schematic illustration (not to scale) of the frequency
30 spectrum of a respective first or second output signal 26, 30 after filtering using a respective first or second Nyquist filter 8, 10.

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The frequencies not blocked by a Nyquist filters are those between -500MHz and 500MHz. This frequency band is indicated in Figure 6 by dotted lines and is hereinafter referred to as the "Nyquist zone 40".

The only two spectral or alias lines that lie within the Nyquist zone 40 are
5 positioned at $\pm 100\text{MHz}$.

At step s10, the first ADC 12 is used to digitise the first filtered signal 28 to produce the first discrete signal 34. Also, the second ADC 14 is used to digitise the second filtered signal 32 to produce the second discrete signal 36.

Thus, in this embodiment the ADCs 12, 14 are 1GHz ADCs.

10 The first discrete signal 34 and the second discrete signal 36 are sent to the processor 15.

At step s12, the first and second discrete signals 34, 36 are measured at the processor 15. In this embodiment, signals are measured to be $\pm 100\text{MHz}$.

To obtain frequency information from the ADC samples it is necessary to
15 transform from the time domain to the frequency domain, for example by performing a Fast Fourier Transform (FFT). For a real FFT the information shown in the negative part of the frequency spectrum is a mirror of that in the positive half of the frequency spectrum. Thus, usually the negative portion of the spectrum is suppressed for display purposes but is retained here for clarity and
20 ease of understanding.

As mentioned above, the input signal 16 consists of two contra rotating

vectors: $\frac{B}{2} \cdot e^{j \cdot \omega_{sig} \cdot t}$ and $\frac{B}{2} \cdot e^{-j \cdot \omega_{sig} \cdot t}$. The measured 100MHz spectral

line is generated by the multiplication of the vector $\frac{B}{2} \cdot e^{j \cdot \omega_{sig} \cdot t}$ by $\alpha_n e^{jn\omega_0 t}$
(where $n=-7$ in this embodiment). Also, the measured -100MHz spectral line is

25 generated by the multiplication of the vector $\frac{B}{2} \cdot e^{-j \cdot \omega_{sig} \cdot t}$ by $\alpha_n e^{jn\omega_0 t}$

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(where $n=7$ in this embodiment). All other products fall outside the Nyquist zone 40.

The frequency/phase of the 100MHz spectral line ($n = -7$) of this embodiment at the first ADC 12 is given by:

$$5 \quad Output_{1(100MHz)} = \alpha_{-7} e^{j \cdot (-7 \cdot \omega_0 + \omega_{sig}) \cdot t} = \alpha_{-7} e^{j \cdot \omega_{100MHz} \cdot t}$$

where: ω_{100MHz} is the angular frequency of the 100MHz spectral line.

Also, the frequency/phase of the 100MHz spectral line ($n = -7$) of this embodiment at the second ADC 14 is given by:

$$10 \quad Output_{2(100MHz)} = \alpha_n e^{j \cdot ((n \cdot \omega_0 + \omega_{sig}) \cdot t - 2 \cdot n \cdot \pi \frac{t_{sep}}{T})} = \alpha_{-7} e^{j \cdot (\omega_{100MHz} \cdot t + 7\pi/8)}$$

At step s12, the phase difference between the first and second discrete signals 34, 36 is determined by the processor 15.

In this embodiment, this phase difference is given by the following formula:

$$15 \quad Exponential\_Phase\_diff = \frac{Output_{2(100MHz)}}{Output_{1(100MHz)}} = \frac{\beta \cdot \alpha_n e^{j \cdot ((n \cdot \omega_0 + \omega_{sig}) \cdot t - 2 \cdot n \cdot \pi \frac{t_{sep}}{T})}}{\beta \cdot \alpha_n e^{j \cdot (n \cdot \omega_0 + \omega_{sig}) \cdot t}}$$

Thus, for $n = -7$ the above formula becomes:

$$Exponential\_Phase\_diff = \frac{Output_{2(100MHz)}}{Output_{1(100MHz)}} = \frac{\alpha_{-7} e^{j \cdot (\omega_{100MHz} \cdot t + 7\pi/8)}}{\alpha_{-7} e^{j \cdot \omega_{100MHz} \cdot t}} = e^{j \cdot 7\pi/8}$$

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For the 100MHz spectral line, the angular phase difference between $Output_{1(100MHz)}$ and $Output_{2(100MHz)}$ is $-n \cdot 2 \cdot \pi \cdot \frac{t_{sep}}{T}$ radians with $n = -7$, i.e. $\frac{7\pi}{8}$ radians.

At step s16, a spectral index value is determined by the processor 15. In this embodiment, the spectral index is defined as the measured phase difference between the respective signals at the respective first and second ADCs 12, 14 divided by the phase step for the first spectral line (i.e. the spectral line $n = 1$). In other words:

$$\text{Spectral\_index} = \frac{\text{Phase\_difference}}{\text{Phase\_step\_for\_first\_indexed\_spectral\_line}}$$

In this embodiment, the phase step for the first indexed spectral line is:

$$-2 \cdot \pi \cdot \frac{t_{sep}}{T} = -2 \cdot \pi \cdot \frac{1}{16} = -\frac{\pi}{8}.$$

15

Thus, in this embodiment the spectral index for the 100MHz spectral line is:

$$\frac{-7\pi/8}{\pi/8} = -7.$$

At step s18, the frequency of the input signal f_{sig} (which in this embodiment is known, but in real world application may be unknown) is determined by the processor 15. In this embodiment, the frequency of the input signal f_{sig} is determined using the following formula:

- 18 -

$$f_{\text{sig}} = -1 * \text{Spectral\_index} * f_s + \text{Measured\_frequency}$$

where Measured_frequency is the frequency measured at the ADCs 12, 14.

In this embodiment, $f_{\text{sig}} = -1 * -7 * 1\text{GHz} + 100\text{MHz} = 7.1\text{GHz}$.

5 Thus, an offset sampling process that advantageously correctly determines the frequency of an input signal f_{sig} is provided.

 An advantage of multiplying the input signal 16 by a sampling waveform, e.g. a pulse signal having a $|\sin(x)/x|$ shaped envelope (i.e. the first or second sampling signal 22, 24), is that the input signal 16 is advantageously mixed
10 down into the Nyquist zone of the ADCs 12, 14. In this embodiment, the 7.1GHz input signal 16 is mixed down such that it is represented by the spectral lines at $\pm 100\text{MHz}$ shown in Figure 6, which are within the $\pm 500\text{MHz}$ Nyquist zones of the ADCs 12, 14. Thus, the ADCs advantageously tend to be able to be used to sample a signal having a frequency outside the Nyquist zone of the
15 ADCs. However, the frequencies outside the Nyquist zone of the ADCs may be ambiguous.

 A further advantage of provided by the present invention is that the above mentioned ambiguity tends to be resolved by using the offset first and second sampling signals 22, 24. The offset between the pulses of the first and
20 second sampling signals 22, 24 produces a phase difference between the first and second discrete signals 34, 36 produced by the first and second ADCs 12, 14 respectively. This phase difference is used by the processor to determine a spectral index value which advantageously identifies the original frequency of the input signal 16.

25 In other words, the invention advantageously mixes down a frequency band of interest into the Nyquist zones of the ADCs, and advantageously exploits offset sampling to resolve the ambiguity that arises when this mixing down is performed.

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An advantage of this approach is that a wide, sparsely populated bandwidth is capable of being monitored with a minimum of hardware, i.e. less hardware than is needed conventionally.

5 A further advantage of the above described technique is that it tends to be able to be implemented in electronic surveillance measures (ESM) systems at a significantly lower price than current technology.

10 It should be noted that certain of the process steps depicted in the flowchart of Figure 2 and described above may be omitted or such process steps may be performed in differing order to that presented above and shown in Figure 2. Furthermore, although all the process steps have, for convenience and ease of understanding, been depicted as discrete temporally-sequential steps, nevertheless some of the process steps may in fact be performed simultaneously or at least overlapping to some extent temporally.

15 Apparatus, including the processor 15, for implementing the above arrangement, and performing the method steps to be described above, may be provided in part or completely by configuring or adapting any suitable apparatus, for example one or more computers or other processing apparatus or processors, and/or providing additional modules. The apparatus may comprise a computer, a network of computers, or one or more processors, for
20 implementing instructions and using data, including instructions and data in the form of a computer program or plurality of computer programs stored in or on a machine readable storage medium such as computer memory, a computer disk, ROM, PROM etc., or any combination of these or other storage media. The apparatus may, for example, include one or more spectrum analysers to
25 determine one or more of the frequencies, and/or one or more Vector Voltmeters to measure a phase.

In other embodiments, an optional additional step may be performed as follows. Using the known shape of the frequency spectrum of the sampling signals 22, 24, the known spectral line spacing, which can be determined from
30 the shape of the sampling signal i.e. the known values of t_0 and T in this embodiment), and the determined value of the original input signal frequency

- 20 -

f_{sig} , the original amplitude of the input signal B can be corrected to compensate for weighting loss (e.g. the $|\sin(x)/x|$ weighting loss). This advantageously tends to provide increased accuracy in estimations of the amplitude of the input signal.

5 In other words, the magnitude of the spectral lines is affected by which harmonic of the sampling spectrum is used to mix the signal into the ADCs' Nyquist bandwidths. Knowing the $|\sin(x)/x|$ amplitude distribution for any given sampling pulse width and the repetition rate, the amplitudes can also be adjusted to compensate for mixing losses. Thus, both corrected amplitude and
10 frequency information is advantageously provided.

The maximum phase that can be measured is less than π . This is because the phase becomes ambiguous, i.e. phases of $+\pi$ and $-\pi$ tend not to be able to be distinguished. Thus, in the above embodiment the maximum spectral line that could be used was $n = 7$ which, with the Nyquist filter
15 bandwidth of 500MHz, means that a maximum input signal of 7.5 GHz may be correctly determined. Thus, using two ADCs it is possible to resolve the input signal over fifteen Nyquist zones. By using input band defining filters for an input signal of 7.5 GHz it is possible to prevent any ambiguities due to higher frequencies.

20 The magnitude of the phase step can advantageously be made smaller by reducing t_{sep} such that a greater number of Nyquist zones can be instrumented before the ambiguity arises. There is a trade off here as the measured phase may be noisy in some systems and therefore the spectral line index will need to be able to cope with this noise. In other embodiments of the
25 invention, a spectral index according to the following formula is used which advantageously tends to alleviate the effects of this noise:

$$\text{Spectral\_index} = \text{Round} \left[\frac{\text{Phase\_difference}}{\text{Phase\_step\_for\_first\_indexed\_spectral\_line}} \right]$$

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where the Round[X] is the value of X rounded to the nearest integer. Using this spectral index, an error up to half of phase step for the first indexed spectral line can advantageously be tolerated

5 The offset sampler enables multiple Nyquist zones to be monitored with only two ADCs and can locate CW lines at the correct frequency. If multiple input frequencies alias to the same frequency resolution bin within the ADCs' Nyquist bands, it is advantageously possible to distinguish between the frequencies aliasing to the same frequency resolution bin by changing the ADC clock frequency.

10 In the above embodiments, the offset sampler described above with reference to Figure 1 is used to perform the offset sampling process described with reference to Figure 2. However, in other embodiments the offset sampling process may be performed with a different apparatus that is configured to perform the offset sampling process.

15 In the above embodiments, the frequency of the input signal f_{sig} is a known value 7.1GHz. However, in other embodiments the input signal may have a different frequency and may be either known or unknown.

20 In the above embodiments, for a set of continuous tones distributed over multiple Nyquist zones, the original input frequency can advantageously be determined. In other embodiments, the input signal may be a different type of signal, for example a pulse or an intermittent continuous wave signal. Frequency and/or amplitude information tends to be able to be determined in the same way as that described above for the continuous wave input signal. The "on" duration for a signal may need to allow sufficient signal to noise, after
25 processing, for the phase to be measured at each ADC output.

30 In the above embodiments, the sampling signals comprise a pulse having a repetition period $T = 1\text{ns}$, a pulse width $t_0 = 62.5\text{ps}$. However, in other embodiments, sampling signals having different repetition periods and/or pulse widths may be used. Also, in other embodiments sampling signals having waveforms of any shape may be used. In particular, in other embodiments any waveforms that provided a sampling function and enable the phase difference

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between two or more outputs to be used to determine the input frequency may be used.

In the above embodiments, the separation time t_{sep} is a sixteenth of a nanosecond ($t_{\text{sep}} = 62.5\text{ps}$). However, in other embodiments the separation time
5 t_{sep} may be a different appropriate value, and may or may not be equal to the pulse width t_0 .

In the description of the above embodiments, to clarify the description, the Nyquist filters are treated as idealised filters (so-called “brick wall filters”) that are effective at the Nyquist frequency of 500MHz (i.e. the Nyquist filters
10 block all frequencies above 500 MHz and let all frequencies between 0MHz and 500MHz pass). Thus, the Nyquist filters are anti-aliasing filters. However, in other embodiments one or more of the Nyquist filters may be a different appropriate type of filter and/or may be effective over a different appropriate frequency range to provide the anti-aliasing function.

15 The following discussion allows further understanding of the above described embodiments and some of their advantages.

As mentioned above in the “Background” section, in the earlier mentioned prior art arrangements, as far as the first two ADCs are concerned of the M interleaved ADCs, one is delayed relative to the other. For those M
20 interleaved ADCs the whole purpose of this delay is for all the M ADCs in the sampling system to be sampling equally spaced in time. In the above described embodiments of the present invention, however, in which “offset sampling” is performed, an aspect is that the timing of the two samples means that overall the sampling is not equally distributed in time.

25 In Figure 9A, for the prior art approach, the effective sampling clock for M interleaved ADCs is shown. By way of comparison, a schematic representation (not to scale) of an example of the effective sampling clock of the offset sampling approach as per the above described embodiments of the present invention (here, for example, sampling clock for two offset sampling gates) is
30 shown in Figure 9B. A difference between the two approaches is evident in the sense that in Figure 9B there can be seen a significant period of time between

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repetitions of the two sampling clock pulses during which there are no other sampling clock pulses. In Figures 9A and 9B, in order to simplify the comparison, the same 'time offset' as used for interleaving M off ADCs as used for Figure 9A has been used for Figure 9B.

5 Thus there is a fundamental difference between the effective sampling in the above described embodiments and the effective sampling of the M interleaved ADCs of the prior art arrangements. Accordingly, unlike the prior art, the above described embodiments can instrument an RF input signal that lies in the range of DC to $P \cdot 2 \cdot f_s / 2$ where P is greater than 1 and can still tend to
10 correctly resolve the frequency of the input signal. In modeling work carried out by the inventor a P of 7.5 has been achieved. This is significantly above the conventional 'Nyquist' limit which the above described prior art arrangements are capable of achieving.

 On first appearance this appears to defy the Nyquist rule that the
15 maximum bandwidth is half the sample rate. However, if one examines the effective sampling clock as used in the above embodiments, one further appreciates a surprising difference between the approaches. A high effective sampling rate is used i.e. the period between samples 1 and 2 is the same as for the prior art M interleaved ADC in Figure 9A. However the sampling is not
20 linear in time. It is the short period between samples 1 and 2 that gives the higher frequency resolution. However one cannot simply convert the interleaved samples from the offset sampling into the frequency domain and obtain the result one requires as one would for a corrected M interleaved ADC system. One has to process the data in a different manner as discussed for the above
25 embodiments. This is a significant surprising difference over the above mentioned prior art.

 As mentioned above in the "Background" section, in the earlier mentioned prior art arrangements described in WO 2008/039528 (Fang), the maximum frequency that the system can instrument is dictated by the frequency
30 response of the ADCs in use. In concept if the ADC had infinite bandwidth then the offset sampling approach would enable the arrangement to instrument as wide a bandwidth as those of the above described embodiments of the present invention. However, in practice, the prior art arrangements described in WO

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2008/039528 (Fang) can instrument only signals that pass through the ADC input bandwidth, whereas in the above described embodiments of the present invention signals between $f_s/2$ and the sampling gates' upper bandwidth limit (nominally defined by the sample gates pulse width) can typically be resolved in
 5 addition to the expected DC to $f_s/2$ frequency range of each ADC.

In the above described embodiments of the present invention, external sampling gates (i.e. the first sampler 4 and the second sampler 6) are used which have offset clocks. The ADCs have clocks with nominally identical timing. It is the offset sampling at these sampling gates that extends the frequency
 10 response to well above that of the ADCs. The two sampling functions have been shown in Figure 3 – i.e. Figure 3 shows sampling waveforms applied to sampling gates.

The frequency spectrum for a narrow pulse has been shown in Figure 4. The spectral lines are spaced $1/T$; i.e. $1/T = f_s$ so spectral lines occur at 0, f_s ,
 15 $2f_s$, $3f_s$ etc.

As the switch or gate is a non linear device the input signal is convolved with the above spectrum this will result in a mixer product due to the sampling waveform and input signal being present within the Nyquist bandwidth of the ADC. For example the inventor has used optical sampling techniques and
 20 achieved sampling pulse widths of the order of 31.25 picoseconds. This enables signals up to about 18 GHz to be mixed down into the Nyquist bandwidth of a 1GHz sample rate ADC.

The spectral content of Sampling_function(0) of Figure 3 is given by

25 1)
$$f(t) = \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j \cdot n \cdot \omega_o \cdot t}$$

The spectral content of Sampling_function(1) of Figure 3 is given by

2)
$$f_1(t) = \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j \cdot n \cdot \omega_o \cdot (t - t_{sep})} = \sum_{n=-\infty}^{\infty} \alpha_n \cdot e^{j(n \cdot \omega_o \cdot t)} \cdot e^{-j \left(\frac{n \cdot 2 \cdot \pi \cdot t_{sep}}{T} \right)}$$

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Thus there is a relative phase shift of

$$3) \quad \text{Phase\_shift\_spectral\_line}(n) = e^{-j\left(\frac{n \cdot 2 \cdot \pi \cdot t_{sep}}{T}\right)}$$

5 between Sampling_function(1) and Sampling_function(0) for spectral line n.

This phase difference is imposed on any common input RF signal, i.e. fsig, that is mixed down by spectral line (n) of the two sampling gates to appear in the ADCs' Nyquist bandwidths' DC to fs/2. Performing a time to frequency
 10 transformation on each ADC output is acceptable as each ADC is sampling at a sample rate of fs and can thus resolve signals from DC to fs/2. This mixed down signal is determined in the frequency domain as fsig_measured(0) at the output of ADC(0) and lies in the range DC to fs/2. For ADC(1) the measured frequency is fsig_measured(1). Determining the phase difference in the frequency domain
 15 between fsig_measured(0) and fsig_measured(1) enables the spectral line to be identified that was used to mix down the input signal fsig to fsig_measured(0). Knowing fsig_measured(0) and the spectral line used to generate fsig_measured(0) and fsig_measured(1), the original input signal fsig can be obtained, within certain constraints.

20 In the above described embodiments of the present invention, the clock to the two ADCs can be the same with no phase difference or time delay unlike in the arrangements described in WO 2008/039528 (Fang) where the time delay difference is imposed at the ADCs. In the above described embodiments of the present invention, the down conversion by the external gating means that input
 25 signals from a significant distance outside the normal frequency response of the ADC can be mixed down into the ADCs Nyquist bandwidth and digitised by the ADCs.

In practical terms the above described embodiments of the present invention can instrument a higher input signal than the arrangements described
 30 in WO 2008/039528 (Fang) because they are not limited by the bandwidth of the ADCs.

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Thus it will be appreciated that, in effect, in the above described embodiments of the present invention, narrow input gates (e.g. the first sampler 4 and the second sampler 6) are provided and arranged to mix the input signal down into the bandwidth of the ADCs 12, 14.

5 In the above embodiments, a sampling clock for only two offset sampling gates is used, as opposed to a complete set for M interleaved ADCs in the prior art arrangements, as can be seen by comparing Figure 9B to prior art Figure 9A. However, the particular choice in the above described embodiments of the value of two i.e. “a sampling clock for only two offset sampling gates” is not
10 essential, and in other embodiments values other than two may be used, provided that the value chosen does not “fill up” the whole span with clock pulses – i.e. provided the value chosen still provides substantially non-even sampling. In other words, in other embodiments, other values may be used, provided external sampling gates are used which have offset clocks.

15

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CLAIMS

1. A method of performing sampling, the method comprising:
sampling an input signal (16) using a first sampling signal (22) to provide
5 a first sampled signal (26);
sampling the input signal (16) using a second sampling signal (24) to
provide a second sampled signal (30);
measuring a frequency of at least one of the first sampled signal (26) and
the second sampled signal (30);
10 measuring a phase difference between the first sampled signal (26) and
the second sampled signal (30); and
determining a frequency of the input signal (16) using the measured
frequency and the measured phase difference; wherein
the second sampling signal (24) has substantially equal frequency to the
15 first sampling signal (22); and
a waveform of the second sampling signal (24) is shifted in time with
respect to a waveform of the first sampling signal (22).
2. A method according to claim 1, wherein the steps of sampling an input
20 signal (16) using a first sampling signal (22) to provide a first sampled signal
(26) and sampling the input signal (16) using a second sampling signal (24) to
provide a second sampled signal (30) comprises using external sampling gates
that have offset clocks.
- 25 3. A method according to claim 1 or claim 2, wherein the first sampled
signal (26) and the second sampled signal (30) are the only two sampled
signals provided.

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4. A method according to any of claims 1 to 3, wherein narrow input gates (4, 6) are arranged to mix the input signal (16) down into the bandwidth of ADCs (12, 14).
- 5 5. A method according to any of claims 1 to 4, wherein the measured frequency of at least one of the first sampled signal (26) and the second sampled signal (30) is a measured alias frequency of the true frequency of at least one of the first sampled signal (26) and the second sampled signal (30).
- 10 6. A method according to any of claims 1 to 5, wherein the waveform of the second sampling signal (24) is substantially equal to the waveform of the first sampling signal (22).
- 15 7. A method according to any of claims 1 to 6, further comprising the step of determining the amplitude of the input signal (16) using the waveforms and magnitudes of the sampling signals (22, 24), values of the frequency of the waveforms of the sampling signals (22, 24), and the determined input signal frequency.
- 20 8. A method according to any of claims 1 to 7, wherein the step of measuring a frequency of at least one of the first sampled signal (26) and the second sampled signal (30) is performed within a Nyquist zone of an analogue-to-digital converter (12, 14).
- 25 9. A method according to any of claims 1 to 8, wherein the step of determining a frequency of the input signal (16) further comprises using a value of a frequency of a sampling signal (22, 24), a value of the time shift that the second sampling signal (24) is shifted in time relative to the first sampling signal (22), and a value of the time period of a sampling signal (22, 24).

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10. A method according to claim 9, wherein the frequency of the input signal (16) is determined using the following formula:

$$f_{\text{sig}} = -1 * \text{Spectral\_index} * f_s + \text{Measured\_frequency}$$

where: f_{sig} is the frequency of the input signal (16);

f_s is the frequency of a sampling signal (22, 24);

Measured_frequency is a measured value of a frequency of at least one of the first sampled signal (26) and the second sampled signal (30); and

$$\text{Spectral\_index} = \frac{\text{Phase\_difference}}{\text{Phase\_step\_for\_first\_indexed\_spectral\_line}}$$

where: Phase_difference is the measured phase difference; and

Phase_step_for_first_indexed_spectral_line is a value of a phase step of a first spectral line in a frequency spectrum of the sampling signals (22, 24).

11. A method according to claim 9, wherein the frequency of the input signal (16) is determined using the following formula:

$$f_{\text{sig}} = -1 * \text{Spectral\_index} * f_s + \text{Measured\_frequency}$$

where: f_{sig} is the frequency of the input signal (16);

f_s is the frequency of a sampling signal (22, 24);

- 30 -

Measured_frequency is a measured value of a frequency of at least one of the first sampled signal (26) and the second sampled signal (30); and

$$\text{Spectral\_index} = \text{Round} \left[\frac{\text{Phase\_difference}}{\text{Phase\_step\_for\_first\_indexed\_spectral\_line}} \right]$$

where: Phase_difference is the measured phase difference; and

Phase_step_for_first_indexed_spectral_line is a value of a phase step of a first spectral line in a frequency spectrum of the sampling signals (22, 24); and

Round[X] is the value of X rounded to the nearest integer.

12. A method according to claim 10 or 11, wherein:

$$\text{Phase\_step\_for\_first\_indexed\_spectral\_line} = -2 \cdot \pi \cdot \frac{t_{sep}}{T}$$

where: t_{sep} is a value of the time shift that the waveform of the second sampling signal (24) is shifted by with respect to the waveform of the first sampling signal (22); and

T is a value of a time period of a sampling signal (22, 24).

13. A method according to any of claims 1 to 12, wherein the step of measuring a frequency of at least one of the first sampled signal (26) and the second sampled signal (30) comprises:

filtering the first sampled signal (26) such that it lies within a Nyquist zone of a first analogue-to-digital converter (12);

using the first analogue-to-digital converter (12), converting to a discrete signal the filtered first sampled signal (28);

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filtering the second sampled signal (30) such that it lies within a Nyquist zone of a second analogue-to-digital converter (14);

using the second analogue-to-digital converter (14), converting to a discrete signal the filtered second sampled signal (32); and

- 5 measuring a value of the frequency of signal converted by the first analogue-to-digital converter (12) and/or a value of the frequency of signal converted by the second analogue-to-digital converter (14).

14. An apparatus for performing sampling, the apparatus comprising:

- 10 a first sampler (4) adapted to sample an input signal (16) using a first sampling signal (22) to provide a first sampled signal (26);

 a second sampler (6) adapted to sample the input signal (16) using a second sampling signal (24) to provide a second sampled signal (30);

- 15 means for measuring a frequency of at least one of the first sampled signal (26) and the second sampled signal (30);

 means for measuring a phase difference between the first sampled signal (26) and the second sampled signal (30); and

 a processor (15) adapted to determine a frequency of the input signal (16) using the measured frequency and the measured phase difference;

- 20 wherein

 the second sampling signal (24) has substantially equal frequency to the first sampling signal (22); and

 a waveform of the second sampling signal (24) is shifted in time with respect to a waveform of the first sampling signal (22).

25

15. A computer program or plurality of computer programs arranged such that when executed by a computer system it/they cause the computer system to operate in accordance with the method of any of claims 1 to 13.

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16. A machine readable storage medium storing a computer program or at least one of the plurality of computer programs according to claim 15.

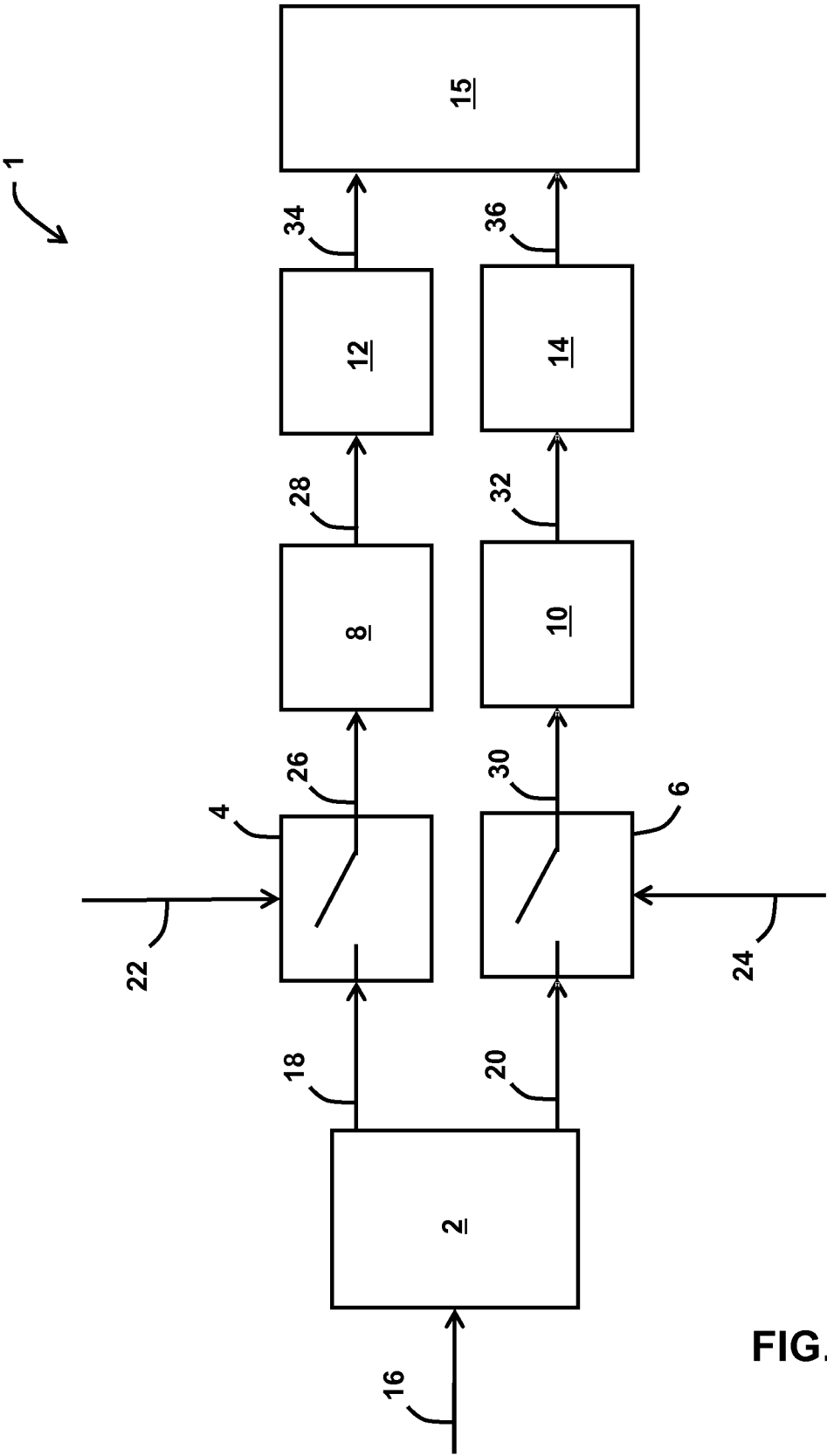
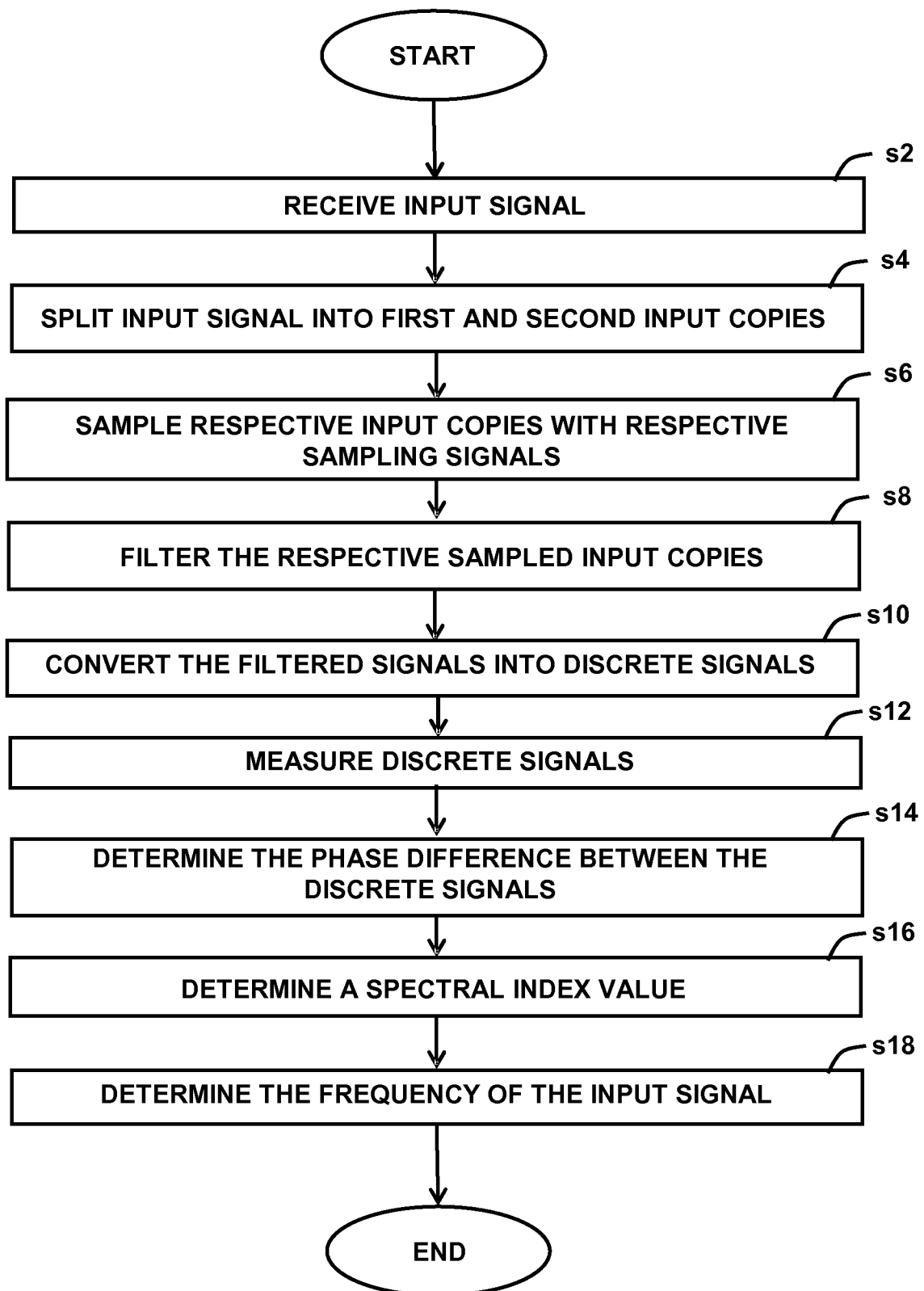


FIG. 1

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**FIG. 2**

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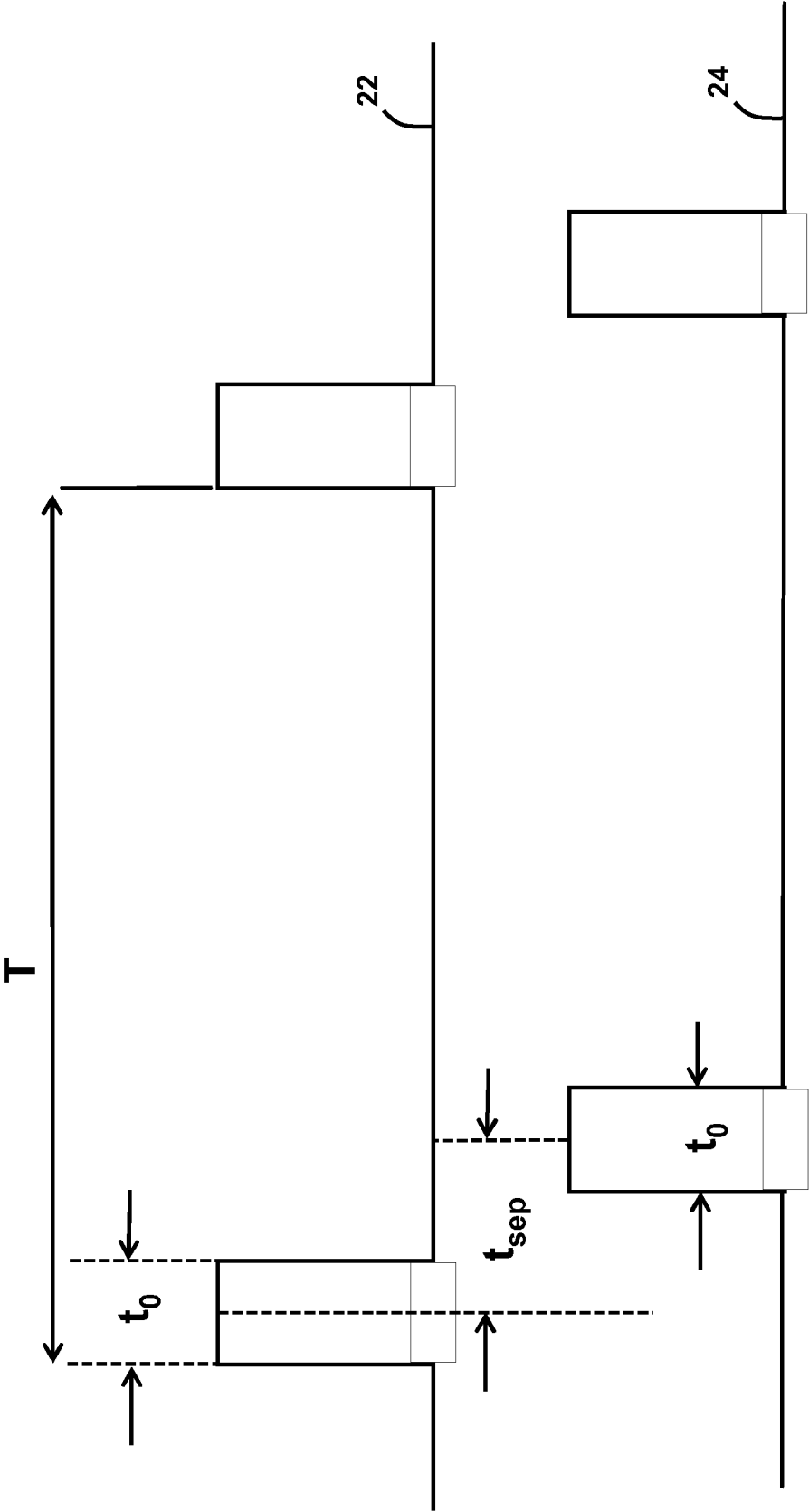


FIG. 3

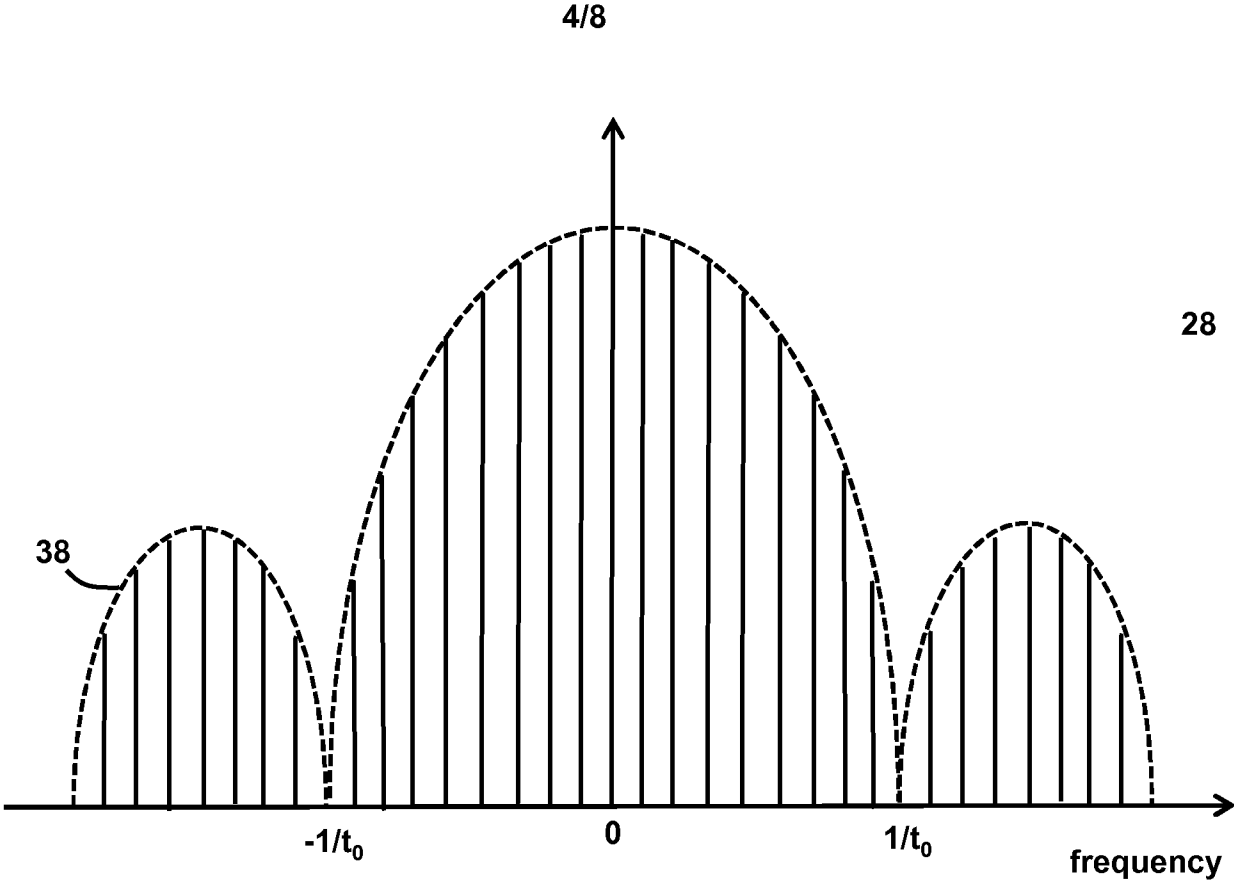


FIG. 4

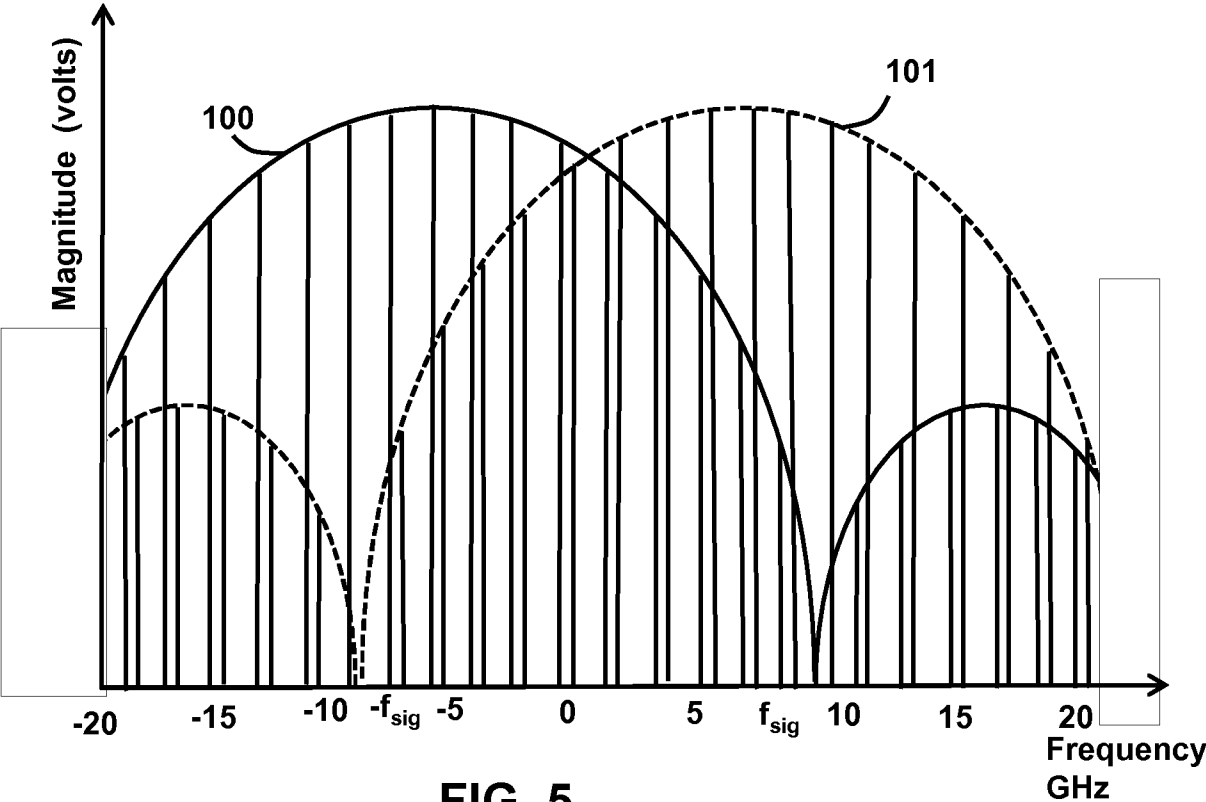


FIG. 5

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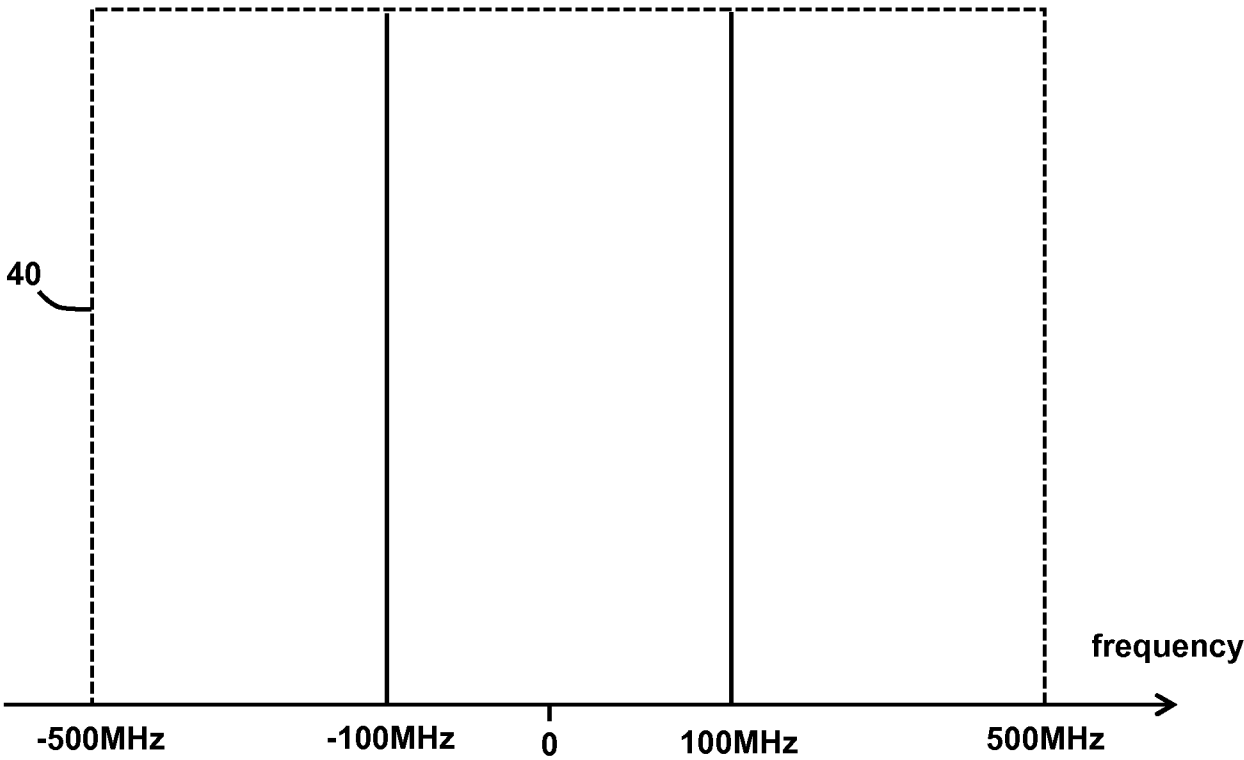
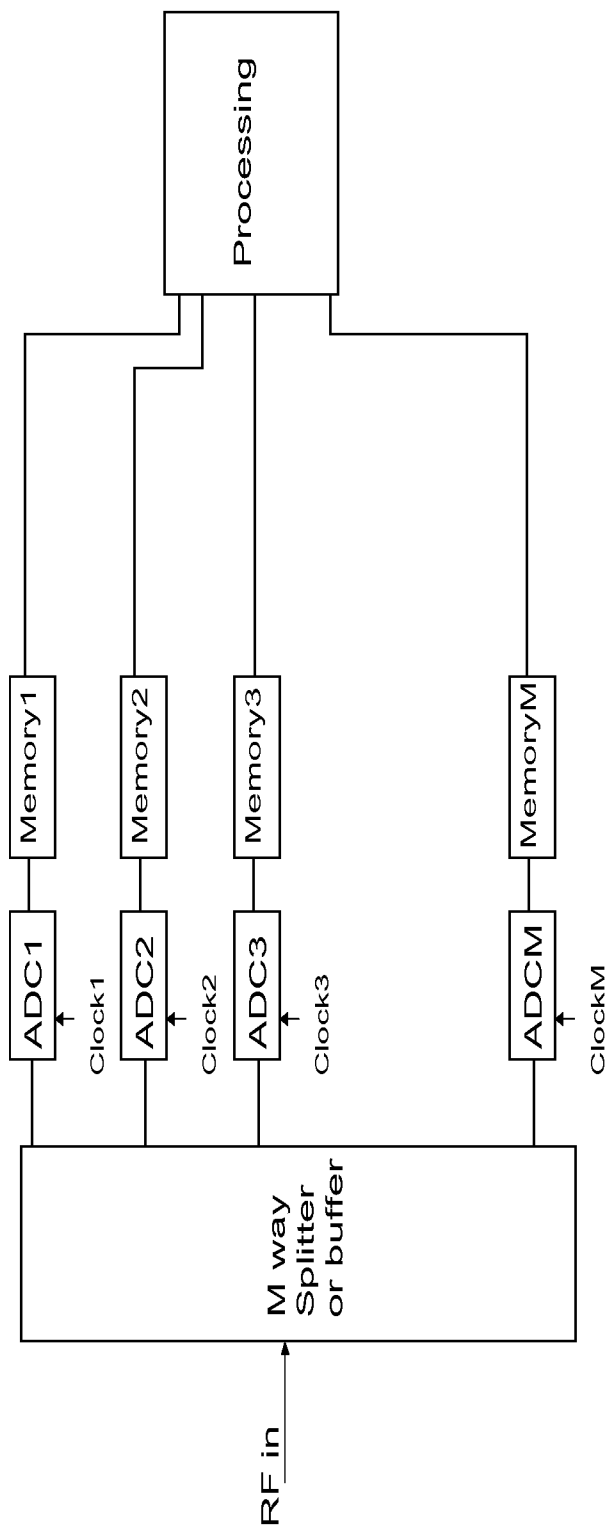
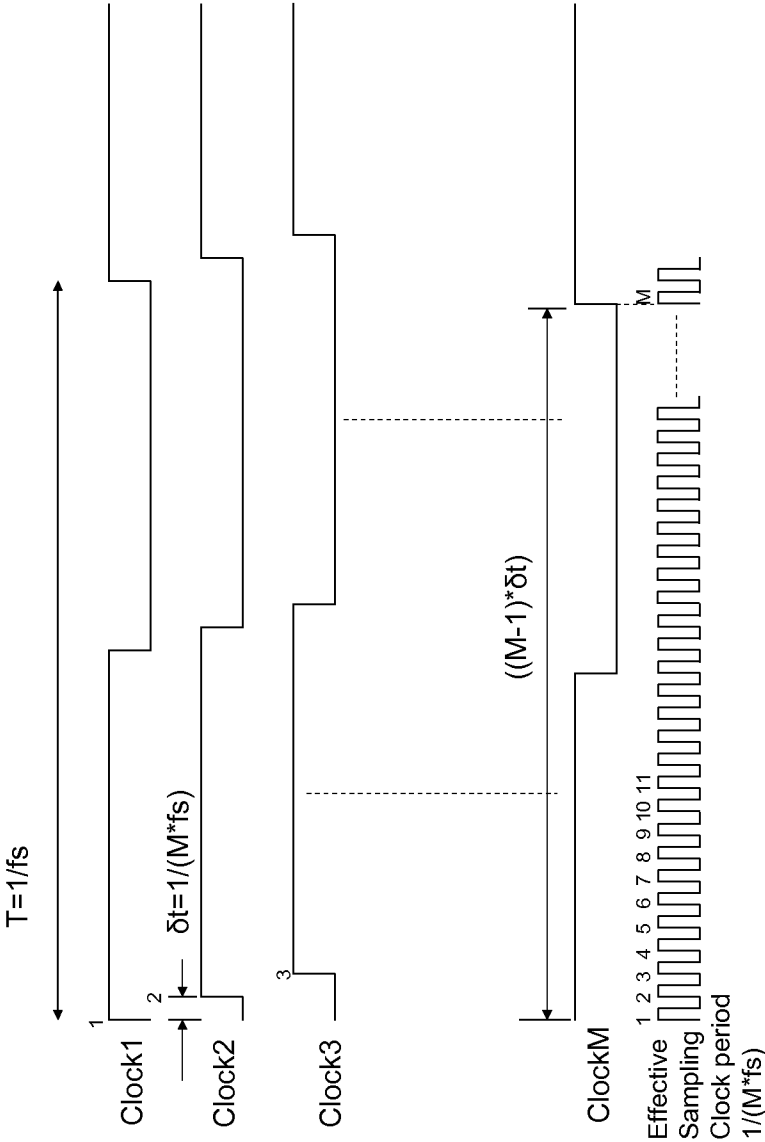


FIG. 6



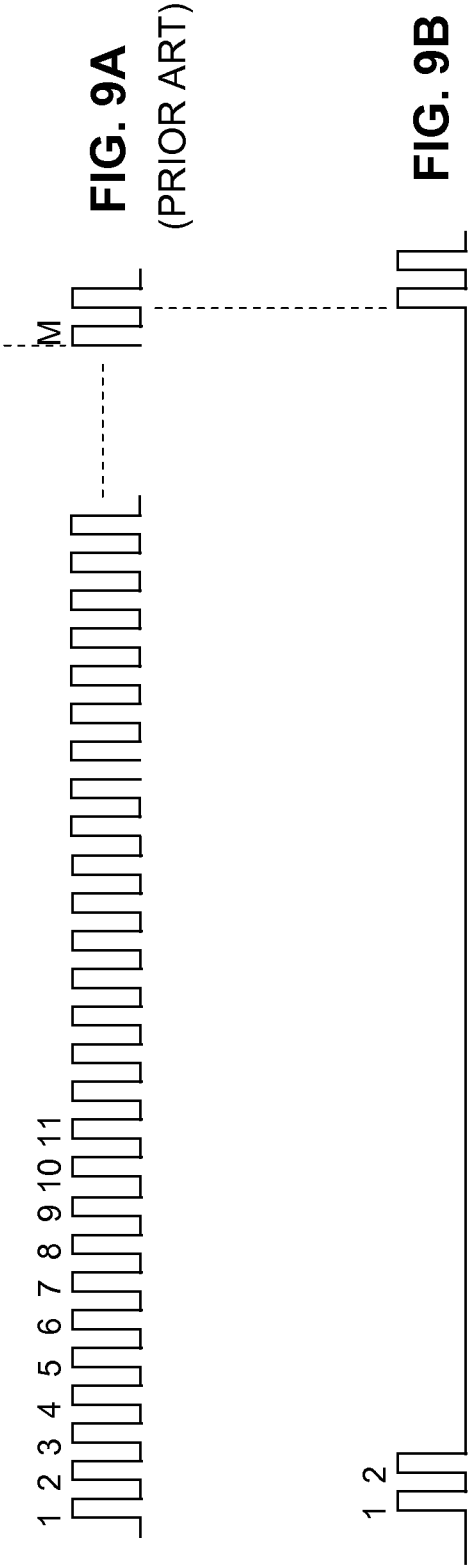
PRIOR ART

FIG. 7



PRIOR ART

FIG. 8



INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2011/050418

A. CLASSIFICATION OF SUBJECT MATTER
INV. H03M1/12 G01R23/00
ADD.

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

H03M G01R H04B H03D

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EP0-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5 099 243 A (TSUI JAMES B Y [US] ET AL) 24 March 1992 (1992-03-24) column 1, lines 28-31; figures 1, 2 column 1, line 62 - column 2, line 3 column 2, line 22 - column 3, line 34 -----	1-16
X	US 2009/027250 A1 (NARA AKIRA [JP]) 29 January 2009 (2009-01-29) page 1, paragraph 8-11; figure 2 -----	1-16
X	US 7 206 707 B1 (TSUI JAMES B Y [US] ET AL) 17 April 2007 (2007-04-17) * abstract; figures 2, 10, 11 column 4, line 62 - column 6, line 2 column 8, line 61 - column 9, line 48 ----- -/--	1-16

☒ Further documents are listed in the continuation of Box C.

☒ See patent family annex.

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Date of the actual completion of the international search

31 March 2011

Date of mailing of the international search report

20/04/2011

Name and mailing address of the ISA/

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NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040,
Fax: (+31-70) 340-3016

Authorized officer

Brosa, Anna-Maria

INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2011/050418

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
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