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Conversion

D/A

Two main methods of digital to analog conversion exist, the first of which depends on the control of binary-weighted currents and a switching mechanism. The binary-weighted current model can also be subdivided into two categories: weighted resistors and non-weighted resistors. Both weighted current methods depend on each bit in the PCM data word to correspond to an increasingly larger amount of current to flow. Each bit also corresponds to an analog switch which allows or disallows the flow of this current depending on the status of each bit. A one simply means that the corresponding switch is closed and the weighted current is allowed to flow, contrasted by a bit status of zero that relates to its switch being open and current not be able to flow. Practically speaking, weighted currents can be achieved by 1) sending the same voltage across resistors in parallel, each being twice the value of the next (and its associated current flow therefore being half) and then summing all currents with an operational amplifier, or 2) sending the same voltage across equal resistors in parallel but setting up a resistance chain in series between each bit switch before it reaches the op amp; the MSB, for example, will be closest to the op amp and pass through only one resistor to achieve high current value while the LSB switch will be at the end of the chain and have its current reduced by the string of series resistors.

The second method for achieving an analog current from PCM data uses timing information to reproduce the sample. Simply, the amount of time current is allowed to flow into an integrator is controlled by the bits themselves. The word value of the PCM data is feed to a clock which lets current build up for a very accurate amount of time correlating to the PCM value. With higher bit levels, it is necessary to split the word up into two halves so that the clock can achieve accuracy without resulting in a super high frequency. High-order and low-order current sources correlate to the MSbyte and the LSbyte, respectively. This process will reduce the clock frequency by two to the power of half the total wordlength and thus make it practical for higher bit level applications.

A/D

There are many ways to convert analog signals to digital information, each with its own benefits and disadvantages. The flash converter is one very quick and efficient means of performing this operation. A reference voltage (variable with resistance) is provided down a cascading string of resistors attached to comparators. The input voltage is fed to every comparator and then related to the reference voltage (which obviously decrease down the line of comparators due to voltage drops across the resistors). A comparator is needed for each quantizing interval, giving a logic (T/F) output. The comparator outputs can be quickly converted to PCM audio. The advantage of flash converters is there immediate speed and accuracy, as well as ability to have a variable input threshold. The disadvantage to flash converters is their unweildiness of having a comparator for every quantization level, especially with higher bit rates. The low-bit applications of delta-sigma modulation and oversampling, though, are well suited for such conversion techniques.

Another common A/D conversion technique is that of successive approximation. With this method, the input voltage is compared to the reference voltage for each bit, starting with the MSB. If the input voltage is greater than the voltage level for this bit, the bit becomes 1 and its reference voltage is retained and added to the next reference voltage; if the input voltage is lesser, the bit becomes 0 and the input voltage gets compared to the next less significant bit, and so on down the line. The main advantage of such a method is that component complexity is reduced, needing only as many comparators as the bit level. Unfortunately, the droop in the input voltage from capacitor loss becomes very severe since the LSBs are computed last (when droop is obviously at its worst).

The third method of conversion which I would like to discuss is probably the most complicated, that of using a dual-current source DAC in a feedback system. This technique uses a ramped integrator (similar in function to the ramp intergrator of D/A) to measure the input voltage. Basically, two currents are used to discharge the capacitor in which the input voltage is stored. The first current determines the first half of the bits (first 8 MSBs in a 16 bit word) while the second current determines the LSBs. The time it takes the first current to discharge the capacitor below a reference level provides the quantization level for the first half of the word. Similarly the time it takes the second current to fully discharge the rest of the capacitor determines the second half of the digital word. This method requires an extremely accurate timing mechanism and discharging currents. As well, dual-source DAC is subject to the same effects of droop as successive approximation.