Understanding PCM Coding

Introduction

The process of converting analog voice signals into Time Division Multiplexed (TDM) Pulse Code Modulated (PCM) format is described and illustrated herein. Application Note No. 570, "Understanding CODEC Timing", by D.J. Donovan is recommended reading as accompaniment to this application note.

Analog time varying voice input information is transmitted over two-wire (2W) pairs (channels) from subscribers. The PCM filter band-limits voice signals to 4kHz, one per channel, and removes power line and ringing frequencies. Research has shown that voice transmission band-limited to 4kHz has enough fidelity for telephony purposes.

Sampling

The process of converting filtered voice information into a digitized pulse train format begins with sampling the voice signal at uniform intervals. These intervals are determined by the Nyquist Sampling Theorem, which simply states that any signal may be completely re-constructed from its representative sampling if it is sampled at least twice the maximum frequency of interest. The telephone system, being a worldwide standard 8kHz sampling system, satisfies Nyquist, as all voice signals are band-limited to 4kHz. When the voice waveform is sampled, a train of short pulses is produced, each representing the amplitude of the waveform at the specific instant of sampling. This process is called Pulse Amplitude Modulation (PAM). The envelope of the PAM samples replicate the original waveform. Figures 1A thru 1D illustrate representative PAM samples for up to 24(30) individual voice channels in a μ-Law (A-Law) telephone system.

There are relatively large intervals between each PAM sample that may be used for transmitting PAM samples from other voice channels. Interleaving several voice channels on a common bus is the fundamental principle of Time Division Multiplexing (TDM). As the number of voice channels on the TDM bus increases, the time allotted to each sample is reduced, and bandwidth requirements increase (See Figure 1E).

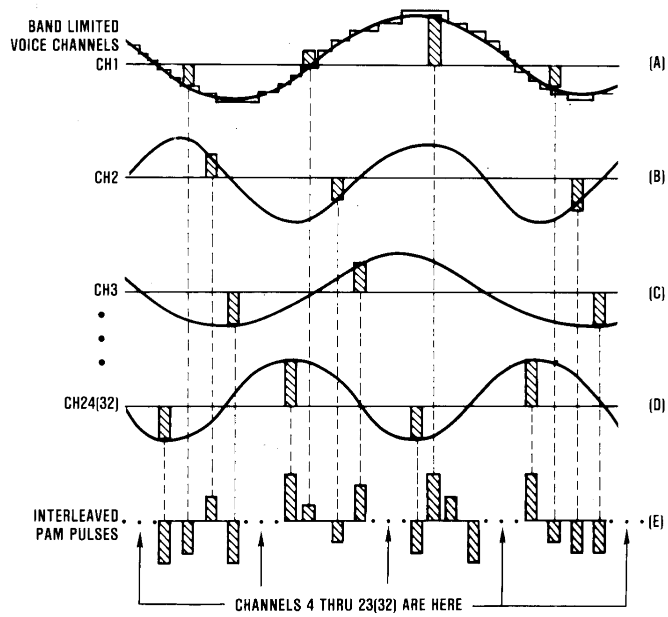

FIGURE 1. (A THROUGH E)

Quantizing

The PAM samples still represent the voice signal in analog form. For digital transmission, further processing is required. Pulse Code Modulation (PCM) is a technique used to convert the PAM samples to a binary weighted code for digital transmission. PCM coding is a two step process performed by the CODEC. The first step is quantization, where each sample is assigned a specific quantizing interval. The second step is PCM coding of the quantizing interval into an 8-bit PCM code word. Each is discussed in the text that follows.

Converting PAM samples to a digital signal involves assigning the amplitude of a PAM sample one of a whole range of possible amplitude values, which are divided into quantizing intervals. There are 256 possible quantizing intervals, 128 positive and 128 negative. The boundaries between adjacent quantizing intervals are called decision values.

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If PAM samples are uniformly quantized, there will be situations where several different amplitude values will be assigned the same quantizing interval during encoding. Then, during decoding, one signal amplitude value is recovered for each quantizing interval which corresponds to the midpoint of the quantizing interval. This results in small discrepancies that occur between the original waveform and the quantized approximation; i.e., infinite analog levels in the original waveform being assigned finite quantizing intervals. These discrepancies result in a quantizing noise or quantizing distortion, the magnitude of which is inversely proportional to the number of discrete quantizing intervals. These noise signals may be of the same order of magnitude as the input signal, thereby reducing the signal to quantizing noise ratio to an intolerable level. For this reason non-uniform quantization is used. Large signals need a smaller number of quantizing intervals, while small signals require a larger number of quantizing intervals. Such a non-uniform quantization process is defined as companding characteristics by both Bell and CCITT.

The PCM CODEC performs this non-uniform or non-linear quantization through µ-Law or A-law companding characteristics shown in Figure 2. This process enhances lower amplitude signals, to allow them to compete with system noise, and attenuates higher amplitude signals, preventing them from saturating the system. This form of signal compression results in a relatively uniform signal to quantization noise ratio, approaching 40dB for a wide range of input amplitudes. Also, the dynamic range approaches that of a 13(11) bit A/D or 80(66)dB for µ-Law (A-Law) companding. The digital realization of this companding process is obtained by a segment and chord piecewise linear approximation to a semi-logarithmic function.

Both the µ-Law and A-law companding characteristics are composed of 8 linear segments or chords in each quadrant. Within each chord are 16 uniform quantization intervals, or steps. With µ-Law, moving away from the origin, each chord is twice the width of the preceding chord, and each group of 16 uniform steps is twice the width of the preceding group. It is also referred to as the 15 segment characteristic. The first chord about the origin in the positive and the negative quadrant are of the same slope and are therefore considered one chord (chord 0). With A-law, the first two chords and step groups in each quadrant are uniform. Successive chords and steps follow the same pattern as µ-Law. A-Law is referred to as the 13 segment characteristic. The first two chords about the origin in the positive quadrant, and the first two chords about the origin in the negative quadrant are all of the same slope and therefore are considered one chord (chord 1). There are 64 uniform steps in chord 1, 32 positive and 32 negative. However, for purposes of encoding and decoding samples that fall into the quantization intervals in chord 1, a different 3 bit chord code (refer to Figure 3) is assigned for the first segment of 16 uniform steps closest to the origin and the next segment moving away from the origin. Chord 1 in A-Law is twice that of chord 0 in µ-Law.

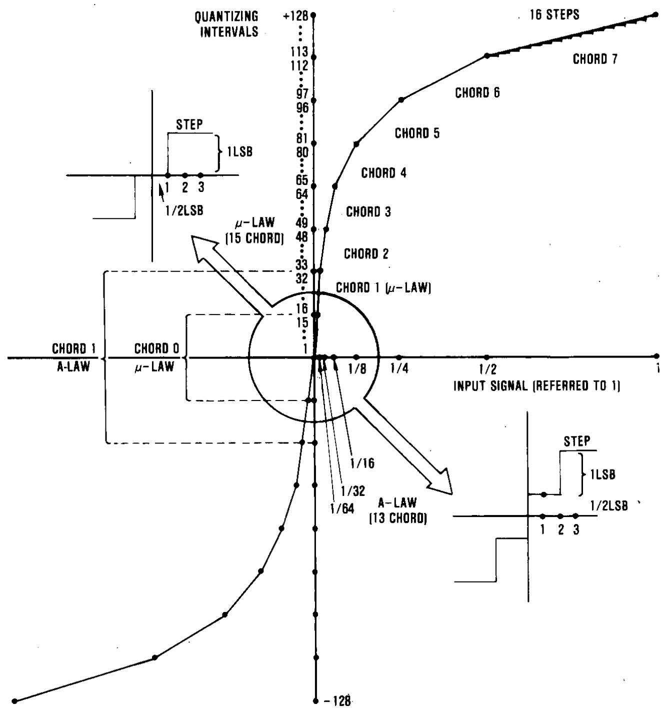

FIGURE 2.

The µ-Law companding characteristic is used primarily in North America and Japan, while A-Law is used primarily in Europe. The differences are minimal and are summarized below:

µ-Law

- Step sizes double for each successive chord
- Virtual edge = ±8159 units
- Input level = 3.172dBm0
- 2 codes for 0 input

A-Law

- Step sizes double for each successive chord after the second chord
- Virtual edge = ±4096 units
- Input level = 3.14dbm0
- No code for 0 input

The input level is determined with reference to the power level at the central office or ‘switch’. That point is referred to as the zero transmission level point (0TLP). All CODEC measurements must be translated to the 0TLP. The unit of translated level is the dBm0 (dB relative to 1mW referred to a transmission level of 0TLP).

There is no absolute voltage standard for the CODEC, however, a standard exists relative to full scale. The point at which the CODEC begins to clip is called the virtual edge. It is measured in normalized voltage units or steps, ±8159 steps for µ-Law and ±4096 steps for A-Law. If a PAM sample representing the peak of a voice input signal hits the virtual edge of a µ-Law system, it has a relative power of +3.172dBm0. The corresponding A-Law relative power is +3.14dBm0. These numbers are chosen to minimize intrinsic gain error at 0dBm0 and 1000Hz.

Encoding

The second stage of conversion to binary PCM data for transmission involves the coding of the 256 quantizing intervals assigned to the individual PAM samples into 8-bit binary words (7 data bits plus 1 sign bit). The MSB in each word is a polarity bit indicating a 1 for positive quadrant quantizing intervals, and a 0 for negative quadrant quantizing intervals. The next three bits represent the chord, and the last four bits identify the step within the chord. The 8-bit PCM word partitioning is illustrated in Figure 3.

BIT NUMBER	1	2	3	4	5	6	7	8	
BIT WEIGHT -MSB	±	2^6	2^5	2^4	2^3	2^2	2^1	2^0	LSB
	SIGN		CHORD			STEP			

A-Law and µ-Law coding about the origin differ. µ-Law defines two codes for 0V input while A-Law defines no code for 0V input (see Table 1). The two µ-Law zero codes represent a normal quantization step that is divided into halves by the y-axis of the companding curve (refer to Figure 2). These half steps represent the lowest resolvable signal of the µ-Law characteristic.

TABLE 1.

INPUT	BINARY EQUIVALENT	µ-LAW	A-LAW
+FULL SCALE	1111 1111	1000 0000	1010 1010
+CENTER	1000 0000	1111 1111	1101 0101
-CENTER	0000 0000	0111 1111	0101 0101
-FULL SCALE	0111 1111	0000 0000	0010 1010

Multiplexing and Transmission

Each 8-bit PCM word is transmitted in its respective time slot, which is assigned to each CODEC by the system controller (See Application Note 570). A number of PCM words may be transmitted consecutively from different channels, creating a PCM TDM signal for transmission. Each CODEC channel has an average data rate of 8K samples/sec x 8-bits = 64kbits/s. This means that within a 1/8kHz = 125µs period, 24(30) PCM words of 8 bits each are transmitted consecutively in a µ-Law (A-Law) system.

µ-Law Systems

For µ-Law systems, the bus format allows 24 groups, or timeslots, of 8-bit PCM words, plus one synchronization (sync) bit for a total of 193 bits per frame (see Figure 4). This sync bit partitions the boundary between timeslots 24 and 1, and allows the time slot counter at the receive end to maintain sync with the transmit end. All signalling information is contained in bit 8 (LSB) of the PCM word. These multiplexed frames of 24, 193-bit channels constitute the 1.544MHz T1 transmission channel.

A-Law Systems

For A-Law systems, the bus format groups data into 32 timeslots of 8-bit PCM words each, giving 30 voice channels plus one 8-bit sync and alarm channel, and one 8-bit signalling channel. The sync and alarm, and signalling in formation are contained in channels 0 and 16, respectively (see Figure 4). Bits 2, 4, 6, and 8 are inverted for transmission per CCITT recommendation. These multiplexed frames of 32, 256-bit channels constitute the 2.048MHz PCM30-CEPT (Committee of European Postal and Telegraph) transmission channel.

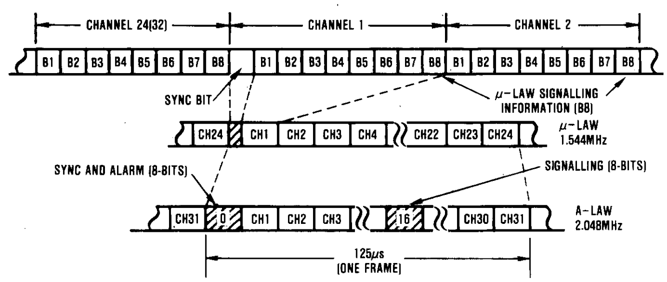

FIGURE 3.

Line Coding

PCM code generated by the CODEC function is in Non-Return to Zero (NRZ) format. It cannot effectively be transmitted directly on a transmission line because the signal contains a DC component and lacks timing information.

An additional coding step is necessary which converts NRZ code to a pseudo ternary code suitable for transmission. Practical coding schemes include Alternate Mark Inversion (AMI), Bipolar with N Zero Substitution (BNZS), and High Density Bipolar 3 (HDB3) coding. These schemes eliminate the DC component of NRZ code, thereby eliminating the troublesome DC wander phenomenon. They also provide a means for detecting line coding errors, and enhance synchronization between transmitter and receiver through reduction of timing jitter. For additional information, refer to Application Note 573, “The HC-5560 Transcoder”, by D. J. Donovan.

Demultiplexing

After transmission, the CODEC must recover the 8-bit PCM words from the TDM signal, sort out, decode, and distribute the PCM information appropriately. The demultiplexing process is fully controlled electronically.

Decoding

The CODEC receive function allocates a signal amplitude to each 8-bit PCM word which corresponds to the midpoint of the particular quantizing interval. The expanding characteristic is the same as that for non-linear companding on the transmit side. If the LSB of a μ-Law PCM word contains signalling information, it is extracted by the CODEC, latched into a flip-flop, and distributed to the CODEC signalling output (Sig_R). This means that there is a lost bit (LSB) in the incoming PCM data stream during a signalling frame. The decoder interprets the missing LSB as a 1/2 (i.e. halfway between a 0 and a 1) to minimize noise and distortion. The PCM words are decoded in the order in which they are received and then converted to PAM pulses. The PAM pulses are summed, then low pass filtered, which smooths the PAM envelope and reproduces the original voice signal.