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APPARATUS AND METHOD FOR MODULATING DIGITAL DATA

The present invention relates to the modulation of digital data using frequency shift keying or phase shift keying. It is particularly applied to the modulation of a digital bit stream 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:

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

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:

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

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:

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

30 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:

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

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

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

- Therefore an FM signal, in which ω_{IFT} 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 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.

The SNR of a transmitted signal may be improved by applying a function, for example a logarithm function, to the transmitted signal. This can be used to alter the characteristics of the transmitted signal and improve the rejection of additive noise from the channel when the signal is demodulated.

- It is known to transmit digital data using such an FM system. The digital data is first expressed as a bit stream representing the bytes and words of the original data. This bit stream is a series of discrete values, unlike an analog 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).

- A method of FSK modulation known in the art is to send the digital bit stream directly to a voltage controlled oscillator (VCO), for example a varactor diode in a tank circuit. As the digital bit stream has two defined values the VCO will output two frequencies, one of which corresponds to a "1" value and the other of which corresponds to a "0" value.

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 often encoded to reduce the DC content of the signal to as close to zero as possible. Various techniques are known such as Manchester coding or Miller coding.

5 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. Thus the transmitted signal no longer switches abruptly between two well-defined frequencies so the harmonics associated with a step change in frequency are eliminated.

) 10 FSK modulators 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 modulator for PSK or
15 FSK modulation. It is a further aim to minimise the cost of production of a modulator for PSK or FSK modulation. A still further aim is to reduce the power consumption of an FSK or PSK modulator. A yet further aim is to increase the transmission rate of a communication system using FSK on the transmission channel without adversely affecting the signal bandwidth. A further aim is to eliminate the need for a varactor diode in an FSK
) 20 modulator.

According to a first aspect of the present invention there is provided a modulator for modulating a digital bit stream using phase shift keying or frequency shift keying, the modulator comprising: a modulation finite impulse response filter having an impulse
25 response corresponding to the transfer function from the original bit stream to the bit stream modulated using frequency shift keying or phase shift keying, for applying frequency shift keying modulation to the bit stream; and digital-to-analog conversion means for converting the output from said modulation filter into an analog signal for transmission.

By modulating the signal using an FIR filter, the present invention allows the signal
30 to be modulated in a single clock cycle. Furthermore the implementation is simple and can

be created easily and cheaply on an integrated circuit.

Advantageously, the modulator may additionally comprise coding means for encoding data in the modulation index, said modulation index having at least two values, said at least two values each resulting in a different amplitude in the output of said modulation filter, allowing the transmission of more than one symbol per transmitted bit by encoding additional symbols through the variation of said modulation index.

By varying the modulation index (and hence the amplitude of the FSK modulated output) more symbols may be transmitted with only a marginal increase in the bandwidth requirements for transmission. 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.

Optionally, the filter may have an impulse response corresponding to the transfer function from the original bit stream to the bit stream modulated using frequency shift keying with an additional function applied.

The additional function may be applied to the transmitted data to assist in the rejection of noise added in the channel. An example of such a function is a logarithm function.

According to a second aspect of the present invention there is provided a modulator for modulating a digital bit stream using frequency shift keying comprising: coding means for encoding more than one symbol from said digital bit stream per transmitted bit by encoding data in the modulation index as well as the transmitted bit, said modulation index having at least two values, and for outputting said modulation index; and modulating means for modulating data using frequency shift keying with said modulation index output from said coding means.

Thus, the transmission rate of the communication may be increased without any significant increase in the bandwidth requirements.

Advantageously, the modulator of the above described first and second aspects may be extended to modulate data on both the in-phase and quadrature components of the carrier.

The transmission rate of the system is then further increased with little change in the bandwidth requirements.

According to a third aspect of the present invention there is provided a method of modulating a digital bit stream using frequency shift keying or phase shift keying, the
5 method comprising: passing said digital bit stream through a filter having an impulse response corresponding to the transfer function from said digital bit stream to the modulated output, the output from said filter being the modulated digital bit stream; and converting said modulated digital bit stream into an analog signal for transmission.

The modulation of a signal can be achieved simply by applying the original bit
10 stream to a filter with an impulse response corresponding to the transfer function from the bit stream to the modulated data. The impulse response will be fixed for given transmission parameters, so it may be fixed or calculated at the start of each modulation burst.

Optionally, the method may be applied to frequency shift keying wherein the modulation index can take at least two defined values and further comprise the step of
15 encoding more than one symbol of said digital bit stream per bit of said modulated output signal by varying the modulation index between at least two values.

According to a fourth aspect of the invention there is provided a method of modulating a digital bit stream using frequency shift keying comprising the steps of: encoding data in the modulation index such that more than one symbol from said digital bit
20 stream is transmitted per transmitted bit, said modulation index having at least two values; and modulating data with frequency shift keying using a modulation index resulting from step of encoding.

Thus, the transmission rate of the communication may be increased without any significant increase in the bandwidth requirements.

Advantageously, the digital bit stream may be split into two channels, and the
25 method of the above described third or fourth aspect applied to modulate one of the said two channels onto the in-phase component of a carrier, and to modulate the other of the said two channels onto the quadrature component of said carrier.

This allows the transmission rate to be further increased with very little effect on the
30 bandwidth requirements.

A preferred embodiment of the invention will now be described, by way of example only, with reference to the accompanying drawing which is a schematic of a FSK modulator according to an embodiment of the present invention.

5 Firstly, the theory underlying the present invention will be explained then a detailed description of one embodiment of the invention will be given.

In order to design an improved modulator for PSK or FSK it is necessary to consider the characteristic equations of the modulating bit stream and the output from the modulator. The theory for a modulator sing FSK with non-return-to-zero (NRZ) coded
10 data will be considered, however the skilled person will realise that the principles may be applied to PSK modulation and data that is not coded with an NRZ method.

The discussion above first considered the case of modulating analog data using FM and then discussed transmission of digital data via the same modulator, giving rise to FSK. It can be seen that FSK has no need of an intermediate frequency and therefore the
15 modulator can instead create a signal as given in equation (4) directly. This will have the additional benefit of removing the need for an IF carrier multiplication step and low pass filter in the demodulator.

If the impulse response that will transform a signal consisting of only one pair of a one and zero encoded in an NRZ format into an FSK modulated signal can be calculated,
20 it can be used to modulate any arbitrary 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 types of filering may be applied, for example to give the data the shape of a sine wave, or an approximate sine wave. Filtering the data to have the form of a
25 cosine function will give the transmitted data minimal transmitted bandwidth.

The input data is assumed to be a square wave. Substituting this into equation 5 gives:

$$x(t) = -\frac{1}{2} A \sin (h \pi / 2) / (h \pi / 2) \quad (5a)$$

where h is an integer defining the order of harmonic and A is the amplitude of the square wave. It will be assumed that $A=1$ for simplicity. With reference to equation (4) we can derive what the output of the modulator given this square wave input should be:

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

Here we are assuming that the square wave $x(t)$ has been transformed to the cosine function $\cos(\omega_m t)$ in the output of the modulator, applying the cosine wave filtering mentioned above.

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

A modulation FIR filter can then be designed which has the correct impulse response to modulate an input bit stream. A signal $x(t)$ may then be modulated simply by feeding it into the modulation filter.

It may be of benefit to apply a function to the modulated data, for example to assist
) 20 in the rejection of additive noise added to the modulated signal in the transmission channel. Such a function can be applied easily by calculating its Fourier transform and using it to multiply equation (4a) in the frequency domain. The impulse response is then calculated and the function will be applied in the modulation step with no extra processing required.

Alternatively, a function may be applied by convolving it with the impulse response
25 in the time domain. The modulation filter is then using the result of the convolution calculation.

It will be appreciated that the coefficients of the FIR filter will in general 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.
30 This fact can be used to further increase the transmission rate in a given system.

If a filter is designed to modulate a signal with a known value of m , the output given an input $x(t)$ with the known value of m is $g'(t)$ as expected. However, if an input signal $x(t)$ with a different value of m is modulated using the filter, it is found that the output signal $g'(t)$ has the correct shape expected if the input had the value of m the filter was designed for,

5 but has a different amplitude.

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 transmitted the output from the filter will be $g'(t)$. If data modulated with $m=2$ is transmitted the output from the filter
) 10 the have the correct shape but will have a different amplitude. Likewise, if data modulated with $m=4$ is transmitted the output from the filter will have the correct shape but with a different amplitude. Thus, data is additionally encoded in the amplitude of the FSK signal. 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 $-m$.

15 This multi-level coding can be implemented easily at the coding stage since equation (5a) shows that m is merely an amplitude factor in the coded data signal $x(t)$.

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 modulator for the quadrature channel. Likewise, the multi-level property created by the
) 20 controlled variation of m can be extended to such a system, thus realising a system similar to Quadrature Amplitude Modulation (QAM). Such a system will have a better signal-to-noise ratio than QAM because it is based on frequency modulation techniques.

Furthermore, the characteristics of the filter may be calculated once and stored (for example when the modulator is manufactured) or may be calculated whenever the
25 modulator is used. Thus, the same modulator could be used to modulate data using different parameters as required.

An embodiment of a FSK modulator for radio signals according to the present invention is illustrated in Figure 1. A digital bit stream is input to the coder 2 which applies coding to reduce the dc content of the signal to as close to zero as possible. Several
30 methods may be used, depending on the characteristics of the individual communication

system, for example Manchester or Miller coding. The coder 2 outputs a binary bit stream which is supplied to the shift register 3.

The FIR filter 6 has an impulse response corresponding to the transfer function from a binary bit stream to the FSK modulated signal. This impulse response is derived as described above. Thus, the output from the FIR filter 6 is the FSK modulated signal.

For maximum performance the FIR filter 6, which applies FSK modulation to the signal, has a set of coefficients stored for each value of m . By storing the coefficients in this way the modulator has improved performance than if a single set of coefficients are used and the amplitude of the signal is varied as described above. The coefficients are pre-calculated and stored in the FIR filter 6. They are then selected by simply specifying the value of m required, for example m , $0.5m$, $-m$ and $-0.5m$.

In the present embodiment four symbols, held in the shift register 3, are encoded per transmitted bit. The polarity symbol held in the shift register is supplied to the FIR filter 6. The remaining three symbols are supplied to an encoder 4 to determine which value of m is required. The encoder 4 supplies the value of m to the FIR filter 6 and thus determines the set of coefficients used in the FIR filter 6.

The FIR filter 6 thus applies FSK modulation only to the polarity bit. The data of the other symbols is encoded in the modulation index used to apply FSK modulation to the polarity symbol.

The next section of the modulator is a filter and DAC 8 which converts the digital output of the filter 6 into an analog signal and filters it remove the harmonics associated with the "stepped" output of a typical sample-and-hold DAC. Final the signal is up-converted by the up converter 10, amplified by amplifier 12 and transmitted through aerial 14.

The above preferred embodiment has not described extending the modulation to include both in-phase and quadrature components. This can be achieved by splitting the input bit stream into an in-phase and a quadrature signal, applying the modulator described to the in-phase signal and simply providing a second modulator as described above for the modulation of the received quadrature signal.

The shift register 3 and encoder 4 may be applied to modulators which do not use

the single cycle FIR modulator described in the first embodiment. The concept of varying the modulation index may be applied to any modulator in which it is possible to input a desired modulation index. Such a modulator may be analog or digital.

Furthermore, it will be appreciated that the shift register 3 and encoder 4 may be omitted if the transmission of more than one symbol per transmitted bit by varying the modulation index is not required. The above described embodiment may also be extended to encode an a different number of symbols per transmitted bit by increasing or reducing the size of the shift register.

10 According to the present invention there is provided an FSK modulator which gives optimal performance with a cheap cost of manufacture. Since the modulation is provided entirely by FIR filters, demodulation can be achieved in 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.

15 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, such as broadband modems.

Whilst the embodiment described above has been applied to radio communication systems, it will be appreciated that the invention may be applied to any other communication using FSK, for example modems.

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 modulator for modulating a digital bit stream using phase shift keying or frequency shift keying, the modulator comprising:

5 a modulation finite impulse response filter having an impulse response corresponding to the transfer function from the original bit stream to the bit stream modulated using frequency shift keying or phase shift keying, for applying frequency shift keying modulation to the bit stream; and

 digital-to-analog conversion means for converting the output from said modulation
) 10 filter into an analog signal for transmission.

2. A modulator according to claim 1 wherein said filter has an impulse response corresponding to the transfer function from the original bit stream to the bit stream modulated using frequency shift keying with an additional function applied.

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3. A modulator according to any preceding claim additionally comprising coding means for encoding data in the modulation index, said modulation index having at least two values, said at least two values each resulting in a different amplitude in the output of said modulation filter, allowing the transmission of more than one symbol per transmitted bit by
) 20 encoding additional symbols through the variation of said modulation index.

4. A modulator according to any preceding claim wherein said bit stream is modulated using frequency shift keying on both the in-phase and quadrature components and additionally comprising a second finite impulse response filter for modulating the quadrature component.

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5. A modulator for modulating a digital bit stream using frequency shift keying comprising:
 coding means for encoding more than one symbol from said digital bit stream per transmitted bit by encoding data in the modulation index as well as the transmitted bit, said modulation index having at least two values, and for outputting said modulation index;

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 modulating means for modulating data using frequency shift keying with said

modulation index output from said coding means.

6. A modulator according to claim 5, wherein said bit stream is modulated using frequency shift keying on both the in-phase and quadrature components, additionally comprising
5 second coding means and second modulating means for encoding and modulating data to be transmitted on said quadrature component.

7. A modulator according to any one of the preceding claims formed as an integrated circuit.

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8. A modem incorporating the modulator of any one of the preceding claims.

9. A method of modulating a digital bit stream using frequency shift keying or phase shift keying, the method comprising:

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passing said digital bit stream through a filter having an impulse response corresponding to the transfer function from said digital bit stream to the modulated output, the output from said filter being the modulated digital bit stream; and
converting said modulated digital bit stream into an analog signal for transmission.

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10. A method according to claim 9, wherein said impulse response of said filter corresponds to the transfer function from said digital bit stream to the modulated output signal with a function applied.

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11. A method of modulating a digital bit stream using frequency shift keying according to claim 9 or claim 10, wherein the modulation index can take at least two defined values and further comprising the step of encoding more than one symbol of said digital bit stream per bit of said modulated output signal by varying the modulation index between at least two values.

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12. A method of modulating a digital bit stream using frequency shift keying comprising the

steps of:

encoding data in the modulation index such that more than one symbol from said digital bit stream is transmitted per transmitted bit, said modulation index having at least two values; and

- 5 modulating data with frequency shift keying using a modulation index resulting from step of encoding.

13. A method of modulating a digital bit stream using frequency shift keying wherein said digital bit stream is split into two channels, and the method of any preceding claim is applied

- 10 to modulate one of the said two channels onto the in-phase component of a carrier, and to modulate the other of the said two channels onto the quadrature component of said carrier.