Applications of PWM II

Voltage regulation

PWM is also used in efficient voltage regulators. By switching voltage to the load with the appropriate duty cycle, the output will approximate a voltage at the desired level. The switching noise is usually filtered with an inductor and a capacitor.

One method measures the output voltage. When it is lower than the desired voltage, it turns on the switch. When the output voltage is above the desired voltage, it turns off the switch.

Audio effects and amplification

PWM is sometimes used in sound (music) synthesis, in particular subtractive synthesis, as it gives a sound effect similar to chorus or slightly detuned oscillators played together. (In fact, PWM is equivalent to the difference of two sawtooth waves. [1]) The ratio between the high and low level is typically modulated with a low frequency oscillator, or LFO. In addition, varying the duty cycle of a pulse waveform in a subtractive-synthesis instrument creates useful timbral variations. Some synthesizers have a duty-cycle trimmer for their square-wave outputs, and that trimmer can be set by ear; the 50% point was distinctive, because even-numbered harmonics essentially disappear at 50%.

A new class of audio amplifiers based on the PWM principle is becoming popular. Called "Class-D amplifiers", these amplifiers produce a PWM equivalent of the analog input signal which is fed to the loudspeaker via a suitable filter network to block the carrier and recover the original audio. These amplifiers are characterized by very good efficiency figures (≥ 90%) and compact size/light weight for large power outputs. For a few decades, industrial and military PWM amplifiers have been in common use, often for driving servo motors. They offer very good efficiency, commonly well above 90%. Field-gradient coils in MRI machines are driven by relatively-high-power PWM amplifiers.

Historically, a crude form of PWM has been used to play back PCM digital sound on the PC speaker, which is driven by only two voltage levels, typically 0 V and 5 V. By carefully timing the duration of the pulses, and by relying on the speaker's physical filtering properties (limited frequency response, self-inductance, etc.) it was possible to obtain an approximate playback of mono PCM samples, although at a very low quality, and with greatly varying results between implementations.

In more recent times, the Direct Stream Digital sound encoding method was introduced, which uses a generalized form of pulse-width modulation called pulse density modulation, at a high enough sampling rate (typically in the order of MHz) to cover the whole acoustic frequencies range with sufficient fidelity. This method is used in the SACD format, and reproduction of the encoded audio signal is essentially similar to the method used in class-D amplifiers.

Companding

In telecommunication, signal processing, and thermodynamics, companding (occasionally called compansion) is a method of mitigating the detrimental effects of a channel with limited dynamic range. The name is a portmanteau of compressing and expanding.

While the compression used in audio recording and the like depends on a variable-gain amplifier, and so is a locally linear process (linear for short regions, but not globally), companding is non-linear and takes place in the same way at all points in time. The dynamic range of a signal is compressed before transmission and is expanded to the original value at the receiver.

The electronic circuit that does this is called a compandor and works by compressing or expanding the dynamic range of an analog electronic signal such as sound. One variety is a triplet of amplifiers: a logarithmic amplifier, followed by a variable-gain linear amplifier and an exponential amplifier. Such a triplet has the property that its output voltage is proportional to the input voltage raised to an adjustable power. Compandors are used in concert audio systems and in some noise reduction schemes such as dbx and Dolby NR (all versions).

Companding can also refer to the use of compression, where gain is decreased when levels rise above a certain threshold, and its complement, expansion, where gain is increased when levels drop below a certain threshold.

The use of companding allows signals with a large dynamic range to be transmitted over facilities that have a smaller dynamic range capability. For example, it is employed in professional wireless microphones since the dynamic range of the microphone audio signal itself is larger than the dynamic range provided by radio transmission. Companding also reduces the noise and crosstalk levels at the receiver.

Companding is used in digital and telephony systems , compressing before input to an analog-to-digital converter, and then expanding after a digital-to-analog converter. This is equivalent to using a non-linear ADC as in a T-carrier telephone system that implements A-law or μ-law companding. This method is also used in digital file formats for better signal-to-noise ratio (SNR) at lower bit rates. For example, a linearly encoded 16-bit PCM signal can be converted to an 8-bit WAV or AU file while maintaining a decent SNR by compressing before the transition to 8-bit and expanding after a conversion back to 16-bit. This is effectively a form of lossy audio data compression.

Many of the music equipment manufacturers (Roland, Yamaha, Korg) used companding for data compression in their digital synthesizers. This dates back to the late 80's when memory chips would often come as one the most costly parts in the instrument.

Manufacturers usually express the amount of memory as it is in the compressed form. i.e. 24MB waveform ROM in Korg Trinity is actually 48MB of data. Still the fact remains that the unit has a 24MB physical ROM. In the example of Roland SR-JV expansion boards, they usually advertised them as 8MB boards which contain '16MB-equivalent content'. Careless copying of the info and omitting the part that stated "equivalent" can often lead to confusion.