Chapter 12

CVSD modulation of audio signals

TEC/NOT/015

This paper introduces Continuously Variable Slope Delta (CVSD) modulation.

The following topics are discussed:

12.1 Overview

This paper introduces CVSD modulation as a method of transmitting audio signals in PCM systems. In particular CVSD is discussed as a variation of Sigma-Delta (ΣΔ) modulation wherein the Delta (Δ) varies as a function of the signal being encoded. Also the issue of reproducing the audio signal is briefly discussed.

Quite often it is desirable to record (or transmit) the pilot's (and co-pilot's) voice along with the instrumentation data to ease synchronization and reduce the number of, or simplify, the storage devices or transmitters required.

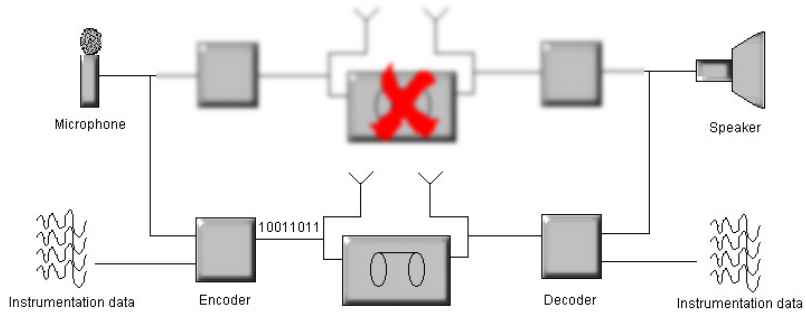

Figure 12-1: Embedding audio signal in instrumentation data

How many bits per second are required?

- Telephone quality transmissions require 8-bit samples at approximately 8 ksps or 64 kbps.
- Compact disk quality transmissions require approximately 10 times this bit-rate (16-bit samples at approximately 40 ksps).
- Compression algorithms always require less bits but require processing and may be less tolerant of bit-errors.

One approach is to take advantage of the fact that audio signals have no DC component and that it is the size and rate of change that is of particular interest.

What follows is an introduction to Sigma-Delta (ΣΔ) modulation and then CVSD modulation (encoding).

12.2 Sigma-delta (ΣΔ) modulation

Sigma-Delta (ΣΔ) modulation is based on the fact that the signal has no DC component and simply transmits a 1 or a 0, depending on whether the signal has increased or decreased since the last sample. The encoder and decoder adds (or subtracts) a Delta **(**Δ**)** from a running value which is compared to the next sample depending on whether a 1 or a 0 is transmitted.

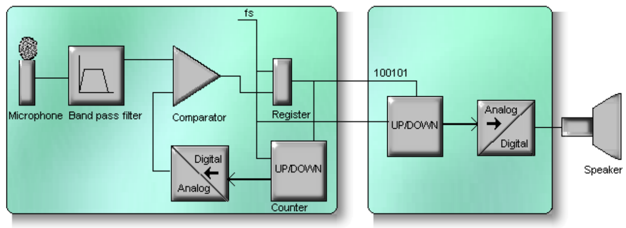

Figure 12-2: Basic implementation of Sigma-Delta ($\Sigma\Delta$) encoder/decoder system

The maximum rate of change for a sine wave of amplitude *A* and frequency *f* is at the zero crossing. If the signal is being sampled at a rate f_s then the maximum change in one sample period is:

$$S_{max} = 2 \cdot \pi \cdot A \cdot \frac{f}{f_s}$$

If telephone quality (8-bit) audio is required, the Delta **(Δ)** must be $1/2^8$ times the A/D input range, and signal frequencies up to 8 kHz are expected. Assuming the signal is close to the rail of the A/D (that is, the A/D range = ±A) then:

$$\Delta = \frac{2 \cdot A}{2^8} > 2 \cdot \pi \cdot A \cdot \frac{8000}{f_s} \Rightarrow 643kHz$$

In other words, for telephone quality, Sigma-Delta (ΣΔ) modulation requires compact disk quality bit-rates. At first, this may appear like a step backwards. However it is worth noting that if the amplitude was typically a quarter of the maximum (or less) and the frequency was typically 1 kHz (or less), then telephone quality would be obtained at a bit-rate of 32 kHz.

12.3 CVSD modulation

In a CVSD system the Delta (Δ) varies depending on the recent rate of change of the input signal. In particular, if the last three outputs from the encoder are 1 then this indicates that the signal is increasing faster then 1Δ per sample, so the Delta (Δ) is increased until a 0 is output.

As the smallest Delta (Δ) can be smaller than 1/256th of the input range (on the KAD/VDC/001 it is 1/1024), the CVSD modulator can have better dynamic performance then an 8 x 8 kbps system for slow or small signals.

However, for fast or large signals, the dynamic performance may not be as good as an 8 x 8 kbps system (hence the KAD/VDC/001 should not be used for tone or phase signals).

A quantitative analysis of CVSD modulation is beyond the scope of this paper, however the following statements may help:

In IRIG-106-98 Chapter 5, the following statement appears:

A qualitative test of CVSD with a tactical aircraft intercom system (ICS) yielded the following results:

- intelligible robotics sounding audio at 12 kbps
- good quality audio at 16 kbps
- audio quality did not significantly improve as the bit rate was increased above 32 kbps

In application note 607 from Intersil (see “References” on page 3) the following appears:

CVSD has better intelligibility then PCM when random bit errors are introduced during transmission.

12.4 CVSD and the KAD/VDC/001

The KAD/VDC/001 continuously samples two audio streams at a constant rate. Both channels need not be sampled at the same rate. The samples (comparator outputs) are then sent to a programmable serial-to-parallel converter. For each channel the bits per word (10/12/14/16) must be specified.

NOTE: A 16-bit word means that the comparator results of 16 samples are stored in one word. For example, 4 x 12-bit words would have exactly the same information as 3 x 16-bit words.

Words are therefore generated at a constant rate; there is an integral number of words per acquisition cycle (major-frame). Data sampling is simultaneous with respect to the acquisition cycle; the serial-to-parallel converter starts a new word at the start of an acquisition cycle.

While the Acra KAM-500 PCM encoders enable data to be inserted in a PCM stream at any location, Chapter 5 of IRIG-106-98 strongly recommends that digitized audio words be evenly spaced in PCM streams.

12.5 CVSD Decoding

Many telemetry ground stations have CVSD-to-audio cards; one implementation is illustrated in the following figure.

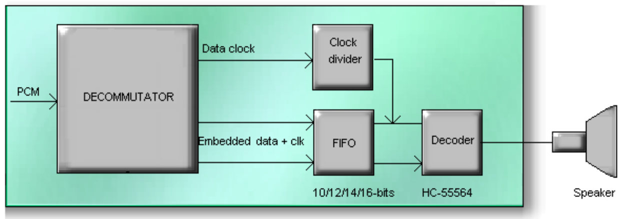

Figure 12-3: Key blocks required for CVSD demodulation

The CVSD data PCM decommutator is marked as an embedded stream and therefore gets clocked out in bursts one word long. If data is evenly spaced (in time) in the PCM stream, then the data can be clocked into a simple one word deep FIFO.

The data is clocked out of the FIFO at a rate determined by the formula:

$$f_s = \text{PCM\_BIT\_RATE} \cdot \frac{\text{CVSD\_BITS\_PER\_FRAME}}{\text{TOTAL\_BITS\_PER\_FRAME}}$$

Interfacing to these cards can be made simpler if the following rules (as recommended by IRIG-106-98 Chapter 5) are followed when defining the PCM frame:

- Space the CVSD words evenly (in time) throughout the data, this eases time correlation and FIFO design.
- Transmit the oldest sample bit first, this simplifies FIFO design.
- Ensure BIT_RATE/f_s is an integer, this simplifies the clock divider design.

The Acra KAM-500 allows users to follow these recommendations or not.

12.6 Conclusion

CVSD is a form of Sigma-Delta (ΣΔ) modulation where the Delta (Δ) changes depending on the signal. It enables digitization of audio signals at bit-rates as low as 9600 bps, therefore allowing the signals to be embedded among instrumentation data without a significant bandwidth cost. With each KAD/VDC/001 two audio channels can be thus encoded.

12.7 References

IRIG-106-98 Chapter 5

Application note 607

Intersil Corporation

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