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(54) Receiver for receiving digital broadcast transmissions

(57) A digital audio broadcast receiver for receiving at least two broadcast ensembles, each ensemble being made up of a plurality of transmission frames and each transmission frame including an information channel and a time interleaved main service channel. The broadcast receiver includes means for acquiring and processing during a plurality of relatively long periods a given broadcast ensemble to extract and reproduce selected audio or other service data from said main service channel. The means for acquiring and processing is also operable to process a further broadcast ensemble at least to detect the presence thereof during one or more relatively short intervals interspersed with said relatively long periods, thereby to obtain data relating to said further ensemble.

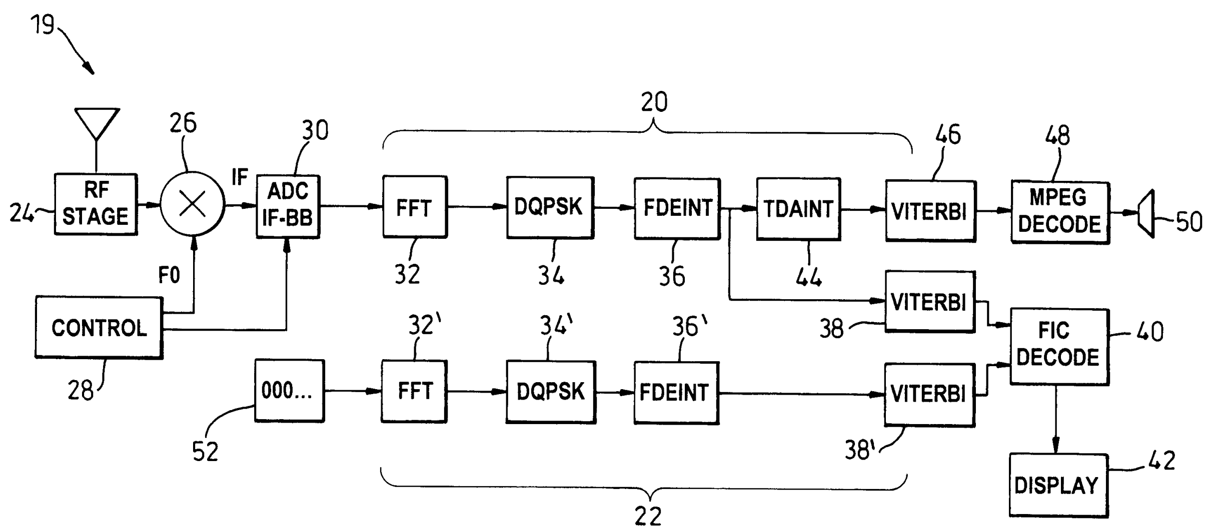

Fig. 3

MAIN ENSEMBLE DECODING

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Description

Field of the Invention

[0001] This invention relates to broadcast receivers and in particular, but not exclusively, to digital audio broadcast (DAB) receivers and to methods of signal processing implemented in said receivers.

[0002] Although primarily concerned with radio broadcast transmissions the invention is not so limited and extends to broadcast transmissions over other media such as optical fibre.

Background to the Invention

[0003] In a DAB system, the broadcast DAB signal may typically comprise a mixture of stereo and mono broadcasts and data services carried in a single ensemble. Thus a ensemble may comprise a variable number of services, depending on the quality (i.e. bit rate) required and this may change during the day dependent on broadcast schedules.

[0004] A typical DAB system is that developed under the Eureka 147 Project with a DAB transmitted signal now specified by ETSI specification ETS 300401, the entire contents of which are incorporated herein by reference. Alternative schemes exist in U.S.A., e.g. In-Band, On-Channel (IBOC), Digital Audio Radio (DAR), and elsewhere.

[0005] The Eureka 147 DAB system is a reliable multi-service digital radio broadcasting system designed specifically for robust reception by mobile portable and fixed receivers using non-directional antennas. The ETS 300 401 signal contains time interleaving and error-correcting codes that are designed to cope with signal loss over a short duration.

[0006] The Eureka 147 system comprises three main components; namely audio coding; transmission coding and multiplexing, and OFDM modulation.

[0007] The audio coding component uses a coding scheme known as MPEG-1 Audio Layer II encoding. Other services to be transmitted may be encoded using suitable coding algorithms.

[0008] The encoded data for the individual services whether audio-based or data/multimedia must be combined by multiplexing the data into a single multiplex for transmission. The multiplex data is assembled into transmission frames which are made up of a synchronisation channel, a fast information channel (FIC) and a main service channel (MSC).

[0009] The synchronisation channel contains a null symbol and a sync symbol (otherwise known as the phase reference signal) which convey reference frequency and timing information to allow receivers to synchronise to and to decode the received DAB signal. The FIC contains information relating to the composition of the multiplex and informs the receivers how to extract and decode the information for individual services. The MSC contains the audio frames or data packets corresponding to different services in the multiplex, and these frames are usually time interleaved and referred to herein as common interleaved frames (CIF). The multiplex data is subjected to convolutional forward error correction channel coding, time interleaving and frequency interleaving to provide strong protection against bit errors.

[0010] The OFDM scheme consists of a large number of orthogonally spaced subcarriers transmitted at a relatively slow symbol rate, and the broadcast signal comprising the group of subcarriers is known as an ensemble. In Eureka 147 DAB Mode I, there are 1536 carriers spaced at 1KHz separation. Each carrier is independently modulated using differential quadrature phase shift keying (D-QPSK) and the multiplex data is distributed amongst all the carriers, occupying approximately 1.54MHz of spectrum. The low symbol rate together with the guard period which is added to each symbol provides significant protection against multipath interference.

[0011] In a conventional DAB receiver, an ensemble is acquired and processed so that all of the services making up the multiplex in the main service channel of the ensemble are available for selection by the user. For example, in U. K., the BBC multiplex may typically contain national services Radio 1, Radio 2, Radio 3, Radio 4, Radio 5 Live and further audio or data services. Information identifying the audio and data services currently provided on the multiplex and how to extract a selected service is contained in the FIC. Having acquired the ensemble containing the BBC multiplex, the user may readily select which audio or data service he required from that multiplex without requiring retuning of the receiver.

[0012] In the U.K. and elsewhere, a number of different frequencies have been allocated for use by the BBC and Independent National and Local Radio for transmitting respective ensembles each containing a multiplex of audio or data services. In conventional DAB receivers, a different ensemble is acquired by switching the receiver RF front end over to the appropriate frequency and waiting for the receiver to acquire and decode the FIC and the MSC of the new ensemble. In many cases however, it is desirable to be able to determine the content of different ensembles without appreciably degrading the current programme. For example information regarding other ensembles may be used to present channel content information on a visual display associated with the receiver. Alternatively, it may be useful to acquire and track the frequency and symbol timing of another ensemble so that switching to that ensemble can be effected rapidly when required. Still further it may useful just to know that another ensemble is present at a different frequency.

[0013] Ordinarily, a multi-channel reception facility would require a separate radio frequency (RF) front-end to allow a further ensemble to be acquired without disrupting reception and reproduction of the current service, and this would increase the cost of the receiver.

Summary of the Invention

[0014] A DAB receiver is described herein with a single RF front-end which, whilst tuned to a primary signal, can acquire and monitor other signals without appreciable disruption to the primary signal, thereby extending the capabilities of the receiver at little cost.

[0015] In this design the powerful time interleaving and error correction capabilities of the Digital Audio Broadcast (DAB) signal are exploited to allow a receiver which is tuned to and decoding a particular ensemble to tune (hop), for a relatively short time, to a second frequency before returning to the primary frequency. By repeatedly hopping, a second ensemble can be acquired, synchronised to and partially demodulated. Provided the hopping is performed sufficiently infrequently, the powerful time interleaving and error correction capabilities of such a broadcast signal mean that any degradation to the audio quality of the primary ensemble due to the loss of samples during the hop, is minimal.

[0016] Accordingly, in one aspect, this invention provides a digital audio broadcast receiver for receiving at least two broadcast ensembles, each ensemble being made up of a plurality of transmission frames and each transmission frame including an information channel and a time interleaved main service channel, said broadcast receiver including means for acquiring and processing during a plurality of relatively long periods a given broadcast ensemble to extract and reproduce selected audio or other service data from said main service channel, said means for acquiring and processing also being operable to process a further broadcast ensemble at least to detect the presence thereof during one or more relatively short intervals interspersed with said relatively long periods, thereby to obtain data relating to said further ensemble.

[0017] In this arrangement the intervals are sufficiently short that there is little or no appreciable degradation of the audio or other service data from said given broadcast ensemble. The effect of the data lost from the given ensemble during said short intervals is diluted and spread over time, due to said interleaving.

[0018] With this technique, searching for a second ensemble in frequency is effectively a background task to main ensemble decoding. Once located, and synchronisation information obtained, very rapid acquisition to that ensemble may be made should the receiver wish to make that ensemble the primary ensemble. With this technique partial decoding of a second ensemble is possible allowing information as to the contents of that ensemble to be obtained. This would be achieved by decoding the fast information channel (FIC) symbols only. Thus the user of such a DAB receiver can be made aware of programme material present in other ensembles without having to explicitly tune to those ensembles. This technique is not limited to only a second ensemble, as acquisition and decoding of further ensembles (possibly refreshing at a lower rate) is a straightforward extension.

[0019] Preferably, said acquiring and processing means includes means for providing in-fill data for being processed to replace data from a portion of said transmission frame from said first ensemble lost during said one or more relatively short intervals. The in-fill data may comprise a stream of binary zeroes or ones, or a stream of random or pseudo-random bits.

[0020] In a system where said broadcast ensembles each comprise a multi-carrier or an orthogonal frequency division ensemble (OFDM) signal incorporating time interleaving and error correcting, the acquiring and processing means may conveniently comprise:-

transform means for converting a received broadcast ensemble signal from the temporal domain to the frequency domain;
demodulation means for demodulating said frequency domain signal;
frequency de-interleaving means for frequency de-interleaving said demodulated signal;
information processing means for processing a portion of the frequency de-interleaved signal corresponding to the fast information channel of the transmission frame to extract information data, and
main service channel processing means for processing a portion of the frequency de-interleaved signal corresponding to the main service channel to time de-interleave said signal and to apply error correction to extract and reproduce said audio or other service data.

[0021] In such a system said demodulation means, said frequency de-interleaving means and said information channel processing means are preferably operable during said relatively short intervals to process information from said further broadcast ensemble.

[0022] In the case where the broadcast ensembles are modulated onto respective radio frequency carriers, said acquiring and processing means preferably comprise:-

a common radio frequency (RF) stage for receiving and amplifying each of said RF broadcast ensembles;

tunable frequency conversion means for modulating a selected broadcast ensemble down from the carrier frequency, and
analogue to digital conversion means for converting the down converted broadcast ensemble to digital form.

[0023] Preferably, said means for acquiring and processing is operable during said one or more intervals to capture a sample sequence from said further broadcast ensemble, and to process said sample sequence to obtain an estimate of the approximate symbol timing.
[0024] Where the receiver is operable to receive broadcast ensembles in any of a plurality of transmission modes, each having different transmission frame structures and timings, said means for acquiring and processing is preferably operable during said one or more intervals to capture a sample sequence from said further broadcast ensemble and to process said sample sequence to determine the transmission mode of said further broadcast ensemble.
[0025] In a case where each of said ensembles is transmitted as a multi-carrier or OFDM signal comprising an ensemble of sub-carriers at a known relative spacing, and each of the said transmission frames of the ensembles includes a sync symbol, said means for acquiring and processing is preferably operable during said intervals to capture from one or more transmission frames of said further broadcast ensemble a complete frame of samples succession of samples making up a partial or complete frame, and preferably further includes sync symbol processing means operable to process each of said captured samples to determine whether it is the sync symbol.
[0026] The sync symbol processing means is preferably further operable to process the sync symbol sample to determine the coarse frequency offset of the ensemble.
[0027] For this purpose a technique of the form described in our co-pending U.K. Published Application No. 2,320,868A (the contents of which are incorporated herein by reference) may be used. Accordingly, where said sync symbol comprises a substantially self-orthogonal sequence which repeats across the sub-carriers, said sync symbol processing means preferably includes means for processing said sync symbol to obtain a signal having a generally regular data structure, and means for applying a comb filter function generally matched to said structure thereby to deduce the coarse frequency offset of said multi-carrier signal.
[0028] In another aspect, this invention provides a method of receiving and processing two or more broadcast ensembles each comprising a plurality of transmission frames, with each transmission frame including an information channel, and a main service channel which is time interleaved, said method comprising acquiring and processing during relatively long periods a given said broadcast ensemble to extract and reproduce audio or other service data, and processing another broadcast ensemble during relatively short periods interspersed with said longer periods, at least to detect the presence thereof. As previously, the processing of the other broadcast ensemble may determine just the presence of another ensemble, and/or the frequency and timing thereof, and/or it may extract information from the information channel.
[0029] In a further aspect, this invention provides a broadcast receiver for acquiring and processing a plurality of different broadcast signals, each said broadcast signal comprising data in time interleaved and/or encoded form, said receiver comprising means for acquiring and processing during a plurality of relatively long periods a given broadcast signal to extract and reproduce audio or other service data, wherein said means for acquiring and processing is also operable to at least detect the presence of a further broadcast signal during one or more relatively short intervals interspersed with said relatively long periods, whereby data from said further broadcast is obtained whilst the audio or other service data from said given broadcast channel is reproduced.
[0030] Whilst the invention has been described above, it extends to any inventive combination of the features set out above or in the following description.

LIST OF FIGURES

[0031] The invention may be performed in various ways and an example thereof will now be described in detail, by way of example only, reference being made to the accompanying drawings, in which:-

Figure 1 is a schematic representation of a DAB transmission frame for mode 1;
Figure 2 is a schematic representation of the useful symbols pertaining to a particular subchannel beginning half way through a mode 4 common interleaved frame (CIF) ;
Figure 3 is a representation of a first embodiment of a DAB receiver in accordance with this invention, illustrating a primary and a secondary, temporally distinct processing path, when decoding a primary ensemble;
Figure 4 is a representation similar to that of Figure 3, but when background decoding a secondary ensemble;
Figure 5 is a diagram illustrating the performance degradation caused by channel hopping in the embodiment of Figures 3 and 4 for hopping ratios of 0, 1/12 and 1/16 respectively, and
Figure 6 is a diagram indicating how the second ensemble capture times may be maximised.

DESCRIPTION OF PREFERRED EMBODIMENT

[0032] The embodiment described below is a DAB receiver demodulating a signal conforming to the ETS 300 401 specification. As noted above, the ETS 300 401 signal contains time-interleaving and error correcting codes that are designed to cope with the loss of signal for a short duration. The signal is based on multicarrier modulation (OFDM) which consists of a large number of orthogonally spaced subcarriers transmitted at a relatively slow symbol rate. The DAB specification has four modes whereby the number of subcarriers, the symbol rate and hence the number of symbols making up a frame varies. Figure 1 shows a typical transmission frame 10. The signal is frame based consisting of two symbols used for synchronisation purposes (called the Synchronisation Channel) 12, a few symbols containing the fast information channel 14 (FIC) and the remainder making up the main service channel (MSC) 16. Each symbol in mode 1 codes for 3072 bits (1536 carriers, with two bits per carrier). In ETSI 300 401 the transmission frame for mode 1 consists of 2 Sync channel symbols, 3 FIC symbols and 72 MSC symbols giving a frame duration of 96ms. The FIC channel is not time-interleaved, which allows fast decoding. The FIC contains details of the contents of the audio subchannels contained in the MSC.

[0033] The MSC is divided into a number of common interleaved frames (CIFs). Each CIF contains all the information to decode 24ms of audio data and they occur at an average rate of one per 24ms for all DAB modes. For instance, a 96ms mode 1 frame has four CIFs, each of N_c = 72/4 = 18 MSC symbols duration and therefore each mode 1 CIF codes for 55296 bits (18 symbols each coding for 3072 bits). Time interleaving operates by spreading data within one CIF over 16 CIFs, hence over a duration of 384ms. Note that the receiver is not required to capture and demodulate every symbol because only part of each CIF is occupied by a particular subchannel. Figure 2 shows the useful symbols pertaining to a particular subchannel beginning half-way through a mode 4 CIF 18. Because of the differential nature of DAB modulation, the symbol prior to a symbol containing wanted information is required and so only these, together with the immediately preceding symbol (referred to as a phase reference signal or PRS), need be captured.

[0034] Figure 3 is a schematic diagram illustrating the basic structure and operation of a DAB receiver 19 producing audio and displaying information in accordance with this invention. For ease of visualisation and understanding, the receiver 18 is shown as having an upper path 20 and a lower path 22. However it should be noted that these paths represent the operation of the receiver at different times, and the upper and lower paths are not separate physical channels. Thus the upper path represents processing of a main broadcast ensemble and the lower path represents a processing facility for use during channel hopping to process a second ensemble without appreciably disrupting reproduction of the selected subchannel main broadcast ensemble.

[0035] The upper path shows a demodulation chain consisting of the necessary inverse operations to decode the signal transmitted according to the ETS 300 401 specification. The path consists of an RF downconversion stage 24 which amplifies and filters the wanted RF signal, a frequency shifting stage 26 which uses a tuning signal from a control 28 to modulate the signal down from the carrier frequency FO to an intermediate frequency (IF) suitable for analog to digital conversion (ADC) at an analog to digital converter 30. Following conversion to baseband a sample capture system strips the guard period off the transmitted symbols and converts them to the frequency domain with a Fast Fourier Transform (FFT) at 32. Each symbol is differentially demodulated by a demodulator 34 and frequency deinterleaved at 36. At this point FIC symbols (not requiring time deinterleaving) are passed to a Viterbi decoder 38 (for channel error correction) and a FIC decoder 40 which passes suitable information to a display device 42. Other MSC symbols, in general containing audio, pass via a time deinterleaver 44, a Viterbi decoder 46 and an MPEG audio decoder 48, to produce an audio signal for reproduction at 50. The Sync channel symbols are used to update the receiver's tracking parameters, namely frequency tracking and symbol tracking.

[0036] The lower path 22 consists of the same symbol processing elements as the FIC processing of the upper path 20. In this case, during the time periods when the primary ensemble alone is being processed, no signal is passed to the FFT unit and the lower path 22 processing is essentially disabled.

[0037] Figure 4 illustrates the same receiver 18 tuned to a new frequency F1 in a frequency hop, to extract data from a second ensemble broadcast at frequency F1. Assuming that acquisition of the second ensemble has already occurred, the switch-over to frequency F1 is timed to coincide with the arrival of the Sync channel and FIC symbols of the second ensemble. The Sync channel symbols allow the receiver 18 to update its tracking parameters whilst the FIC symbols are passed to the lower path 22 for FIC decoding. The lower path is supplied with the downconverted digital baseband signal from the analog to digital converter 30.

[0038] Zeroed samples are passed from a source 52 to the upper path 20 for the duration of the time tuned to frequency F1. The time deinterleaving spreads the zeros over a large number of bits, resulting in errors spread more evenly over time. The Viterbi decoder is presented with a higher than usual error rate which, depending on the conditions, may be fully or partially corrected.

[0039] Also, the frame-based transmission and processing allows short time slots or channel hop intervals to be allocated for background processing the second ensemble (i.e. the lower path 22) interspersed with the much longer periods during which the main ensemble is processed (i.e. the upper path 20), without disrupting the reconstruction of the selected subchannels of the main ensemble being reproduced.

[0040] The time deinterleaving spreads the effects of zero insertion over 16 CIFs, and hence zeros inserted in CIF (n) are spread over data passed to the Viterbi decoder 38 for CIF(n)...CIF(n+15). It is necessary to maximally spread the times that data is zeroed to avoid concentrating errors. Hence what we refer to as a channel hopping ratio of 1/12 would be performed by zeroing data once every twelve CIFs.

[0041] In the following analysis, the time spent absent from the main ensemble is denoted by D_1 and the time spent capturing data from the secondary ensemble is denoted D_2 ($D_2<D_1$). Practical values for D_1 and D_2 are discussed below. D_2 is the interval during which data from the RF front end is supplied to the lower path 22 for processing.

[0042] The extent by which the increase in errors degrades the signal depends on many factors. The main dependencies are the level of error protection encoded into the signal, the operating environment and the percentage of time spent in the second channel. The ETS 300 401 specification defines a range of protection levels which range from level 1 (strong) to level 5 (weak). The stronger levels are used for mobile receiver environments where the effect of Rayleigh fading causes large amplitude and phase changes over time. In general, a receiver 19 will judge the suitability of switching to a channel hopping mode depending on the protection level of the main signal and the operating environment.

[0043] For instance a stationary receiver 19 demodulating a level 1 signal, spending 1/16th of its time in the second channel will suffer virtually no degradation in performance. On the other hand, a rapidly moving mobile receiver 19 demodulating a signal with a weak protection level could be affected to the point where noticeable audio distortion occurs.

[0044] Figure 5 shows the bit error rate (BER) vs. carrier to noise ratio for a typical case where the receiver 19 is operating in a static environment tuned to a 192 kbps protection level 3 main subchannel without channel hopping (hopping ratio = 0), shown in solid line. Also shown is the performance degradation caused by channel hopping to a second ensemble for 1/12th and 1/16th of the time, shown in dotted and chain dotted lines respectively. The small loss of performance of ~0.6dB and ~0.4dB respectively (at a BER of 10e-3) is likely to be acceptable.

[0045] The method by which a second ensemble is acquired whilst continuing to decode a main ensemble falls into two distinct parts. Firstly coarse symbol timing determines both the transmission mode of the second ensemble and the approximate location of the start of a symbol. Thereafter the second ensemble is located accurately in frequency and time.

[0046] Coarse symbol timing is obtained by tuning to the trial frequency F1, corresponding to the suspected frequency of the second ensemble, capturing a random segment of contiguous samples, and returning to the main ensemble. Longer capture times will, in general, improve performance. These samples are background processed to determine both the transmission mode of the ensemble (if present) and an approximate location of the timing of each symbol. The mode and symbol timing may be determined quite robustly using a correlation technique that exploits the cyclic repetition (i.e. of the symbol in the time domain i.e. the guard period).

[0047] Specifically, given *l* samples of complex signal *x* and trial mode with T'_u, T'_s the useful symbol period and total symbol length respectively, a correlation vector ***r*** is constructed using:

$$r(k) = \sum x(i)x^*(i + T'_u)$$

$$i = k, k + T'_s, \ldots\ldots\ldots\ k + mT'_s$$

for

$$k = 0 \ldots T'_s - 1$$

and

$$m = T'_s \lfloor l / T'_s \rfloor - 1$$

$\lfloor \cdot \rfloor$ denotes rounding to the lowest integer

* denotes complex conjugation

[0048] The correlation is repeated for all four possible modes and the maximum correlation peak, together with the timing offset is recorded for each. Note that mismatching the trial mode and the actual mode gives a substantially noise-like correlation result. If sufficient difference exists between the highest and second highest correlation peak the mode and timing offset is deemed to be found otherwise the acquisition is aborted.

[0049] If successful, this results in the following second ensemble mode and symbol timing information:

$\Delta t'$ second ensemble symbol timing offset from start of captured sequence which began at T_0
T'f second ensemble frame length
T's second ensemble total symbol length
T'u second ensemble useful symbol length, i.e. the total symbol length less the symbol guard period.

[0050] Having obtained a coarse estimate of the starting times of the second ensemble symbols, a whole frame's worth of logically consecutive samples are captured and scanned for the presence of a Sync symbol. Because of the repeating frame structure of the signal, a complete frame of samples can be captured over time which are not contiguous. For instance if a mode 1 signal has been detected, a mode 1 frame (96ms in duration) could be captured in four lots of 24ms, each capture separated by 384ms. In fact, the scanning proceeds on the first capture and if a Sync symbol is not found, the next capture occurs and so on until four captures have occurred. If the Sync symbol has failed to be identified, the acquisition process is terminated.

[0051] Our published patent application GB 2320868 A specifies the steps needed to obtain a coarse frequency offset and a simple way of validating the correlation result by determining the ratio of peak energy to inter-peak energy. This SNR-like measurement can be used to indicate the presence of a Sync symbol.

[0052] The receiver therefore implements the following procedure. K symbols are captured each of T'u in length, (in increments of T's) beginning at sample $T_1 = T_0 + \Delta t' + nT's$, where K x T's is <= D_2, and n is chosen such that captured sequences are about 384ms apart (for 1/16th hopping ratio).

[0053] An FFT is performed on each symbol followed by the coarse frequency calculations and SNR measurement as described in GB 2320868A. If the resultant SNR measurement is sufficiently high the procedure is stopped, having found the coarse frequency offset $\Delta f'c$.

[0054] If the Sync symbol has not been found in the first capture block, further blocks of K symbols are captured starting at $T_{k+1} = T_k + KxT's + nT'f$, until the total number of symbols captured and scanned exceeds the number in a second ensemble frame. If unsuccessful on capturing a complete frames worth of symbols, the second ensemble acquisition is reset.

[0055] Upon successfully finding a valid Sync symbol and the coarse frequency offset, the location of other symbols is known. Both fine frequency and fine timing adjustments may then be made using normal DAB tracking processing.

[0056] A second ensemble may be continuously tracked (and hence demodulated) in time and frequency using the channel hopping method by exploiting the fact that, in practice, while the channel transfer function may change rapidly a receiver 19 is not required to respond rapidly. This is largely due to the inherent robustness of OFDM with DQPSK subcarriers and its lack of a requirement to perform adaptive equalisation. Hence updating the second ensemble frequency offset and symbol timing every, say, 384ms will in most circumstances be adequate. Note that the required capture times for the second ensemble Sync Channel/FIC are much less than 24ms. For instance, in mode 1 only 5 symbols are required from a DAB frame containing 77 symbols.

[0057] Tracking to the main ensemble will operate normally except in the case where the Sync Channel and/or FIC of the second ensemble overlap in time with those of the main ensemble. The same slow tracking arguments apply to the main ensemble which, in any case, will be serviced the majority of the time.

[0058] In this embodiment during channel hopping the receiver demodulates the FIC only of the second ensemble. This is updated infrequently according to ETSI 300 401 (the multiplex can change at most only once every six seconds and in most cases much less frequently) and is protected with a relatively strong convolutional code which helps with imperfect tracking.

[0059] Turning now to the aspect of second ensemble capture times, maximising the capture times can be useful during acquisition for several reasons. The correlation technique used in the coarse timing and mode determination phase benefits from longer captured sequences. Similarly the speed of Sync symbol scanning benefits from longer capture.

[0060] In this analysis, the usual case is considered where only a single CIF's worth of data is discarded from the main ensemble within a long hopping period. In this case the capture time can still be greater than a CIF duration if the number of wanted symbols is less than the maximum number of symbols in a CIF,N_c. The maximum main ensemble absence time D_1 is obtained by discarding symbols straight after the last one captured up until the phase reference symbol of the next wanted CIF. Figure 6 illustrates the case whereby $D_1 = 2N_c-N_w-1$ symbols duration where N_w is the number of wanted symbols per CIF. For the case where all N_c symbols per CIF are required for demodulation, the maximum value of the time spent absent from the main ensemble D_1 is the duration of N_c -1 symbols. In mode 1, D_1 = 17x1.25 = 21.25ms.

[0061] Considering stabilisation times, there are practical reasons that will limit the actual capture time D_2 of the second ensemble to be somewhat less than the time absent from the main ensemble (D_1). The main limitation is dictated by the time required to retune the hardware to a new frequency. This typically involves waiting for local oscillators (LO) and automatic gain control (AGC) circuits in the RF stage to stabilise. With a stabilisation time of D_y, only $D_2=D_1-2D_y$ is available for capturing second ensemble data. For typical values of D_y < 5ms there is no difficulty in tracking to and demodulating the FIC of a second ensemble. The acquisition step of scanning for successive Sync symbols is clearly prolonged due to the greater number of hops required to scan a frame's worth of samples.

[0062] Turning now to automatic gain control (AGC), large differences between the power of the main and second ensembles requires an AGC which can adapt quickly to avoid excessive stabilisation times D_y. The AGC implemented in a DAB receiver typically consists of at least two autonomous parts - an analog control residing in the early stages of the RF unit and a digitally controlled gain signal that operates prior to and following the ADC. In normal operation, both controls are typically quite slow acting (ie. time constants much greater than useful values of D_y). Immediately following a retune to a new frequency, a rapid AGC adaption needs to occur. Hence the receiver utilises a hardware signal to instruct the early RF stage AGC to reset and quickly readapt within a time < D_y is required. In addition, software within the receiver is provided to restore the state (gain level) of the digitally controlled AGC to that previously used during the last hop. If the level is not previously known or sufficiently inaccurate, a rapid convergence mode is used.

[0063] The channel hopping ratio of $1/16^{th}$ and the nominal capture time of 24ms are given merely as examples. Many variations are possible and have to be weighed against BER degradation and implementation complexity.

[0064] The architecture in Figures 3 and 4 is only high level and other valid means of demodulating are possible. For instance, the insertion of zeros at the FFT stage is not necessary, as they could be inserted after frequency deinterleaving to achieve the same effect. Also, insertion of ones or random data is an equally valid alternative to zeros.

Claims

1. A digital audio broadcast receiver for receiving at least two broadcast ensembles, each ensemble being made up of a plurality of transmission frames and each transmission frame including an information channel and a time interleaved main service channel, said broadcast receiver including means for acquiring and processing during a plurality of relatively long periods a given broadcast ensemble to extract and reproduce selected audio or other service data from said main service channel, said means for acquiring and processing also being operable to process a further broadcast ensemble at least to detect the presence thereof during one or more relatively short intervals interspersed with said relatively long periods, thereby to obtain data relating to said further ensemble.

2. A digital audio broadcast receiver according to claim 1, wherein said acquiring and processing means includes means for providing in-fill data for being processed to replace data from a portion of said transmission frame from said first ensemble lost during said one or more relatively short intervals.

3. A digital audio broadcast receiver according to claim 2, wherein said in-fill data comprises a stream of binary zeroes or ones, or a stream of random or pseudo-random binary values.

4. A digital audio broadcast receiver according to any preceding Claim, wherein said broadcast ensembles each comprise an orthogonal frequency division multiplex (OFDM) signal incorporating error correction and time-interleaving, and said acquiring and processing means comprise:-

   transform means for converting a received broadcast ensemble signal from the temporal domain to the frequency domain;
   demodulation means for demodulating said frequency domain signal;

frequency de-interleaving means for frequency de-interleaving said demodulated signal;
information processing means for processing a portion of the frequency de-interleaved signal corresponding to the fast information channel of the transmission frame to extract information data, and
main service channel processing means for processing a portion of the frequency de-interleaved signal corresponding to the main service channel to time de-interleave said signal and to apply error correction, to extract and reproduce said audio or other service data.

5. A digital broadcast receiver according to Claim 4, wherein said transform means, said demodulation means, said frequency de-interleaving means and said information channel processing means are operable during said relatively short intervals to process information from said further broadcast ensemble.

6. A digital audio broadcast receiver according to any of the preceding Claims wherein said broadcast ensembles are modulated onto respective radio frequency carriers, and said acquiring and processing means comprise:-

a common radio frequency (RF) stage for receiving and amplifying each of said RF broadcast ensembles;
tunable frequency conversion means for modulating a selected broadcast ensemble down from the carrier frequency, and
analogue to digital conversion means for converting the down converted broadcast ensemble to digital form.

7. A digital audio broadcast receiver according to any preceding Claim, wherein said means for acquiring and processing is operable during said one or more intervals to capture a sample sequence from said further broadcast ensemble, and to process said sample sequence to obtain an estimate of the approximate symbol timing.

8. A digital audio broadcast receiver according to any of the preceding Claims, wherein said receiver is operable to receive broadcast ensembles in any of a plurality of transmission modes, each having respective transmission frame structures and timings, and said means for acquiring and processing is operable during said one or more intervals to capture a sample sequence from said further broadcast ensemble and to process said sample sequence to determine the transmission mode of said further broadcast ensemble.

9. A digital audio broadcast receiver according to any of the preceding Claims wherein each of said ensembles is transmitted as a multi-carrier or OFDM signal comprising an ensemble of sub-carriers at a known relative spacing, and each of the said transmission frames of the ensembles includes a sync symbol, and said means for acquiring and processing is operable during said intervals to capture from one or more transmission frames of said further broadcast ensemble a complete frame of samples, and further includes sync symbol processing means operable to process each of said captured samples to determine whether it is the sync symbol.

10. A digital audio broadcast receiver according to Claim 9, wherein said sync symbol processing means is further operable to process the sync symbol sample to determine the coarse frequency offset of the ensemble.

11. A digital audio broadcast receiver according Claim 10, wherein said sync symbol comprises a substantially self-orthogonal sequence which repeats across the sub-carriers and said sync symbol processing means includes means for processing said sync symbol to obtain a signal having a generally regular data structure, and means for applying a comb filter function generally matched to said structure thereby to deduce the coarse frequency offset of said multi-carrier signal.

12. A method of receiving and processing two or more broadcast ensembles each comprising a plurality of transmission frames, with each transmission frame including an information channel, and a main service channel which is time interleaved, said method comprising acquiring and processing during relatively long periods a given said broadcast ensemble to extract and reproduce audio or other service data, and processing another broadcast ensemble during relatively short periods interspersed with said longer periods, at least to detect the presence thereof.

14. A broadcast receiver for acquiring and processing a plurality of different broadcast signals, each said broadcast signal comprising data in time interleaved and/or encoded form, said receiver comprising means for acquiring and processing during a plurality of relatively long periods a given broadcast signal to extract and reproduce audio or other service data, wherein said means for acquiring and processing is also operable to at least detect the presence of a further broadcast signal during one or more relatively short intervals interspersed with said relatively long periods, whereby data from said further broadcast is obtained whilst the audio or other service data from said given broadcast channel is reproduced.

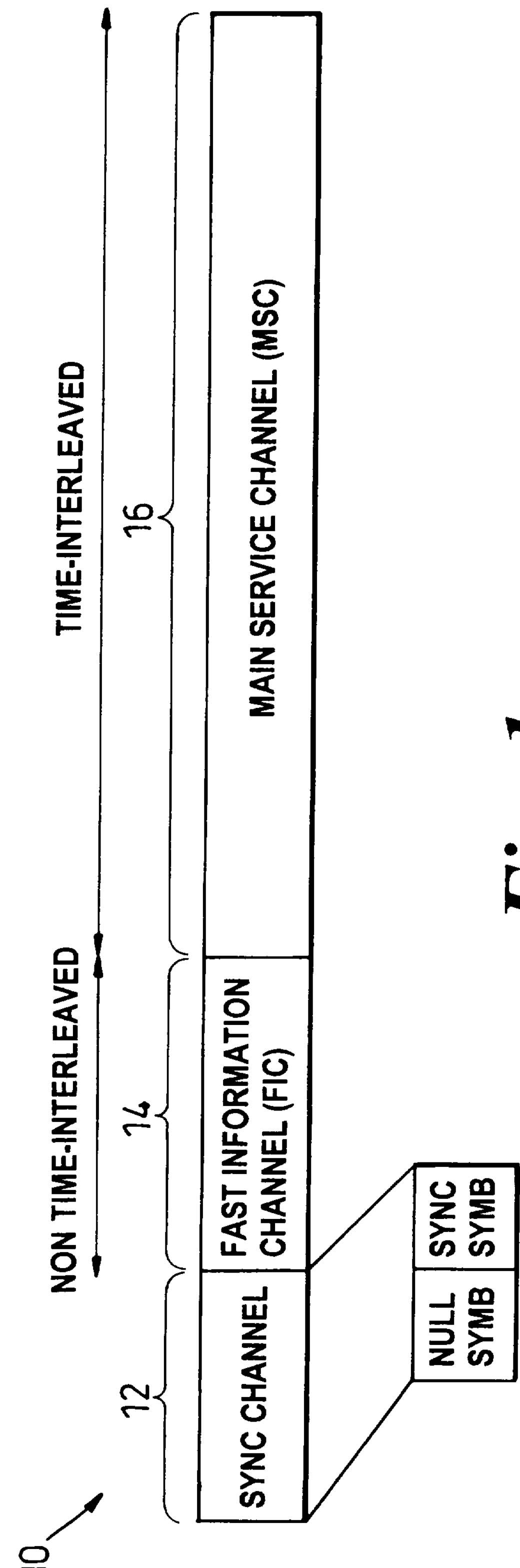

Fig. 1 A DAB FRAME

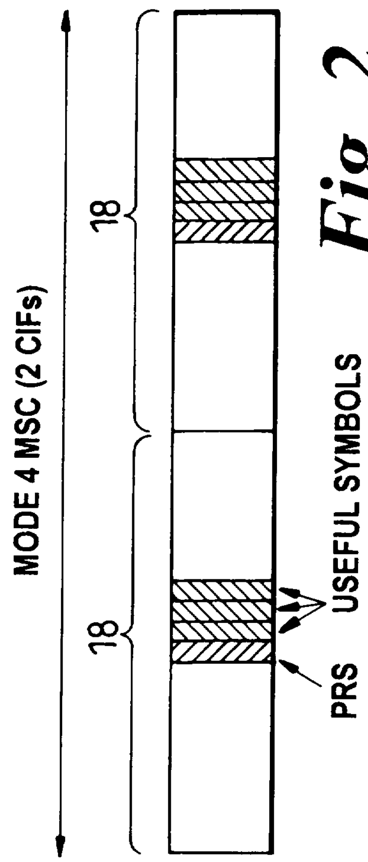

Fig. 2 PARTIAL DEMODULATION

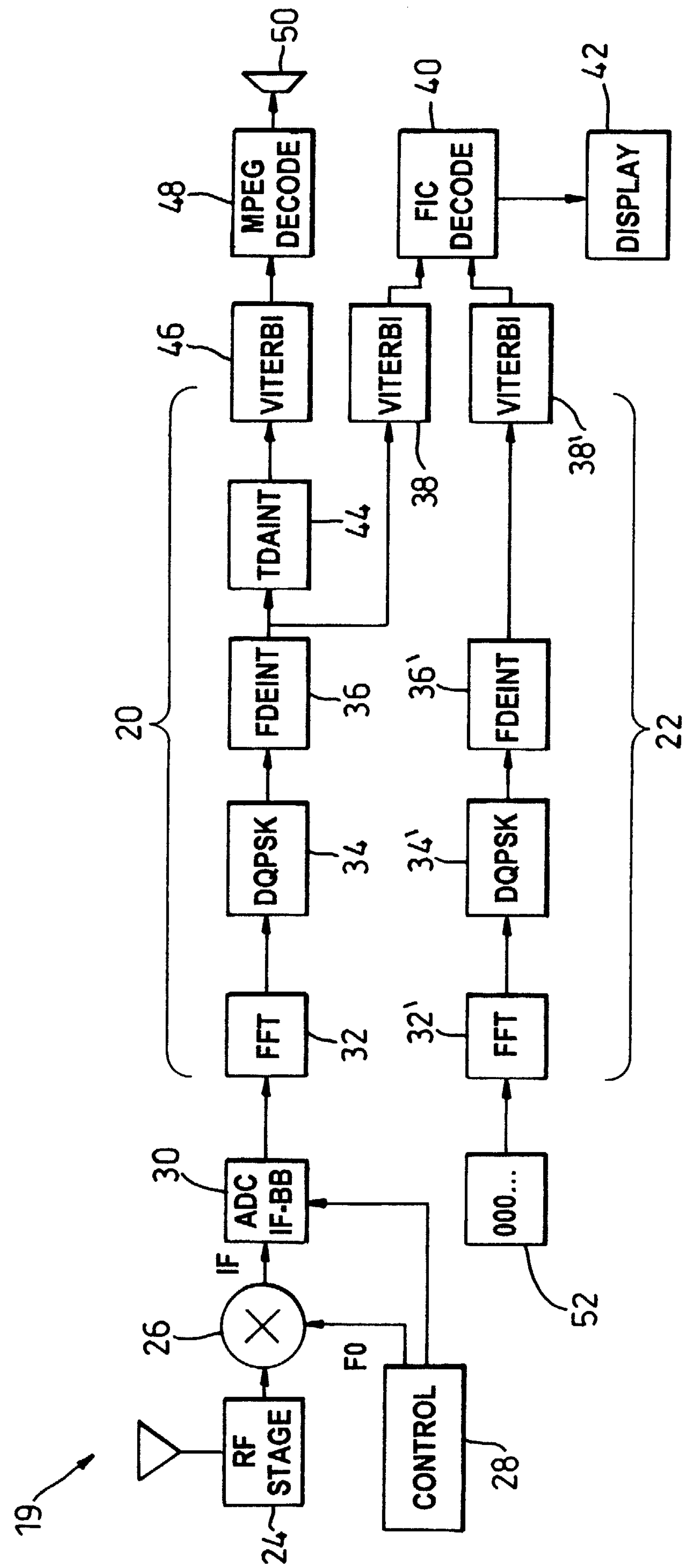

Fig. 3
MAIN ENSEMBLE DECODING

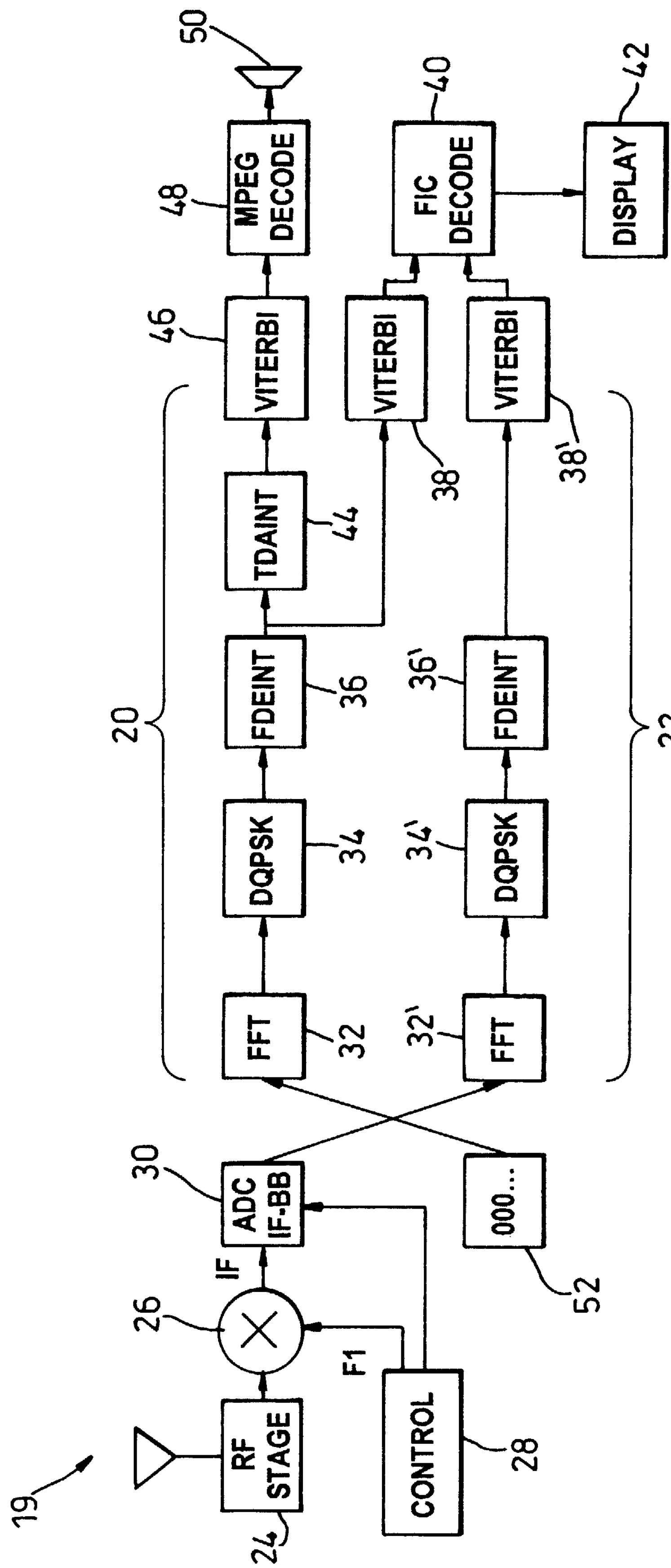

Fig. 4
SECOND ENSEMBLE DEMODULATION
(CHANNEL HOPPED)

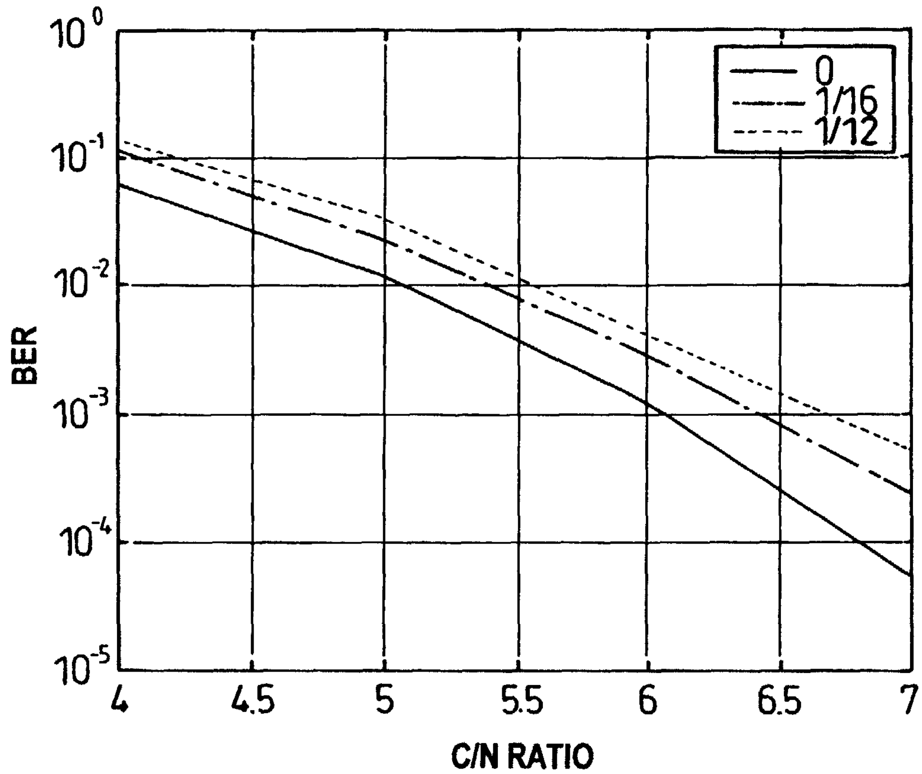

Fig. 5

AWGN PERFORMANCE FOR DIFFERENT CHANNEL HOPPING RATIOS (PROTECTION LEVEL 3)

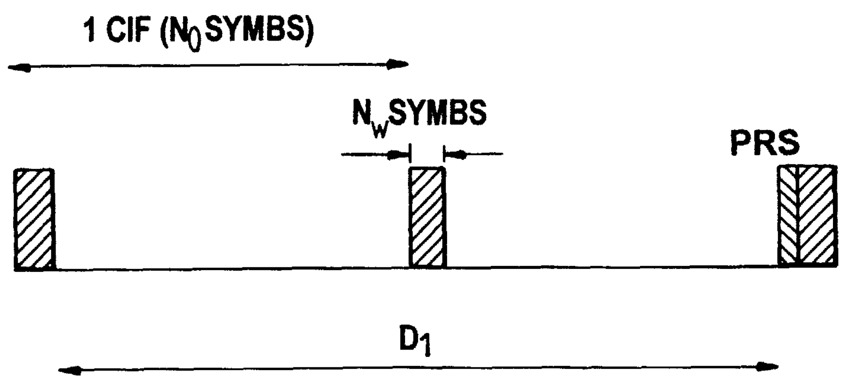

Fig. 6

MAXIMISING CAPTURE TIME