(12) UK Patent Application (19) GB (11) 2 373 151 (13) A

(43) Date of A Publication 11.09.2002

(21) Application No **0125219.6**

(22) Date of Filing **19.10.2001**

(30) Priority Data
(31) **0025659** (32) **19.10.2000** (33) **GB**

(71) Applicant(s)
Radioscape Limited
(Incorporated in the United Kingdom)
2 Albany Terrace, LONDON, NW1 4DS, United Kingdom

(72) Inventor(s)
Gavin Robert Ferris

(74) Agent and/or Address for Service
Origin Limited
52 Muswell Hill Road, LONDON, N10 3JR, United Kingdom

(51) INT CL[7]
H04H 1/00

(52) UK CL (Edition T)
H4P PABC

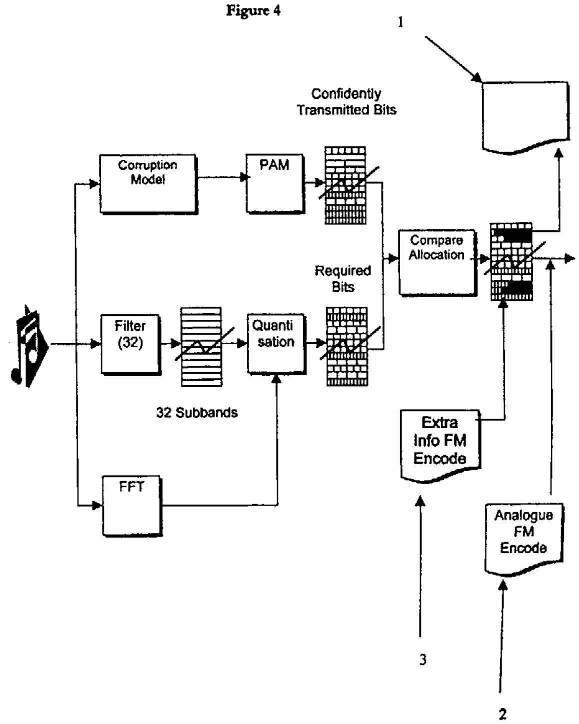
(56) Documents Cited
EP 1041766 A2 **WO 2000/069100 A1**
R. Barron and A. Oppenheim, "A Systematic Hybrid Analog/Digital Audio Coder", Proc. 1999 IEEE Workshop on Application of Signal Processing to Audio and Acoustics, pp35-38

(58) Field of Search
UK CL (Edition T) **H4P PABC PABX PAX**
INT CL[7] **H04H 1/00 , H04L 5/02 27/32**
Online: WPI, EPODOC, JAPIO

(54) Abstract Title
Hybrid analogue/digital media transmission or communication system

(57) Disclosed is a method of transmitting a signal comprising a main analog FM component and digital OFDM sidebands (see Figure 1). At the receiver the digital data in the sidebands is combined with the analogue data to produce a signal with a higher fidelity than the analogue signal alone. At the transmitter the expected fidelity of the analogue signal is compared with a target fidelity to determine the digital data required to ensure that the signal produced at the receiver has the target fidelity. With this system the sidebands require less bandwidth than with conventional IBOC systems and therefore additional data can be transmitted. The target fidelity is calculated using a psycho-acoustic model (PAM). In an alternative embodiment AM analogue signals are used instead of FM analogue signals.

Figure 4



GB 2 373 151 A

Figure 1

Prior Art

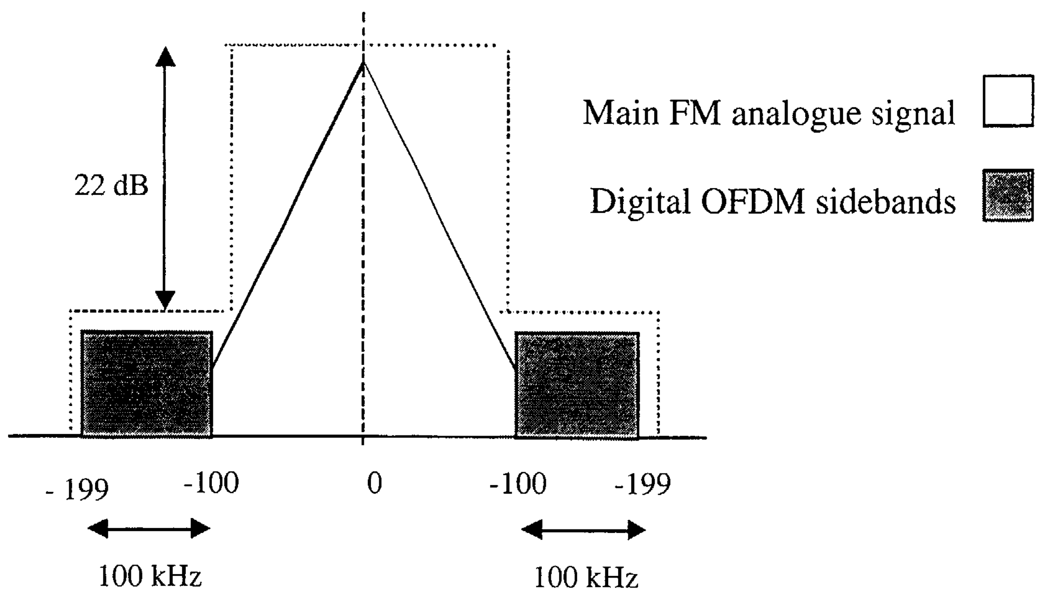


Figure 2

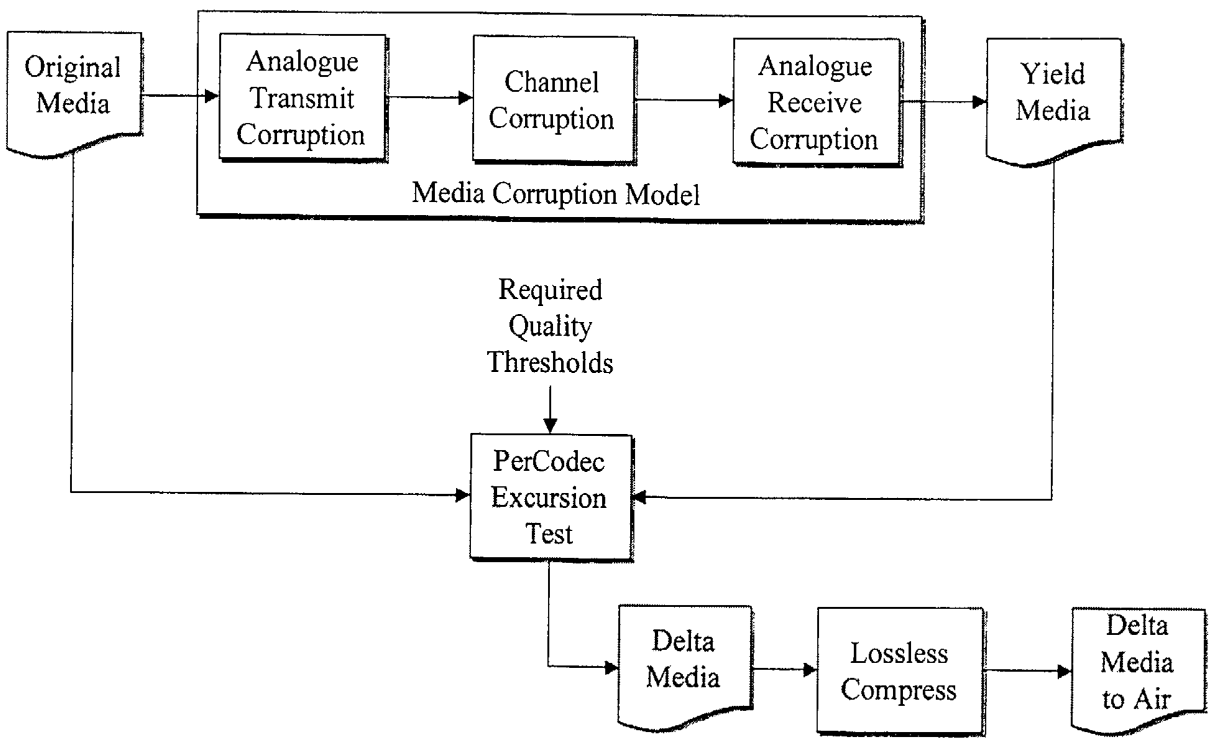

Original Media
Analogue Transmit Corruption
Channel Corruption
Analogue Receive Corruption
Yield Media
Media Corruption Model
Required Quality Thresholds
PerCodec Excursion Test
Delta Media
Lossless Compress
Delta Media to Air

Figure 3

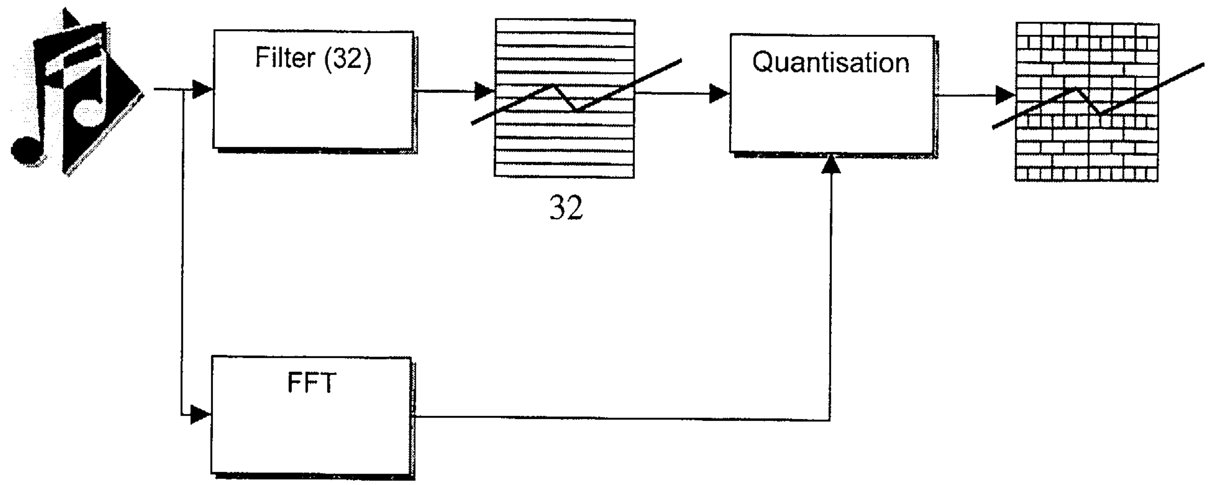
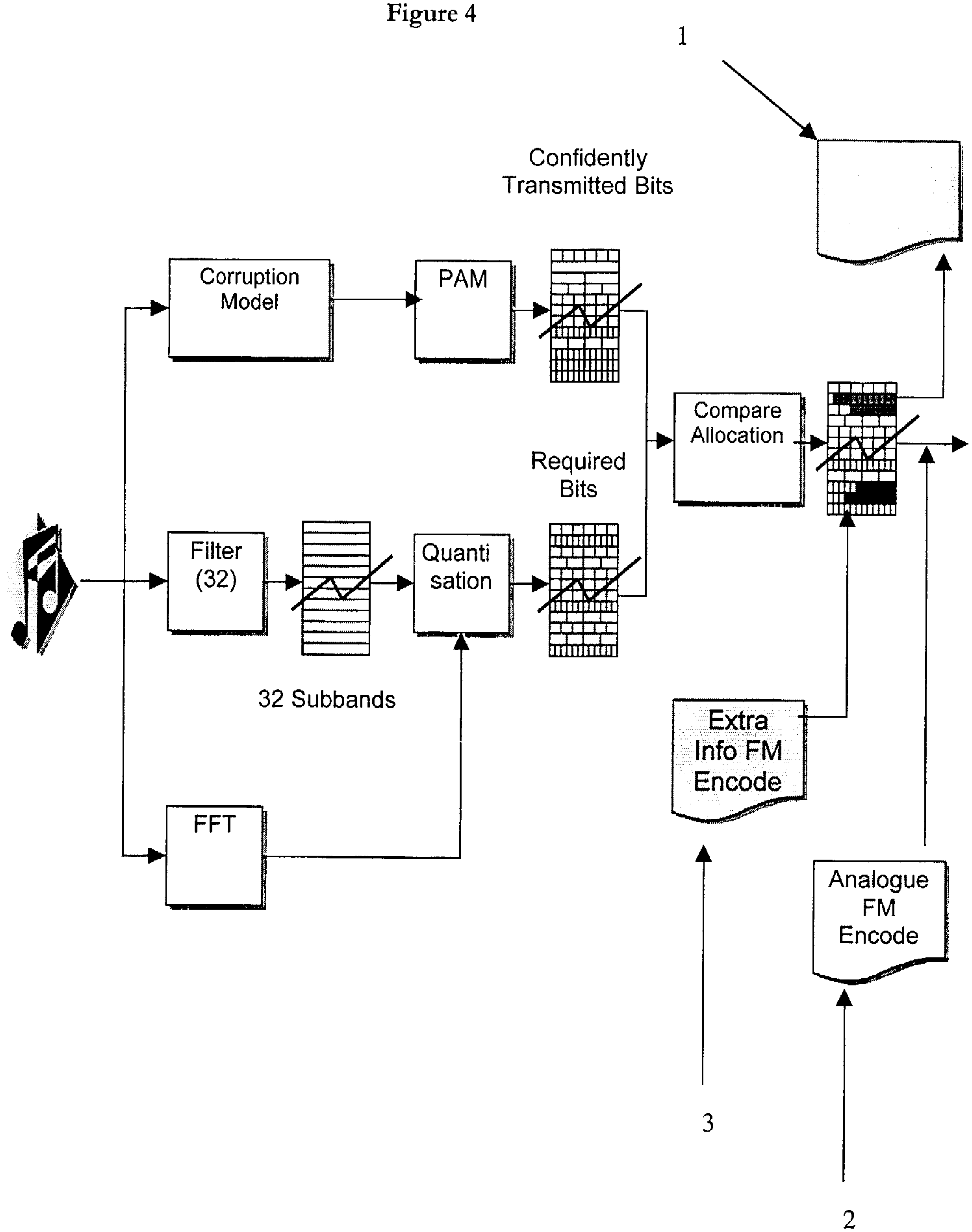


Figure 4


1
Confidently
Transmitted Bits
Corruption
Model
PAM
Compare
Allocation
Required
Bits
Filter
(32)
Quanti
sation
32 Subbands
Extra
Info FM
Encode
FFT
Analogue
FM
Encode
3
2

HYBRID ANALOGUE/DIGITAL MEDIA TRANSMISSION OR COMMUNICATION SYSTEM

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates to hybrid analogue/digital transmission or communication systems. Such systems send signals in both digital and analogue format and are designed typically to preserve, during a transition from analogue to entirely digital, (a) the usefulness of existing analogue legacy equipment and (b) existing channel allocations.

2. Description of the Prior Art

Hybrid analogue/digital transmission or communication systems are found in many different contexts and can carry media of any type, including music, audio and video, as well as data. One particular example of a hybrid analogue/digital transmission system is the in-band on channel ('IBOC') audio system which uses a combination of analogue FM with digital information inserted into the sidebands. In the IBOC system, a high-quality perceptual audio codec (e.g., EPAC) is utilised to generate two copies of the original audio channel, which is simultaneously utilised as the input to the conventional analogue FM carrier chain. One copy of the digitally encoded signal is delayed (with respect to the analogue and the other digital copy) to provide a degree of resilience to time-coherent channel corruption.

The digital information is then inserted into the sidebands of the modulated analogue FM signal using an OFDM coding methodology, at a significant power reduction from the original signal. Depending upon the mode, some additional carriers may be inserted over the top of the FM core signal itself.

On the decode chain, the decoder blends up to three streams – the digitally processed analogue component, the un-delayed digital stream, and the delayed digital stream – to yield the highest quality output. **Figure 1** shows the analogue and digital bandwidths used in IBOC

In IBOC, the transmitted power for the digital sidebands is 22dB below that of the analogue FM carrier. The subcarriers are spaced about 200 Hz apart, and are modulated using QPSK. The USA digital radio system proposal is similar, and it is likely that the joint iBiquity radio system (ibiquity.com) will therefore follow a similar model.

Whilst IBOC provides good protection against a number of potential channel corruptions, it essentially ignores (so far as digital transmission is concerned) the fact that the analogue FM signal is present, and at a significantly higher power than the OFDM IBOC carriers, other than as a (third) redundant channel that can contribute to the blending process. Although IBOC does envisage a mode in which 'all digital' transmissions are used (where there is no FM carrier and hence room for more digital carriers at higher power), the primary driver for IBOC is to maintain the status quo in terms of legacy equipment capability and channel allocations, so for many applications, the analogue FM component may reasonably be expected to be present for many years to come. Where an analogue component is present, it is in effect ignored for digital data transmission purposes; whilst hybrid systems use analogue as well as digital signals to ensure redundancy of information, they fail to exploit fully the data carrying potential of the digital component, which in IBOC is at a 2 x 64 kbps source rate. As spectrum becomes an increasingly scarce resource, it will become increasingly important to exploit band allocations to the full.

SUMMARY OF THE PRESENT INVENTION

In a first aspect of the invention, there is a method of sending digital data in conjunction with an analogue signal for reception at a receiver at a target fidelity, comprising the following steps:

(a) comparing a model which defines the target fidelity against a model of the likely fidelity of the analogue signal taking into account noise and/or corruption;

(b) determining what digital data would have to be combined with the analogue signal in order to enable the target fidelity to be achieved;

(c) combining the analogue signal with that digital data;

(d) sending the combined analogue signal and digital data.

Typically, the digital data combined with the analogue signal does not occupy all of the bandwidth available for digital data, so allowing additional data to be sent: the additional data is revenue earning. In this way, the present invention allows scarce and expensive spectrum to be re-exploited.

Hence, in contrast with the prior art, this invention utilises the analogue signal (e.g. high-powered FM) as a key part of the signal chain. At its core, the invention in one implementation uses and exploits:

- A target throughput fidelity for the output audio signal, with respect to the originally presented audio, which is expressed as a set of quality parameters with respect to a psychoacoustic model (PAM).
- A model for the likely channel effects to which the analogue signal will be subjected (this is our expected reasonable channel); and
- A model for the noise that will be injected by the transmission and reception parts of the signal chain.

We are then able to work as follows: combine the noise/corrupting components into a single model of the yield signal, calculate the audible excursions that this would suffer with respect to the original as calculated by or with a PAM, and then generate a 'delta code' information stream that provides us with a sufficient statistical level of confidence to remove the error within the limits of fidelity set by our PAM. This delta coding may then be losslessly compressed before being sent to air in lieu of the 2x redundant full digital streams, to minimise bandwidth. The overall system is shown in **Figure 2**.

In another aspect, there is a method of receiving a combined analogue signal and digital data in which a decoder combines the digital data with the analogue signal to yield a resultant signal at a target fidelity, in which the digital data has been selected prior to being sent:

(a) using a process in which a model defining the required target fidelity is compared against a model of the likely fidelity of the analogue signal taking into account noise and/or corruption; and

(b) and only that digital data is selected which, when combined with the analogue signal, enables the target fidelity to be achieved.

Additional aspects and details are defined in the claims.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will now be described with reference to the accompanying drawings in which:

Figure 1 shows the bandwidth allocations used in the IBOC hybrid digital/analogue radio system;

Figure 2 is a block representation of the current inventive system;

Figure 3 is a block representation of a Musicam encoder; and

Figure 4 is a block representation of the encoding process used in the current inventive system.

DETAILED DESCRIPTION OF THE PREFERRED IMPLEMENTATION

Digital Delta Coding Scheme for Media with Analogue Transmission Component and Known Corruption Model

The present invention can be thought of as a delta scheme. A variant of the IBOC system, modified using this delta scheme, is now described. The modified IBOC system can provide a very good increase in perceived quality of the received signal for a very low bit rate consumption due to the delta coding itself. Of course, where there is FM corruption in excess of that stipulated by the model, then the signal would be lost for that moment, and the triple-redundant original IBOC signal is likely to be more resilient in absolute terms (particularly, with regard to corruptions that are localised in time, such as travelling through an underpass, where the 4s time diversity of the original scheme's coding will prove useful). However, the delta coding method provides the benefit of high economy and good performance, yielding digital quality in any area where the FM radio would work acceptably (as opposed to redefining those areas in which the radio works *per se*), and frees a large quantity of the 2x64 kbps in the hybrid mode for pure data applications, which can be revenue generating.

Although delta coding is not directly supported in the IBOC standard, it can easily be retrofitted to receivers, since the delta stream may be carried as a standard IBOC transparent data stream. For those legacy receivers that do not understand the coding, their audio blender will 'see' a single conventional FM station, and the receiver will therefore work just as well as any normal analogue FM receiver. However, where both the FM and the delta audio are present, *and* the receiver has the appropriate software to perform the extraction and recombining, then the user will be presented with a digitally reconstituted stream, which will be of higher fidelity than the FM.

Delta Coding Scheme Details

The overall delta coding scheme as described above is applicable, in principle, to any media stream (not just audio), where an analogue carrier subject to a known corruption model is enhanced by the simultaneous transmission of a delta channel with respect to that psychometric codec (PerCodec) and with respect to a desired fidelity envelope.

For IBOC digital audio, a number of different delta coding schemes could be utilised. Various degrees of sophistication are possible, but for simplicity and ease of understanding, an Musicam (MPEG 1 layer 2) is discussed as a possible embodiment below.

Musicam Sample Embodiment

According to the standard Musicam algorithm, bandlimited, time packetised, input digital PCM audio is fed through a 32-way filterbank, and then a PAM (psychoacoustic model) is used to compute the allowable signal to noise ratio for each of these bands, which, ultimately, is used to set a quantisation rate for each band (number of bits needed to represent the information in the band for a given time instant, without this creating audible artefacts). The PAM is sensitive to both time and frequency localised masking effects. Finally (not relevant to us here), the audio is bit packed into an output frame. The overall process in block form is shown in **Figure 3**.

Standard Musicam Processing for one Timeslot (Simplified)

In our modified delta scheme, we have a probabilistic model of the total corruption that the audio signal will undergo as a result of transmission on an FM carrier through a particular channel. Each sub-band will have, as a result of this model, a resolution confidence expressed in bits for each audio frame. Next, for each sub band, we can take this resolution confidence and *compare it* with the *required* number of bits for that frame computed with respect to the PAM of the original Musicam codec. We then proceed as follows for each subband:

- If there are *exactly the required number* of bits for the frame, then the FM representation is as accurate as the digitised one, and no information need be transmitted for that subband for that timeslot. These bits are labelled 2 in **Figure 4.**
- If there are *fewer* bits of information that we can rely on being carried in the FM signal for a subband as compared to the amount required, then the additional information must be added to the delta file. These bits are labelled 1 in **Figure 4**. The delta file is periodically written out to the processing buffer for a transparent

data channel on the transmitter. In the preferred embodiment this 'flush' happens after each audio frame is processed (e.g., every 24 ms). This delta file will generally be found not to consume the full 2 x 64 kbps source rate transparent data capability, so the sideband OFDM may therefore be utilised to carry additional revenue bearing services in addition to the delta stream.

- If there are *more* bits of information carried by the corrupted FM than are (according to the PAM) audible, then we may in fact utilise that extra information, using digital processing at the transmitter. The extra bits of resolution in the subband for the particular timeslot, which, according to the PAM are inaudible and hence redundant, may be overwritten using appropriate processing with additional payload information, labelled 3, which thereby becomes encoded into the FM audio representation itself. This yields an additional information store that may be utilised (without reducing the audio quality on conventional FM analogue receivers, since the bits modified are, by definition, inaudible), above and beyond the standard IBOC sideband payload.

In the digital audio processing, a steganographic marker can be inserted to mark the 'start of analysis' frame for the receive chain. Ensuring the alignment of the analysis-synthesis window at the receive side could alternatively be performed with respect to the digital (sideband) symbols; for example, a modulo alignment over their numbering scheme is possible: if the PAM window was 24 ms and the symbol duration was, say, 1ms, then there would be 1 PAM window every 24 symbols, and if there was a frame numbering associated with the digital symbols, then we could have the PAM analysis start, by stipulation, where this number modulo 24 equals 0. Aligning the analysis in this way would obviate the need for the steganography to lock the PAMs at both ends.

Figure 4 below illustrates the encoding process.

Adding Delta Bits to FM and Overwriting Inaudible Bits in FM

A variant system may be envisaged in which a time delay is imposed at the transmission side, so that a 'bit buffer' may be built up of redundant (and hence overwritable) bits within the audio carried by the analogue FM, such that the additional required 'delta' bits can then be filled into this buffer, using both carry back (utilising the transmission delay) and, potentially, some carry forward (if there is an additional receiver delay). In one potential embodiment the use of this technique could obviate the requirement for separate sideband OFDM subcarriers, at least to carry the primary digital audio payload or delta file. However, in the normal and preferred embodiment of the system, only one frame of delay is utilised at the FM encoder, and each PCM frame is analysed as just rehearsed, then any extra bits (labelled 3 in **Figure** 4) will be set to hold payload information before the data is requantised and sent to the FM exciter.

In another preferred embodiment, which is a logical extension of the Musicam paradigm discussed above, MP3 encoding is utilised instead. In yet another preferred embodiment, multi-stream encoders (such as Lucent's EPAC) may be utilised successfully.

Decoding Process for the Example Musicam Variant

In the receiver, the decoder would begin by first starting to decode the FM signal in the normal manner, using digitised IF. This would initially be fed through as the only input to the audio blender. Next, an audio encoder frame search would begin, to locate the steganographic markers in the audio marking the point at which the analysis-synthesis window is placed on the transmit side.

Once this (for Musicam, 24ms) frame sync has been located, the decoder begins to process the audio by subjecting it to the 32-band filterbank, to generate the coefficients for each frame. Simultaneously, the 'delta' stream is decoded from the transparent IBOC data channel in the normal manner. The 'delta' coefficients (more LSB resolution, essentially) are then applied as necessary to the audio. Using an in-band PAM, the decoder will also be able to determine which bits are excess and carry payload information, and will then be able to extract these and make them available on a port for further processing (NB – convolutional coding may be utilised in the payload bits to assist in valid extraction, together with a high level block code for correction and validation).

The reconstructed bits are then subjected to the normal multiband filter synthesis and windowing to generate the output (enhanced) PCM which may be rendered in the normal manner.

Additional Notes and Points

The system may also be successfully operated in AM hybrid mode, although there the likely channel corruptions are more severe and so the benefits are likely to be significantly reduced. As mentioned above, the concept of the delta encoder, although here described for use with IBOC and audio, has application to any hybrid analogue/digital media transmission or communications system.

CLAIMS

1. A method of sending digital data in conjunction with an analogue signal for reception at a receiver at a target fidelity, comprising the following steps:

   (a) comparing a model which defines the target fidelity against a model of the likely fidelity of the analogue signal taking into account noise and/or corruption;

   (b) determining what digital data would have to be combined with the analogue signal in order to enable the target fidelity to be achieved;

   (c) combining the analogue signal with that digital data;

   (d) sending the combined analogue signal and digital data.

2. The method of Claim 1 in which the digital data combined with the analogue signal does not occupy all of the bandwidth available for digital data, so allowing additional data to be sent.

3. The method of Claim 2 in which the additional data is revenue earning.

4. The method of any preceding claim in which the digital data is a delta signal which is losslessly compressed prior to being sent.

5. The method of any preceding claim in which the target fidelity is generated with respect to a psycho-acoustic model.

6. The method of any preceding claim in which the model of the likely fidelity of the analogue signal takes into account noise and/or corruption by modelling the likely channel effects to which the analogue signal will be subjected and the additional noise that will be injected by the transmission and reception parts of the signal chain.

7. The method of Claim 6 in which each model is a probabilistic model.

8. The method of any preceding Claim in which the analogue signal is FM radio and each sub-band has, as a result of a noise/corruption model used to generate the model of the likely fidelity of the analogue signal, a resolution confidence expressed in bits for each audio frame, which is compared with the required number of bits for that frame computed from the model with respect to which the target fidelity of the digital data is specified.

9. The method of Claim 8 in which, where there are exactly the required number of bits for a given frequency subband for a particular frame, then the FM representation is treated as being as accurate as the digitised one, and no digital information is sent for that subband for that time slot (frame).

10. The method of Claim 8 or 9 in which, if there are fewer bits of information that can be relied on in the FM signal for a subband compared to the amount required to yield the target fidelity, then digital data is added to the analogue signal such that the target fidelity is reached.

11. The method of Claim 10 in which the digital data is added as a delta file which is periodically written out to a processing buffer for a transparent data channel on the transmitter.

12. The method of claim 10 or 11 where the digital data is added to the FM signal using overlay modulation.

13. The method of claim 12 where the overlay modulation scheme is some form of OFMD (orthogonal frequency division multiplexing), utilising some combination of either or both of phase and amplitude modulation on each carrier.

14. The method of Claims 8 – 13 in which, if there are more bits of information carried by the corrupted FM than are according to a psycho-acoustic model audible, then that redundant information is overwritten processing with additional payload information.

15. The method of Claim 14 in which an analysis-synthesis window at the receive side is aligned with the sent signal.

16. The method of Claim 15 in which a steganographic marker is inserted to mark the beginning of the analysis synthesis window used by the PAM.

17. The method of Claim 15 in which alignment is based on the digital sideband symbols.

18. The method of any preceding claim in which a time delay is imposed at the transmission side, so that a 'bit buffer' is built up of redundant (and hence overwritable) bits within the audio carried by the analogue FM, such that the additional required 'delta' bits can then be filled into this buffer, using both carry back (utilising the transmission delay) and, optionally, some carry forward (if there is an additional receiver delay).

19. The method of Claims 8 – 18 in which only one frame of delay is utilised at the FM encoder, and each PCM frame is analysed, such that any extra bits are set to hold payload information before the data is requantised and sent to the FM exciter.

20. The method of Claims 8 – 19 in which MP3 encoding is utilised.

21. The method of Claims 8 – 19 in which multi-stream encoders are used.

22. A method of receiving a combined analogue signal and digital data in which a decoder combines the digital data with the analogue signal to yield a resultant signal at a target fidelity, in which the digital data has been selected prior to being sent:

(a) using a process in which a model defining the required target fidelity is compared against a model of the likely fidelity of the analogue signal taking into account noise and/or corruption; and

(b) and only that digital data is selected which, when combined with the analogue signal, enables the target fidelity to be achieved.

23. The method of Claim 22 in which the decoder:

(a) initially decodes the analogue signal in the normal manner, using digitised IF and feeds this as the only input to an audio blender;

(b) an audio encoder frame search begins to locate any steganographic markers in the audio marking the point at which an analysis-synthesis window is placed on the transmit side;

(c) once this frame sync has been located, the decoder begins to process the analogue signal by subjecting it to a 32-band filterbank, to generate the coefficients for each frame; and simultaneously, a 'delta' stream is decoded from the digital data.

24. The method of Claim 23 in which the delta coefficients are then applied as necessary to the audio subband coefficients and using an in-band PAM, the decoder determines which bits are excess and carry payload information, extracts these and makes them available on a port for further processing.

25. The method of Claim 23 in which the reconstructed bits are subjected to a normal multiband filter synthesis and windowing to generate the output (enhanced) PCM.

26. The method of Claim 24 or 25 in which channel coding techniques are used to protect any excess 'payload' bits encoded into the FM and to assist in the 'blind' detection of any such stream.

27. The method of Claim 26 in which the channel coding techniques include: convolutional encoding and block coding.

29. An encoder programmed to perform the method of Claims 1 – 21.

30. A receiver programmed to perform the method of Claims 22 – 27.

31. A method of generating revenue based on re-utilising bandwidth, in which the bandwidth is made available through a method of sending digital data as defined in Claim 2 and any claim dependent on Clam 2.

32. A hybrid analogue/digital signal generated using the method of Claims 1 – 21.

15

Application No:	GB 0125219.6	**Examiner:**	John Cullen
Claims searched:	1-32	**Date of search:**	28 June 2002

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.T): H4P (PABC, PABX, PAX)

Int Cl (Ed.7): H04L 5/02, 27/32; H04H 1/00

Other: Online: WPI, EPODOC, JAPIO

Documents considered to be relevant:

Category	Identity of document and relevant passage		Relevant to claims
A	EP 1041766 A2	(LUCENT) See entire document, but particularly paragraphs 5, 16, 24, 25 and 58, and Figs. 2-4A.	---
X, P	WO 00/69100	(MIT) See entire document, but particularly the Abstract, lines 2-6 of p2, line 16 of p8 to line 24 of p9, lines 2-11 of p11 and line 17 of p12 to line 13 of p13, and Figs. 1, 2, 7 and 8.	1, 2,4, 5,22
X	R. Barron and A. Oppenheim "A Systematic Hybrid Analog/Digital Audio Coder", Proc. 1999 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, pp35-38, see entire document.		1-7, 22, 31

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
		E	Patent document published on or after, but with priority date earlier than, the filing date of this application.
&	Member of the same patent family		