Error Detection and Correction

The ability to store audio signals as digital numbers makes possible many types of processing which would be difficult or impossible with analog techniques. There are, however, problems associated with the storage and retrieval of digital audio signals. In the case of analog recorders, the main limitations are due to the physical constraints of the medium: noise, distortion, and dropouts. While digital recording systems do not suffer from noise or distortion, they are susceptible to dropouts. When a dropout occurs, the data cannot be read. The longer it lasts, the more data is lost. While analog systems also suffer from dropouts, it is unusual for the signal to be entirely corrupted: it merely loses amplitude and possibly some of the high frequency components. When it returns, it is still the correct signal. When a digital data stream is disrupted by a dropout, all data is lost and corruption of the signal is complete. Therefore, digital recording systems must either be able to insure complete data integrity (as is the case of solid-state memories like DRAM computer memory) or provide for the detection and correction of any data errors that might occur.

The simplest way of detecting errors is to simply add up the sum of a certain number of words and store the sum with the data. If the replayed data does not add up to the so-called check sum, the data has an error. This system may be able to detect errors, but it cannot usually reconstruct the original data. Another way to minimize the effect of data loss is to spread the bits of each word out between the bits of previous and subsequent words: the process known as interleaving. This reduces the likelihood that all the bits of any word are lost, making the job of reconstructing the data a bit easier. To further increase the ability to determine what the original data were after a dropout, more data may be added to the signal in a way in which it can be used to reconstruct the original data even if large amounts of data are lost. These methods use sophisticated mathematical processes to allow data reconstruction and are known collectively as error detection and error correction.

Interleaving

Interleaving is the process of distributing data from data words over wide areas of the tape. This is accomplished by mixing bits from several words in a fixed way along the tape. When a dropout occurs, it will damage the data from several adjacent bits, but probably only one or two bits from each word will be altered, instead of all of the bits from one or two words. This allows for better reconstruction of the original data. A further development of this scheme is cross-interleaving, which uses two distinct codes and separates them by a time delay, further insuring that the error will be correctable.

Parity

If we divide the data by a number and store the remainder, we can later verify that the data played back is correct by seeing if the remainder generated by the played back data is the same.

Checksum

A more useful type of error detection is to sum the data over some period and store the sum. This gives some ability to recover the original data if the checksum fails to match on playback.

CRCC

This stands for cyclic redundancy check code: a form of checksum. Data in blocks is divided by an arbitrary constant and the remainder is appended to the data. On playback, the data is again divided by the same constant: a remainder of 0 indicates valid data. This scheme works best for burst errors (errors involving short periods of data disruption).

Reed-Solomon Code

This scheme uses Galois fields (number sets which when raised to powers or arithmetically combined always generate another member of the field.) This allows regeneration of data since only specific values can be contained in the original data.

CIRC

Cross-interleaved Reed-Solomon coding: used in CD coding. Uses two Reed-Solomon codes with an interleave between.

Coding and Modulation Schemes for Digital Audio

Modulation

Modulation refers to the combination of two or more signals into a composite that conveys information from both. Examples with which we are familiar include amplitude (AM) and frequency (FM) modulation used in radio communications. If we were to record raw audio data directly to tape, we would need an enormous bandwidth, since each bit would need to be coded as a distinct symbol with an on and off, each requiring a polarity reversal in the magnetic medium. We also need to transfer the clock information to keep everything synchronized. A method of reducing the number of distinct transitions is required, preferably one that also contains information about where in the bit stream a data word begins. Also, consider the case of a stream of consecutive 1's...this would provide a constantly high voltage, and consequent magnetic polarization, which would act like a DC signal and could eventually polarize the magnetic heads, as did the dc bias used in early analog magnetic recorders. Our modulation scheme should guarantee that DC does not appear in the signal.

Pulse code modulation

The basic coding scheme used in digital audio is pulse code modulation: signal amplitudes are measured in 16-bit A/D converters and sent out as serial data in which a logical one is coded by a positive voltage and a logical zero is coded by a zero (or negative) voltage. This scheme does not directly convey clock information when sequential bits do not change, nor does it allow for error detection, so further modulation is necessary.

Eight-to-ten modulation

Many coding schemes exist, and several are employed in computer disk drives and digital audio systems The DAT uses eight-to-ten modulation, where 8-bit bytes are encoded as 10-bit symbols uniquely assigned to represent the possible 8-bit bytes. This allows the choice of symbols with an exact balance of 1's and 0's, eliminating DC from the system. It also maintains a relatively low channel bandwidth by eliminating sequences of alternating 1's and 0's. The limited wavelength range generated by this modulation code also allows the signal itself to erase an underlying signal, thus eliminating the need for an erase head.

Eight-to-fourteen modulation

The compact disk uses a different modulation scheme. Instead of 8-10 modulation, the CD uses 8-14 (EFM) modulation. EFM maximizes the number of transitions possible with an arbitrary pit and land length, determined by the wavelength of the laser light used to read the data. (On CDs, ones are coded as edges of pits and zeros as pits and lands.) EFM modulation allows high-density recording by reducing the effective minimum wavelength below the resolving power of the laser.

Digital Audio Transmission

Digital audio signals can be routed between devices much like analog signals, although the actual signal is digital. Two commonly used protocols are used for stereo digital signal transfer: the **AES/EBU**(AES3) and **S/PDIF**(IEC 958 type II) systems. The AES/EBU (Audio Engineering Society/European Broadcast Union) system uses a differential (balanced) bit-serial high voltage (3-10 V) data stream that contains audio and other (sub-code) data of a specific form. The S/PDIF (Sony/Philips Digital Interface Format) system uses unbalanced low-voltage (0.5 V) bit-serial code with slightly different sub-code definitions. The AES/EBU protocol uses XLR connectors and is considered a professional audio format, while the S/PDIF system (also known as IEC-958 type II) uses RCA connectors and is intended as a consumer format. Both formats send data at almost 3 Mbits/sec, so special cabling must be used for long distances.

Because these systems are “self-clocking” (that is, they carry sample clock information as well as audio and sub-code data), special care must be taken to prevent jitter, drift in the clocking accuracy, from degrading system performance. Cabling of the proper impedance must be employed to keep internal electrical reflections from occurring in the cabling and confusing the D/A converter. Both the S/PDIF and AES/EBU systems expect properly terminated connections: the impedance of the input must be matched to the system impedance: 75 ohm for S/PDIF and 110 ohm balanced for AES/EBU.

In addition to the stereo digital interface formats, there are several multi-channel formats. With the ADAT came the 8-channel **ADAT lightpipe** protocol for digital audio interchange. This protocol uses optical fiber to transmit 8 channels in one direction, so that input and output are transmitted separately. The ADAT lightpipe is self-clocking and can transmit up to 24-bit words. Tascam introduced their multi-channel protocol with the DA-88. Called **TDIF**, this protocol is bidirectional and carries 8 channels and a clock over wires contained in a single cable for both directions. The **MADI** (AES10) Multi-channel Audio Digital Interface protocol allows up to 56 channels of digital audio to be transmitted on coaxial cable up to 150 feet. MADI is limited to 48 kHz sample rates, although higher rates may be implemented by sharing channels. MADI is unidirectional, requiring two cables plus a clock for bidirectional connection. If optical transmission is used, the MADI system may be used over distances up to 2 km. A new multi-channel protocol from Yamaha is the **mLan** system, using FireWire hardware interfaces to send multi-channel digital audio and/or MIDI data. Up to 100 audio channels may be transmitted over an IEEE-1394 cable. Through licensing, Yamaha hopes to encourage other manufacturers to adopt mLan as a standard way of interconnecting musical instruments and audio equipment over long distances using repeaters.