BBC Engineering Design Information

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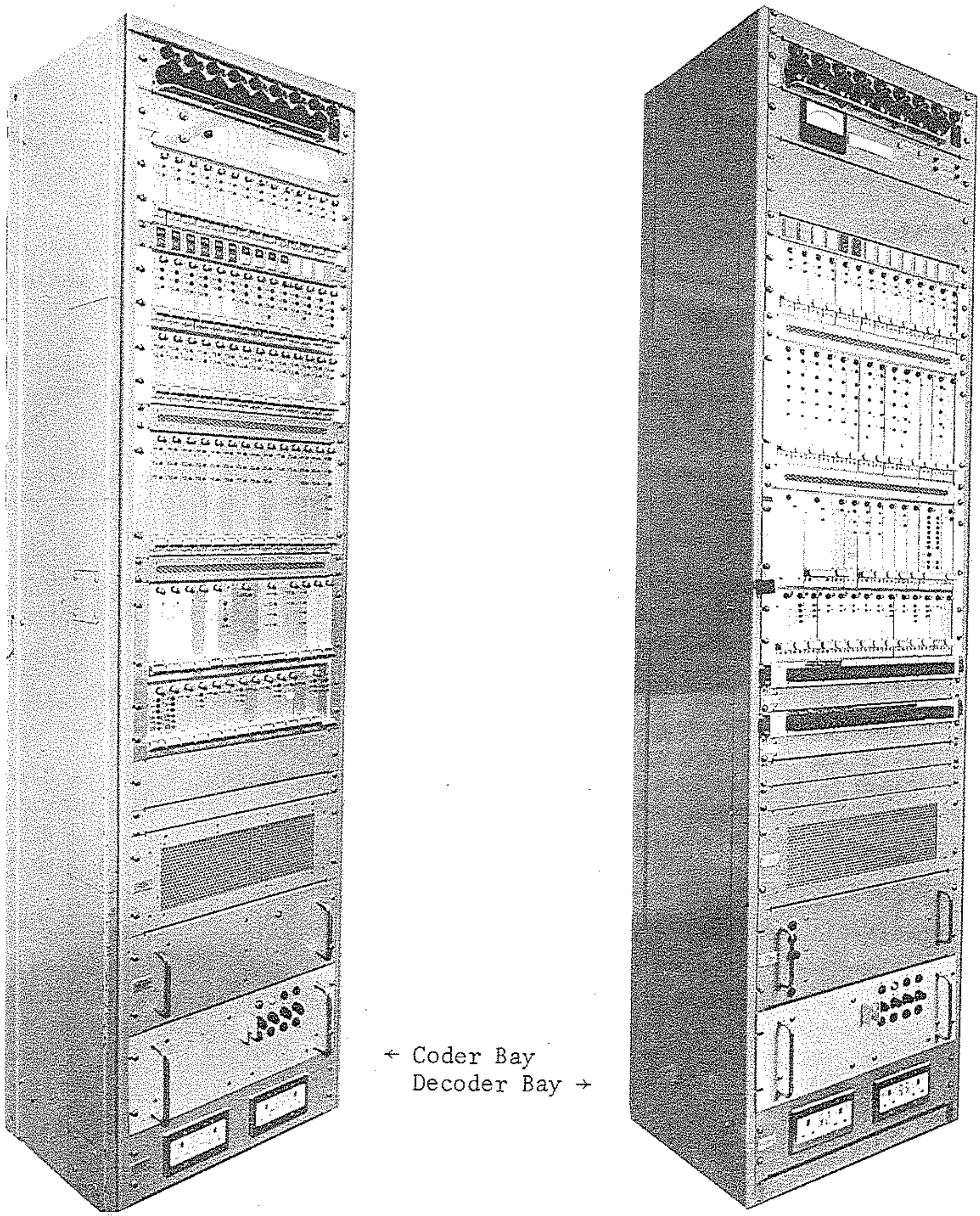
BBC 13-channel PCM System for High-quality Sound Distribution



← Coder Bay
Decoder Bay →

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The transmission of stereo signals from studios to transmitters requires high-performance circuits capable of conveying two audio signals over hundreds of miles without risk of variation in signal level or transit time; the quality of the stereo image is seriously damaged by small variations which have no perceptible effect on mono signals.

The BBC has been experimenting for more than ten years with digital techniques applied to audio signals; the 'sound-in-syncs' system for television sound distribution was the first practical outcome and the same basic techniques, using pulse-code modulation, have now been further developed to provide a multi-channel system for distributing stereo and mono signals to radio transmitters. The system carries 13 high-quality audio signals on one wide-band analogue circuit of a type suitable for monochrome television. Because of the intrinsic stability and accuracy of the digital approach, pulse-code modulation is ideally suited to the needs of long-distance stereo circuits.

The system

This system employs a pulse-code modulation time-division-multiplex method to provide up to 13 high-quality music circuits (each with a 15kHz bandwidth and signal-to-weighted-noise ratio of approximately 70dB). The equipment comprises Analogue-to-Digital Converters (ADCs), Digital-to-Analogue Converters (DACs), a multiplexer and a de-multiplexer, together with error-protection, data and control units.

Each audio input is sampled at a rate of 32kHz and converted into a series of 13-bit words. Pairs of channels can be used together for stereo transmissions.

The 13 coded audio channels, together with framing information, parity bits and a data channel, are combined in the multiplexer, the output of which is a 6.336Mbit/s serial bit-stream.

At the receiving terminal, the decoder converts the PCM signals back to the original analogue form, after demultiplexing and error concealment.

A 14th channel is used both to ensure that the timing circuits in the decoder are in synchronism with those in the coder and to carry switching instructions to equipment at the receiving terminals.

Block diagrams of the PCM coder and decoder are shown below.

PCM coder

Although the PCM system has a capacity of 13 audio channels, not all channels are being equipped for initial use in the BBC. Any remaining channels are fitted with dummy data generators to ensure continuity of the bit-stream output. Channel 14 is a framing-and-data channel.

Each audio input signal is fed through a 15kHz low-pass filter to a limiter. For stereo, pairs of limiters are interconnected in such a way that the gain of the two channels is varied at the same rate and in the same sense. The limiters operate only when the peak input signal-level exceeds +10dBm0, i.e. there is a 2dB margin before limiting occurs. The output of the limiter is passed through a second 15kHz low-pass filter to an ADC.

In the ADC, which is of the ramp-and-counter type, the audio signal is sampled for 2.2µs at a rate of 32kHz. Each sample is converted to a 13-bit binary word, to which is added a 14th bit for parity-checking. The logic level of this parity bit is determined by the five most-significant bits of the 13-bit word. This parity bit is used in the monitoring process described below. The output of each channel ADC is passed to a multiplexer circuit.

The purpose of the multiplexer is to carry out a time-division-multiplex process on the fourteen inputs. The 14-bit binary words on each of the fourteen inputs of the multiplexer are selected in turn and converted into a serial output, the multiplexer output being a continuous stream of pulses transmitted at a rate of 6.336Mbit/s.

The bit-stream output is passed through a filter having a sine-squared impulse response and thence to the output amplifier.

Timing of the coding process depends upon a 12.672MHz crystal oscillator in a clock unit. Appropriate switching waveforms are generated internally and distributed to the various sub-units as required.

PCM decoder

At the decoder the bit-stream input is applied to an input processing unit which is designed to give optimum discrimination between logic-0 and logic-1 levels in the presence of noise and distortion. The reconstituted output from this unit is taken to the clock-synchronising unit.

To ensure correct decoding, the timing signals generated in the decoder must be locked to the incoming bit-stream. This is achieved in the clock-pulse regeneration and synchronising units. The received framing-and-data channel carries a framing pattern, which is a fixed 9-bit sequence (and a single-bit frame-identification pulse). The incoming framing pattern is compared with a sequence held in the decoder, and decoding is initiated when coincidence occurs between the two patterns. If any timing differences exist, a voltage is developed which is used to vary the frequency of a crystal-controlled clock oscillator to ensure that the frames are correctly synchronised.

Locally-generated clock pulses are fed to a de-multiplexer to derive shift pulses which are used to route the individual 14-bit pulse groups to the correct channel DACs.

Each channel DAC re-forms the audio samples, which are then fed through a 15kHz low-pass filter to an output amplifier.

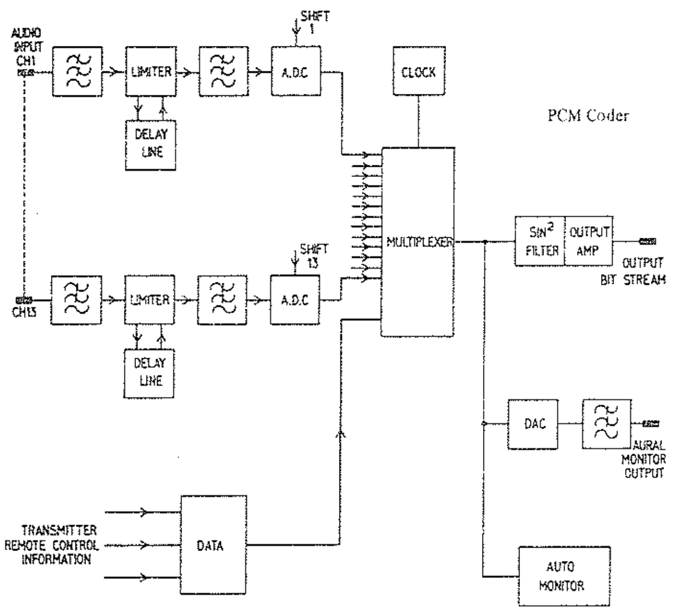
A separate data-decoder unit is fed from the clock-pulse regenerator and synchronising units. This decoder converts a 4-bit data number carried in the framing-and-data channel into switching signals used for controlling equipment at the receiving terminals. The data channel carries 16-bit messages which are split into 4-bit groups for transmission.

Monitoring

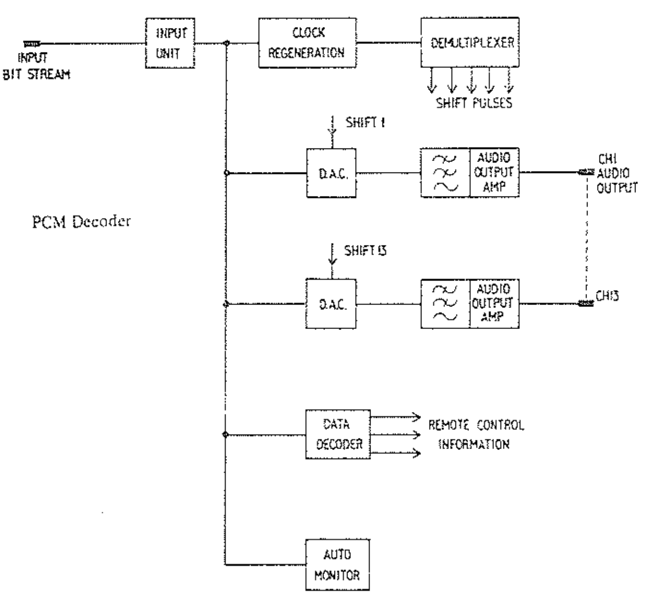
At both coder and decoder, a 5-bit ADC, which generates only the first five bits of the 13-bit sample is fed sequentially with audio signals. In the sending terminal the audio is derived from the limiter, and in the receiving terminal from an output-audio point. The 5-bit samples are compared with the five most-significant bits of the corresponding samples in the bit-stream. If a significant number of differences are detected an alarm is given.

The coder is also fitted with a de-multiplexer and a 13-bit DAC which can be selected manually to any channel for aural monitoring.

At the decoder, each DAC performs a parity check and, in the event of a failure, repeats the previous valid sample. A measurement of this error rate is made in each DAC, and if this becomes excessive, a mute may be applied to the audio output. The incidence of parity failure in each channel is conveyed to a central point where an assessment of overall error rate is made; should this become excessive, or if frame-synchronisation is lost, a mute is applied to all channels.



PCM Coder



PCM Decoder