



APPARATUS AND METHOD FOR DEMODULATING DATA

The present invention relates to the demodulation of digital data modulated using frequency shift keying or phase shift keying. It is particularly applied to the demodulation of a digital bitstream coded and transmitted at a high data rate.

In Frequency Modulation (FM), data is used to vary the frequency of a carrier signal. The carrier signal has a given nominal frequency and this is altered (either increased or decreased) by the data signal. The characteristic equation of a FM signal is given by:

$$y(t) = A \cos[\omega_{IF}t + mf(t)] \quad (1)$$

where, $y(t)$ is the FM signal, ω_{IF} is an intermediate frequency (IF) of the FM carrier signal in radians/sec, $f(t)$ is the modulating (data) signal, m is the modulation index and A is the carrier amplitude. For radio transmission, the FM signal is then up-converted to a higher frequency for transmission from an antenna. For conventional audio FM radio signals, a frequency of approximately 100 MHz is used. To recover the original signal $f(t)$, the up-converted signal, $y'(t)$, is received through an antenna, down-converted to give $y(t)$ and then multiplied by $\sin(\omega_{IF}t)$ to give a signal $g(t)$ according to the following equation:

$$g(t) = \sin(\omega_{IF}t) A \cos[\omega_{IF}t + mf(t)] \quad (2)$$

where $\sin(\omega_{IF}t)$ is a locally generated IF equal to the IF used in the modulation of $f(t)$, used to separate the modulated signal from the nominal FM carrier. Expanding equation (2) gives:

$$\begin{aligned} g(t) &= A \sin(\omega_{IF}t) \{ \cos(\omega_{IF}t) \cos[mf(t)] - \sin(\omega_{IF}t) \sin[f(t)] \} \\ &= \frac{1}{2} A \{ \sin[2 \omega_{IF}t + mf(t)] - \sin[mf(t)] \} \end{aligned} \quad (3)$$



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Generally, $\omega_{IF}t$ is large compared to the frequency of $mf(t)$ so a low-pass filter can be used to remove the high frequency term in $g(t)$, leaving:

$$g'(t) = -\frac{1}{2} A \sin[mf(t)] \quad (4)$$

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In cases where $mf(t)$ has small amplitude, $\sin[mf(t)]$ is approximately $mf(t)$ which leads us to:

$$g'(t) = -\frac{1}{2} A mf(t) \quad (5)$$

- 10 Therefore an FM signal, in which $\omega_{IF}t$ is large compared to the frequency of $f(t)$ and $mf(t)$ has small amplitude, can be demodulated by (i) multiplying the received signal by the same IF used in the modulation process and then, (ii) passing it through a low pass filter.

- The transmission rate of the data may be increased without adversely
15 affecting the bandwidth by modulating both the in-phase and quadrature components of the system. Data is transmitted on both the components allowing the transmission of twice as much data in the same bandwidth.

- It is known to transmit digital data using such an FM system. The digital data is first expressed as a digital bit stream representing the bytes and words of the
20 original data. A digital bit stream is a series of discrete values, unlike an analogue signal which can vary continuously. Therefore, examining equation (1) above shows that the modulated signal will switch between defined frequency values. A system such as this is known as Frequency Shift Keying (FSK).

- An example of a simple FSK demodulator would consist of two narrow band-
25 pass filters. One filter is tuned to the frequency of a "0" bit and the other tuned to the frequency of a "1" bit. The output of the two filters is compared; the one with the highest amplitude is selected as the current bit value of the signal.

- To allow the signal to be transmitted efficiently over a channel which cannot transmit DC signal energy well, for example a radio channel, the digital bit stream is
30 often encoded to reduce the DC content of the signal to as close to zero as possible. Various techniques for this are known, such as Manchester coding or Miller coding.

After coding, the data signal is filtered to remove the square edges and reduce the harmonics present, consequently reducing the bandwidth required for transmission. This filtered data signal is then fed into a standard FM Modulator. The transmitted signal no longer contains two well-defined frequencies so the harmonics associated with a step change in frequency are eliminated.

FSK demodulators implemented on programmable Digital Signal Processors (DSPs) are known. However the implementation is complicated and requires many DSP instruction cycles. This requires the use of fast DSPs with a consequential increase in power consumption and cost.

It is an aim of the present invention to provide an improved demodulator for FSK or PSK. It is a further aim to minimise the cost of production of a demodulator for FSK or PSK. A still further aim is to reduce the power consumption of an FSK or PSK demodulator. A yet further aim is to increase the transmission rate of a communication system without adversely affecting the signal bandwidth.

According to a first aspect of the present invention there is provided a demodulator for demodulating a received signal representing an original binary bit stream, the original binary bit stream being modulated using phase shift keying or frequency shift keying, the demodulator comprising: analog to digital conversion means for converting said received signal into a digital signal; a demodulation finite impulse response filter, having said digital signal as an input and an impulse response corresponding to the transfer function from said digital signal to the original data signal, for filtering said digital signal and for outputting a signal corresponding the original bit stream.

By using a demodulation filter operating in the digital domain, the present invention allows the signal to be demodulated in a single clock cycle, reducing the power consumption. Furthermore the implementation is relatively simple, because the complexity is transferred to the design stage. The demodulation filter can be created easily and cheaply on an integrated circuit.

Preferably, if the demodulator is applied to a signal modulated using frequency shift keying, the demodulator may further comprise intermediate

frequency multiplication means, having an input of said digital signal, for multiplying said digital signal by a sine wave, or a digital approximation to a sine wave, and outputting a multiplied signal, said sine wave having a frequency approximately equal to the intermediate frequency of the received signal; and a low-pass finite impulse response filter, having an input of said multiplied signal, and having a low-pass characteristic for filtering said multiplied signal and for outputting a low-pass filtered signal; wherein, said demodulation finite impulse response filter is for filtering said low-pass filtered signal, said demodulation filter having an impulse response corresponding to the transfer function from said low-pass filtered signal to the original data signal.

In this case the demodulator is constructed from simple and cheap additional components which are well suited to manufacture as integrated circuits. Furthermore, all the components may be designed to operate in a single clock cycle.

In one preferable alternative, when the demodulator is applied to a signal modulated using frequency shift keying, the demodulator may further comprise intermediate frequency multiplication means, having an input of said received signal, for multiplying said received signal by a sine wave in the analog domain, and outputting a multiplied signal to said analog to digital conversion means, said sine wave having a frequency approximately equal to the intermediate frequency of the received signal; and a low-pass finite impulse response filter, having an input of said digital signal, and having a low-pass characteristic for filtering said multiplied signal and for outputting a low-pass filtered signal; wherein, said demodulation finite impulse response filter is for filtering said low-pass filtered signal, said demodulation filter having an impulse response corresponding to the transfer function from said low-pass filtered signal to the original data signal.

Advantageously, the original binary bit stream may be filtered so that it has the shape of a cosine wave.

The design of the demodulation finite impulse response filter is then simplified because it will have real coefficients.

The demodulator may be further simplified by combining the action of the low-pass and demodulation FIR filters in a single FIR filter. This will give a

corresponding reduction in the cost of implementing the demodulator.

Advantageously, the modulation index of the received signal can take at least two values, said at least two values when taken in combination with the transmitted bit, allowing the transmission of more than one symbol per transmitted bit.

5 Varying the modulation index has only a marginal effect on the bandwidth required. Increasing the modulation index will result in a small increase in bandwidth. However, decreasing the modulation index will give a small reduction in bandwidth. Therefore, there is the potential to transmit at a higher data rate without adversely affecting the bandwidth.

10 According to a second aspect of the present invention, there is provided a method of demodulating a received signal representing an original binary bit stream, the original binary bit stream being modulated using phase shift keying or frequency shift keying, the method comprising the steps of: converting the received signal into a digital signal; and demodulation filtering said digital signal with a demodulation
15 filter designed to have an impulse response that corresponds to the transfer function from said digital signal to said original binary bit stream, to generate a demodulated signal which corresponds to the original bit stream.

A preferred embodiment of the invention will now be described, by way of
20 example only, with reference to the accompanying drawings in which:

Figure 1 is a schematic of a FSK demodulator according to a first embodiment of the present invention; and

Figure 2 is a schematic of a FSK demodulator according to a second embodiment of the present invention.

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Like parts are denoted by like references.

The theory underlying the present invention will be now be explained; a detailed description of the preferred embodiments will be given later.

30 In order to design an improved FSK or PSK demodulator it is necessary to consider the characteristic equations of the transmitted data and also the desired

output from the demodulator. The discussion below assumes the use of data coded using NRZ coding. However, the skilled person will appreciate that the principles may be applied to data that is not coded using an NRZ code. The discussion will be given in terms of a signal modulated using FSK, however, the person skilled in the art will appreciate that the principles may also be applied to demodulating data modulated using PSK.

If the impulse response that will transform an FSK signal consisting of only one pair of a one and zero encoded in an NRZ format can be calculated, it can be used to demodulate any arbitrary FSK signal consisting of random NRZ formatted data with the same modulation index.

In the discussion above it was mentioned that the digital data is filtered before being transmitted. Various filters may be used, for example the wave may be shaped into a sine wave, or approximately sine wave shape. Filtering the data to have the form of a cosine function will give the transmitted data minimal transmitted bandwidth. Assuming this is the case then it can be assumed that the modulating signal $f(t)$ is a cosine wave. With reference to equation (4), the output, $g'(t)$, after multiplying and low-pass filtering the received signal will be:

$$g'(t) = -\frac{1}{2} A \sin[m \cos(\omega_m t)] \quad (4a)$$

where A is the carrier amplitude (scaling factor) and ω_m is the modulating frequency of the cosine wave representing the data. The modulation index m therefore represents the amplitude of this wave representing the data. Typically a large value for m is chosen to improve the signal-to-noise ratio (SNR) of the communication system.

We can demodulate this signal by considering what the output, $x(t)$, from the demodulator should be. For data assumed to be represented by a cosine wave this gives:

$$x(t) = m \cos(\omega_m t) \quad (5a)$$

The operation of the demodulator can therefore be described in the frequency domain as the transfer function from the received signal, $g'(t)$, to the output, $x(t)$. This function can be calculated by taking the Fourier transform of equation (4a) and equation (5a). Transformed equation (5a) is then divided by transformed equation 5 (4a) to give the transfer function, expressed in the frequency domain. The impulse response of the system is then derived by taking the inverse Fourier transform of the transfer function.

A FIR filter can then be designed which has the correct impulse response to demodulate a received signal. In this case, the assumption that the modulating signal 10 was a cosine wave means the FIR filter has only real coefficients. A signal $g'(t)$ can then be demodulated simply by sampling it at a given frequency and feeding it into the FIR filter. The output from this filter is the signal $x(t)$ sampled at the clock frequency of the FIR filter.

It will be appreciated that the coefficients of the FIR filter will in general 15 depend on the sampling frequency, transmitted data rate, modulation index m and the amplitude A . Therefore, in general the filter coefficients will be set for a given communication system. This fact can be used to further increase the transmission rate in a given system.

If a filter is designed to demodulate a signal with a known value of m then the 20 output, given an input $x(t)$ modulated with that value of m , is $x(t)$ as expected. However, if an input $x'(t)$ modulated with a different value of m is input, the output is not $x'(t)$. The shape of the wave is the same, but the amplitude differs from $x'(t)$. The amplitude differs by a constant factor determined by the difference between the modulation index of $x'(t)$ and the modulation index that the filter was designed for. 25 The value of this constant changes with a sine-related function as the modulation index of the input is varied, assuming the same demodulation filter is used.

This allows the transmission of more than one bit per transmitted symbol by using additional encoding which varies the modulation index. For example, a system could use an FIR filter designed for $m=3$. When data modulated with $m=3$ is 30 demodulated the output from the filter will be $x(t)$. If data modulated with $m=2$ is demodulated the output from the filter the have the correct shape but will have a

higher amplitude. If data modulated with $m=4$ is demodulated the output from the filter will have the correct shape but with a lower amplitude. A binary system using this principle might take values of $+m$ and $-m$ while a quaternary system might use values of $+m$, $+0.5m$, $-0.5m$ and $-m$.

5 It will be appreciated that this system could be applied to both the in-phase and quadrature components of the transmitted signal by simply providing a second FSK FIR demodulator for the quadrature channel and a quadrature mixer. Likewise, the multi-level property created by the controlled variation of the modulation index can be extended to such a system, thus realising a system similar to Quadrature
10 Amplitude Modulation (QAM). Furthermore, when extended in this way, the system of the present invention will have a higher immunity to noise than QAM because it will inherit the signal-to-noise ratio benefits of an FM over an AM system. The system will also benefit from all the other advantages of an FM system over an AM system.

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An embodiment of a FSK demodulator for radio signals according to the present invention is illustrated in Figure 1. A transmitted signal is received through the aerial 2, is amplified by the amplifier 4 before being passed to a down conversion circuit 6. This down-converts the signal to remove the transmission frequency.

20 The signal then passes through an IF carrier multiplier 8 which multiplies the down-converted signal to give a signal according to Equation (2). The signal from the IF carrier multiplier is then converted into a digital signal by an Analogue-to-Digital Convertor (ADC) 10. An oversampler 11 over samples the output from the ADC 10 at the operating frequency of clock 18. Next, the output from the
25 oversampler 11 is passed through an FIR filter 12 having a low pass characteristic to filter out the IF component. The output from FIR filter then represents Equation (4a) sampled at the clock frequency

The next component is the FIR demodulator 14. This is designed to have an impulse response corresponding to the transfer function from the output of the low-pass filter 12 to the demodulated output as described above, with the knowledge of
30 the transmission parameters of the modulated signal. The output from the FIR

demodulator 14 thus represents the demodulated signal, sampled at the clock frequency. Finally the decoder 16 decodes the signal $x(t)$ and outputs a pure digital bit stream corresponding to the original transmitted data.

The decoder 16 decodes the signal according to the scheme used, for example Manchester coding. If the coding of the received signal has been extended to include varying the modulation index to transmit more than one symbol per transmitted bit, the decoder 16 will also interpret this data. The decoder 16 may be omitted if the data has not been coded, but in most practical situations coding will be used to improve performance.

10 An optional integrator 17 and DAC 19, are also shown in Figure 1. These act to improve the demodulator performance by correcting any small errors between the frequency of the received signal and the local oscillator frequency in the demodulator.

The implementation may be further simplified by combining the FIR demodulator 14 and low-pass FIR filter 12 into a single FIR filter.

A second embodiment of the invention is illustrated in Figure 2. In this embodiment, the IF carrier multiplication takes place in the digital domain. The construction of the demodulator is the same as the above first embodiment, except as described below.

20 After the received signal has been down-converted, it passes through an analog anti-aliasing filter 20. A 1-bit ADC 22, for example a comparator, is then used to convert the analog signal into a digital signal. An oversampler 23 then oversamples the output of the ADC 22 at the frequency of the clock 18.

The IF carrier multiplier 24 carries out signal multiplication in the digital domain. In the present embodiment this is done by calculating the exclusive-OR function of the output from the ADC with an approximated sine wave of the correct frequency. The sine wave approximation has the form $[1,0,1,0,\dots]$ in return-to-zero format and $[1,0,-1,0,1,0,-1,0,\dots]$ in non-return-to-zero format. Therefore, the construction of the IF carrier multiplier 24 is simplified, and the number of analog components required is reduced.

The output from the IF carrier multiplier 24 is then passed directly to the

Low-pass FIR filter 12 and demodulation proceeds in the same manner as for the above first embodiment.

The above first and second embodiments have not described extending the modulation to include both in-phase and quadrature components. This can be
5 achieved by applying the demodulator described to the in-phase signal and simply providing a second demodulator as described above for the demodulation of the received quadrature signal.

The skilled person will appreciate that the above first and second
embodiments may be applied to Zero Intermediate Frequency (ZIF) demodulation
10 with appropriate modifications.

According to the present invention there is provided an FSK demodulator which gives optimal performance with a cheap cost of manufacture. The demodulation is provided entirely by FIR filters, so demodulation can be achieved in
15 one clock cycle, and thus is ideally suited to applications with low power consumption requirements. The present invention is therefore advantageously applied to portable and low-power FSK applications. Additionally, since all operations can be carried out in one clock cycle, the present invention is also advantageously applied to communication systems utilising a high transmission rate,
20 such as broadband modems.

Whilst the above embodiments have been applied to radio communication systems it will be appreciated that the present invention could be applied to any communication system using FSK, for example modems.
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Whilst an embodiment of the invention has been described above, it will be appreciated that this description does not limit the scope of the invention, which is defined in the appended claims.

CLAIMS

1. A demodulator for demodulating a received signal representing an original binary bit stream, the original binary bit stream being modulated using phase shift keying or
5 frequency shift keying, the demodulator comprising:

analog to digital conversion means for converting said received signal into a digital signal;

- a demodulation finite impulse response filter, having said digital signal as an input and an impulse response corresponding to the transfer function from said
10 digital signal to the original data signal, for filtering said digital signal and for outputting a signal corresponding the original bit stream.

2. A demodulator according to claim 1, wherein said original binary bit stream has been modulated using frequency shift keying, the demodulator further comprising:

- 15 intermediate frequency multiplication means, having an input of said digital signal, for multiplying said digital signal by a sine wave, or a digital approximation to a sine wave, and outputting a multiplied signal, said sine wave having a frequency approximately equal to the intermediate frequency of the received signal; and

- a low-pass finite impulse response filter, having an input of said multiplied
20 signal, and having a low-pass characteristic for filtering said multiplied signal and for outputting a low-pass filtered signal; wherein,

said demodulation finite impulse response filter is for filtering said low-pass filtered signal, said demodulation filter having an impulse response corresponding to the transfer function from said low-pass filtered signal to the original data signal.

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3. A demodulator according to claim 1, wherein said original binary bit stream has been modulated using frequency shift keying, the demodulator further comprising:

- intermediate frequency multiplication means, having an input of said received signal, for multiplying said received signal by a sine wave in the analog domain, and
30 outputting a multiplied signal to said analog to digital conversion means, said sine wave having a frequency approximately equal to the intermediate frequency of the

received signal; and

a low-pass finite impulse response filter, having an input of said digital signal, and having a low-pass characteristic for filtering said multiplied signal and for outputting a low-pass filtered signal; wherein,

5 said demodulation finite impulse response filter is for filtering said low-pass filtered signal, said demodulation filter having an impulse response corresponding to the transfer function from said low-pass filtered signal to the original data signal.

4. A demodulator according to any one of the preceding claims wherein the binary bit
10 stream is filtered so that it has the shape of a cosine wave.

5. A demodulator according to claim 1 or claim 2, wherein the characteristics of said low-pass and said demodulation finite impulse response filters are combined and implemented in a single finite impulse response filter.

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6. A demodulator according to any one of the preceding claims wherein the modulation index of the received signal can take at least two values, said at least two values each resulting in a different amplitude in said output of said demodulation filter, allowing the transmission of more than one symbol per transmitted bit by
20 encoding additional symbols through the variation of said modulation index.

7. A demodulator according to any one of the preceding claims wherein said received signal is modulated using frequency shift keying on both the in-phase and quadrature components and additionally comprising a second low-pass filter and a second
25 demodulation filter for demodulating the quadrature component.

8. A demodulator according to claim 7 in which said second low-pass and said second demodulation filters are combined and implemented in a single filter.

30 9. A demodulator according to any one of the preceding claims formed as an integrated circuit.

10. A modem including the demodulator of any preceding claim.

11. A method of demodulating a received signal representing an original binary bit stream, the original binary bit stream being modulated using phase shift keying or

5 frequency shift keying, the method comprising the steps of:

converting the received signal into a digital signal;

demodulation filtering said digital signal with a demodulation filter designed to have an impulse response that corresponds to the transfer function from said digital signal to said original binary bit stream, to generate a demodulated signal

10 which corresponds to the original bit stream.

12. A method according to claim 11 wherein the original binary bit stream is modulated using frequency shift keying and further including the steps of:

15 multiplying said digital signal by a sine wave, or a digital approximation of a sine wave, said sine wave having a frequency approximately equal to the intermediate frequency of said received signal, to generate a multiplied signal;

low-pass filtering said multiplied signal with a finite impulse response filter having a low-pass characteristic, to generate a low-pass filtered signal; and wherein,

20 said step of demodulation filtering is applied to said low-pass filtered signal, said demodulation filter having an impulse response corresponding to the transfer function from said low-pass filtered signal to said original binary bit stream.

13. A method according to claim 11 wherein the original binary bit stream is modulated using frequency shift keying and further including the steps of:

25 multiplying said received signal by a sine wave in the analog domain prior to said step of converting, said sine wave having a frequency approximately equal to the intermediate frequency of said received signal, to generate a multiplied signal;

low-pass filtering said digital signal with a finite impulse response filter having a low-pass characteristic, to generate a low-pass filtered signal; and wherein,
30 said step of demodulation filtering is applied to said low-pass filtered signal, said demodulation filter having an impulse response corresponding to the transfer

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function from said low-pass filtered signal to said original binary bit stream.

14. A method according to any one of claims 11 to 13, wherein the original binary bit stream is modulated using frequency shift keying and the modulation index can take
5 at least two defined values, further comprising the step of deducing the value of the modulation index from the amplitude of the output of said demodulation filter.