

Digital Audio Systems

While analog audio produces a constantly varying voltage or current, digital audio produces a non-continuous list of numbers. The maximum size of the numbers will determine the dynamic range of the system, since the smallest signal possible will result from the lowest order bit (LSB or least significant bit) changing from 0 to 1. The D/A converter will decode this change as a small voltage shift, which will be the smallest change the system can produce. The difference between this voltage and the voltage encoded by the largest number possible (all bits 1's) will become the dynamic range.

This leads to one of the major differences between analog and digital audio: as the signal level increases, an analog system tends to produce more distortion as overload is approached. A digital system will introduce no distortion until its dynamic range is exceeded, at which point it produces prodigious distortion. As the signal becomes smaller, an analog system produces less distortion until the noise floor begins to mask the signal, at which point the signal-to-noise ratio is low, but harmonic distortion of the signal does not increase. With low amplitude signals, a digital system produces increasing distortion because there are insufficient bits available to accurately measure the small signal changes.

There is a difference in the type of interference at low signal levels between analog and digital audio systems. Analog systems suffer from thermal noise generated by electronic circuitry. This noise is white noise: that is, it has equal power at every frequency. It is the "hiss" like a constant ocean roar with which we are so familiar. The low-level noise generated by digital systems is different: it is correlated with the signal because it is really a form of signal distortion produced by quantizing errors and has a distinctly unpleasant sound: something like a "gritty" or "grainy" sound. Most listeners find digital noise to be less acceptable than a comparable amount of analog white noise. Do we need to use noise reduction with digital systems as we found necessary on analog recorders? Well, the answer is yes, sort of. As we will later see, there are digital techniques that may be used to reduce the undesirable low-level distortions without processing the signal as we did for analog noise reduction.

Real A/D converters:

Most of the trouble with digital systems is involved in the transition from analog to digital and back again. The process must be perfectly accurate and very fast in order to adequately capture and reproduce the analog signal. Recent developments in integrated circuit manufacturing have led to great improvements in the conversion process and significant improvement in the sound of digital audio systems. Much of the improvement has dealt with increased accuracy of the conversion process by using more bits: while some systems, notably the compact disc, still only store 16 bit words, modern converters quantize to 24 bits, at least "on paper". This helps because most of the distortion produced in the A/D conversion results from the non-linearities associated with the lowest-order bits in the converter. There is an inherent limit to the accuracy of quantization determined by the least significant bit: there is always 1/2 LSB uncertainty due to the size of the smallest signal quantizable. The use of modern 24-bit converters guarantees linearity to at least 20 bits and most systems now store 24-bit audio data directly, though the lowest several bits may not be linearly converted. Floating point representation of audio data is popular, but converters still use fixed-point techniques to make the conversions.

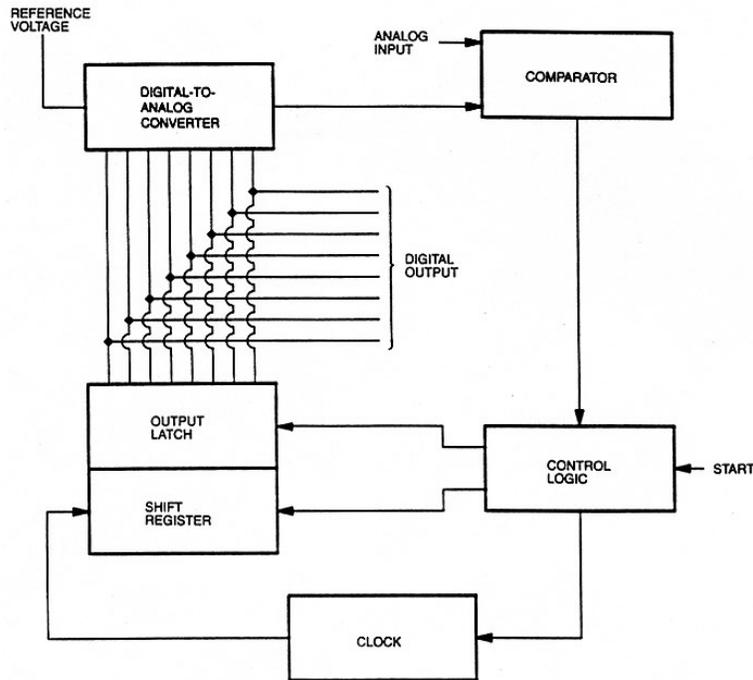


Figure 1: Successive approximation register A/D converter

A simple practical audio A/D converter is known as the **successive approximation register (SAR)**. This circuit works by setting a register and converting the value to analog via a D/A converter. The analog output from the register is continuously compared with the input voltage while the register is repeatedly incremented or decremented by control logic until the two are equal, at which time the digital number in the register is used as the value of the analog input voltage. In general, n D/A conversions must occur per sample period in an n -bit converter, so the clock must run at nearly 1 MHz in a one channel 16 bit converter.

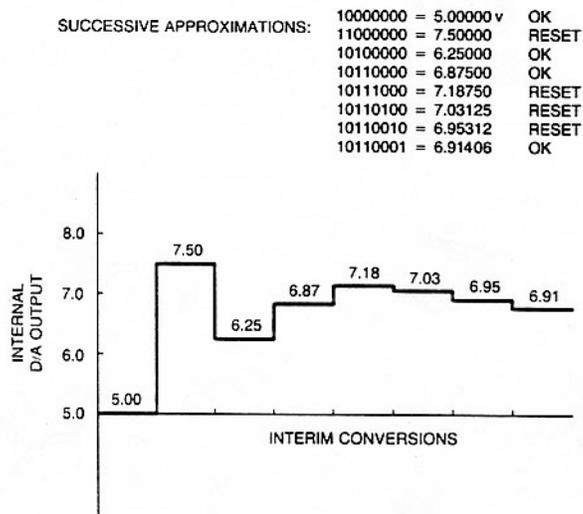


Figure 2: SAR series of approximations

In older systems, the stereo channels were multiplexed through a single A/D converter which had to run at double speed. This also resulted in a small time shift between the channels. In order to guarantee that the

analog input voltage doesn't change while the conversion is taking place, the analog voltage must be held constant while the A/D process takes place. This is done with a circuit called a **sample-and-hold**. It stores a single voltage value on a capacitor while the conversion proceeds and then reads in a new value for the next conversion. Using dual sample-and-hold circuits clocked together, the small time delay between multiplexed channels could be eliminated. Current A/D converters do not need to multiplex.

A more current type of A/D converter is known as a 1-bit or sigma-delta type converter, now used for Direct Stream Digital (DSD) recording as well as for conversion to multi-bit words. This converter works by very rapidly comparing the difference between the input and summed previous samples taken at a very high rate (oversampled), quantized as either 0 (difference is positive) or 1 (difference is negative). If the difference is 0, the pulse alternates between 1 and 0. The series of pulses can be summed to generate the multibit codes actually used for data storage, or in the case of DSD, the single-bit bitstream itself is stored. This technique trades the inherent linearity of a one-bit quantizer for the greatly increased sample rate necessary. An additional benefit of this type of converter is the ability to use digital signal processing to shape the noise generated in the process in such a way as to place it predominantly in high frequencies which are inaudible and easily removed by simple analog filtering in the process known as noise shaping. Once conversions are made, the data may be transmitted inside a computer or distributed between devices using digital data interchange formats.

Digital Audio Transmission:

Digital audio signals can be routed between devices much like analog signals, although the actual signal is digital. In addition to the audio data, the clock information needs to be transmitted to the receiver in order to preserve the sample timing. 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 which contains audio and other (subcode) 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 subcode 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 low capacitance 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 subcode 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 or other data receiver. 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 multichannel 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. This interchange protocol has remained in use even though the ADAT itself is long out of production. It provides an inexpensive way to connect 8 channels of digital audio at sample rates of up to 48 kHz (or 96 kHz by channel-splitting). Tascam introduced their own multichannel protocol with the DA-88. Called **TDIF**, this protocol is bidirectional and carries 8 channels and a clock over multiple pairs of wires contained in a single cable for both directions. The TDIF protocol for each pair is similar to the AES/EBU stereo format. The **MADI** (AES10) Multichannel 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 proprietary multichannel protocol from Yamaha is the **mLan** system, using FireWire hardware interfaces to send multichannel 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. More recently, the **Dante** interface protocol has been developed by the Australian company Audinate. Dante uses the standard ethernet protocol to transmit digital audio over existing networks. The stream may be recorded to a connected computer without the use of an audio interface, simply using the built-in ethernet port and software. Many equipment manufacturers support the Dante interface protocol. The advantage of existing Ethernet infrastructure and data reliability now possible make systems like Dante likely to replace proprietary multichannel hardware interchange formats.

Consumer Digital Tape Recorders:

While digital tape is largely obsolete, the history of its development reveals much about technological advancements we now take for granted. The first commercially available digital recorders made use of videotape recorders as the actual storage medium. The PCM (pulse code modulators) converters digitized the audio signal and converted it to a video signal. These recorders were the Sony PCM F-1 and the later PCM 501, 601, and 701 processors. They allowed 14 or 16 bit quantization with better error correction in the 14 bit mode. The main problem with these units was due to poor tracking on the VCR which led to dropouts and adjustment of the finicky VCR tracking control was frequently required. Only the 601 offered digital I/O. PCM converters are rarely used today and are sometimes available used for a few dollars.

The DAT (digital audio tape) recorder was developed later, incorporating both the A/D and D/A converters and the actual tape transport mechanism into a single unit. The DAT offered better machine to machine matching and greater simplicity of operation. DAT recorders offer better error correction and automatic tracking, making them preferable to PCM/video recorders. DATs also offer subcode indexing not available on PCM converter systems. Most DAT recorders were 16-bit machines, although there were 24-bit variants. DAT machines are still sometimes encountered in studios for playing older recordings, but reliability issues caused their rapid replacement by more modern storage media like flash memory.

Extending the basic technology of the DAT, rotary-head multitrack digital recorders were developed. The Alesis ADAT and the Tascam DTRS recorders used video transports with rotary heads optimized for digital audio to create video-tape-based recorders capable of 8-channel simultaneous recording and playback. These machines are still in limited use and have some advantages, since they may be linked easily to provide large track counts and may dub with no signal degradation. Tapes may easily be cloned and shared between studios, making collaborative work easy. The ADAT machines managed to ultimately provide 20-bit recording at 48 kHz, while the DTRS format ultimately allowed 24-bit recording at up to 48 kHz sample rates and higher rates can be accomplished by “bit-splitting”, effectively doubling the sample rate at the expense of halving the track count. Although rotary-head technology was an interim solution to the problem of digital recording, it provided users with adequate resources to record multichannel audio for many years. Computer-based recording systems have replaced rotary head recorders in recent years.

Other interim recording technologies made use of perceptual coding to reduce the amount of audio data required for storage, resulting in systems like the MiniDisc, which used magneto-optical disks which were read optically. MiniDiscs could be re-written, so punch-ins and overdubbing are possible. Since they employed psychoacoustic principles to reduce the data actually stored, they were not a good choice for master recordings,

which may later be altered through equalization or other processing. Stereo recorders using CD-R media are still available as stand-alone recorders without resorting to data reduction techniques, though their popularity is in decline as media in general are losing ground to streaming of audio sources.

Professional Digital Recorders

Professional stationary-head, reel-to-reel digital recorders were around for some time, with Sony, Mitsubishi, and other manufacturers making 24, 32, and 48-track machines. Different systems developed: DASH (Digital Audio Stationary Head) and ProDigi competed for users. Ultimately, the DASH format won out, but currently these expensive machines lost popularity to much cheaper digital systems like the ADAT and hard-disk systems. They could record up to 48 tracks on 1/2" tape and could be edited manually, but all of this is now possible on cheaper and simpler systems. For example, a Sony 3348HR 24-bit/48 channel machine cost over \$250,000. Six DA-78HR machines were under \$15,000 and the modular nature of these machines made maintenance less of a problem. Interesting variants included the Yamaha DMR/DRU series, which used stationary-head technology and proprietary 8mm tape cassettes to record 8 tracks of 20-bit digital audio at up to 48 kHz. The DMR-8 included 24 track moving-fader automation, full SMPTE sync, and 3 built-in digital processors in a professional porta-studio system. Since a full system would cost over \$50,000, it was abandoned when the ADAT hit the market. While many of these systems are still available on the used market, their utility is marginal when compared to computer-based systems now considered standard. Digital audio has advanced beyond the stage of dedicated hardware in favor of personal computers with either built-in or external audio conversion. The era of digital tape was short-lived, for good reason.

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