Digitization and Encoding for Transmission in PCM

Digitization as part of the PCM process

In conventional PCM, the analog signal may be processed (e.g., by amplitude compression) before being digitized. Once the signal is digitized, the PCM signal is usually subjected to further processing (e.g., digital data compression).

PCM with linear quantization is known as Linear PCM (LPCM).[1]

Some forms of PCM combine signal processing with coding. Older versions of these systems applied the processing in the analog domain as part of the A/D process; newer implementations do so in the digital domain. These simple techniques have been largely rendered obsolete by modern transform-based audio compression techniques.

- DPCM encodes the PCM values as differences between the current and the predicted value. An algorithm predicts the next sample based on the previous samples, and the encoder stores only the difference between this prediction and the actual value. If the prediction is reasonable, fewer bits can be used to represent the same information. For audio, this type of encoding reduces the number of bits required per sample by about 25% compared to PCM.
- Adaptive DPCM (ADPCM) is a variant of DPCM that varies the size of the quantization step, to allow further reduction of the required bandwidth for a given signal-to-noise ratio.
- Delta modulation is a form of DPCM which uses one bit per sample.

In telephony, a standard audio signal for a single phone call is encoded as 8,000 analog samples per second, of 8 bits each, giving a 64 kbit/s digital signal known as DS0. The default signal compression encoding on a DS0 is either µ-law (mu-law) PCM (North America and Japan) or A-law PCM (Europe and most of the rest of the world). These are logarithmic compression systems where a 12 or 13-bit linear PCM sample number is mapped into an 8-bit value. This system is described by international standard G.711. An alternative proposal for a floating point representation, with 5-bit mantissa and 3-bit radix, was abandoned.

Where circuit costs are high and loss of voice quality is acceptable, it sometimes makes sense to compress the voice signal even further. An ADPCM algorithm is used to map a series of 8-bit µ-law or A-law PCM samples into a series of 4-bit ADPCM samples. In this way, the capacity of the line is doubled. The technique is detailed in the G.726 standard.

Later it was found that even further compression was possible and additional standards were published. Some of these international standards describe systems and ideas which are covered by privately owned patents and thus use of these standards requires payments to the patent holders.

Some ADPCM techniques are used in Voice over IP communications.

Encoding for transmission

Pulse-code modulation can be either return-to-zero (RZ) or non-return-to-zero (NRZ). For a NRZ system to be synchronized using in-band information, there must not be long sequences of identical symbols, such as ones or zeroes. For binary PCM systems, the density of 1-symbols is called ones-density.[2]

Ones-density is often controlled using precoding techniques such as Run Length Limited encoding, where the PCM code is expanded into a slightly longer code with a guaranteed bound on ones-density before modulation into the channel. In other cases, extra framing bits are added into the stream which guarantee at least occasional symbol transitions.

Another technique used to control ones-density is the use of a scrambler polynomial on the raw data which will tend to turn the raw data stream into a stream that looks pseudo-random, but where the raw stream can be recovered exactly by reversing the effect of the polynomial. In this case, long runs of zeroes or ones are still possible on the output, but are considered unlikely enough to be within normal engineering tolerance.

In other cases, the long term DC value of the modulated signal is important, as building up a DC offset will tend to bias detector circuits out of their operating range. In this case special measures are taken to keep a count of the cumulative DC offset, and to modify the codes if necessary to make the DC offset always tend back to zero.

Many of these codes are bipolar codes, where the pulses can be positive, negative or absent. In the typical alternate mark inversion code, non-zero pulses alternate between being positive and negative. These rules may be violated to generate special symbols used for framing or other special purposes

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