Analog data → Digital signal.

The process of transforming analog data into digital signal involves two steps. In the first step, the Analog data is converted in digital data (this process is known as digitization). Then in the next step, the digital data is ~~used to~~ encoded as a digital signal by using NRZ-L or other encoding techniques.

The device used for converting analog data into digital form for transmission and subsequently recovering the original analog data from the digital, is known as Codec (Coder-decoder). Two principle techniques used in Codecs are pulse code modulation (PCM) and delta modulation (DM).

pulse code modulation (PCM) — PCM is based on the sampling theorem: If a signal f(t) is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency, then the samples contain all the information of the original signal. The function f(t) may be reconstructed from these samples by the use of a lowpass filter. Thus, according to this sampling theorem, if voice data is limited to frequencies below 4000 Hz, 8000 samples per second would be sufficient to characterize the voice signal completely. these samples are known as pulse amplitude modulation (PAM) samples or PAM samples. Each sample then must be assigned a binary code for the purpose of digitization. If the original signal is assumed to be bandlimited with a bandwidth of B, PAM samples are taken at the rate of 2B or once every $1/2B$ seconds. If each sample represents L bits, then there should be $M=2^L$ different levels ($L=\log_2 M$). Each PAM sample is then approximated by being quantized into one of 2^L different levels. Since the quantized

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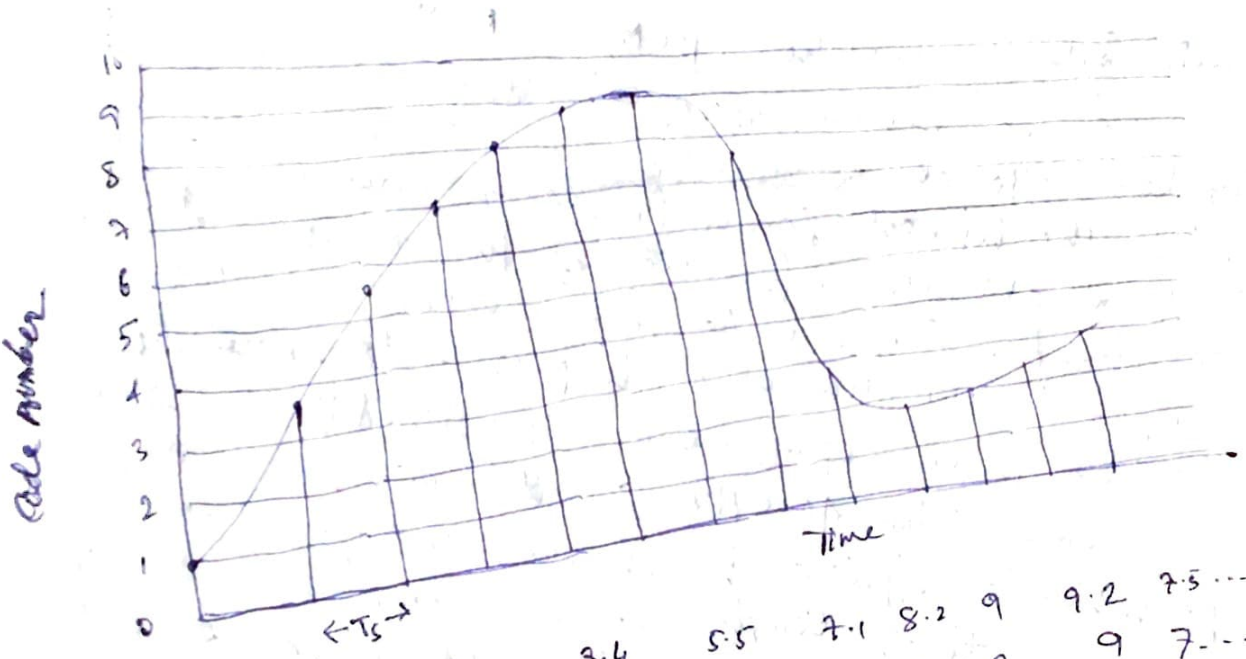
values are only approximations, it is impossible to recover the original signal exactly. This effect is known as quantizing error or quantizing noise. Ofcourse, this approximation can be minimized by increasing the number of different levels, ~~that~~ ~~this~~ that is, by increasing L, but this will also increase the data rate. For example, an 8-bit sample, which allows 256 quantizing levels, implies a data rate of 8000 samples per second = 8000 × 8 = 64 kbps. is needed for a single voice signal. We know that increase in data rate will increase BER. In general, each additional bit used for quantizing increases SNR by about 6 dB which implies a factor of 4 power increase. [$SNR_{dB} = 20 \log 2^n + 1.76 dB = 6.02n + 1.76$ dB]

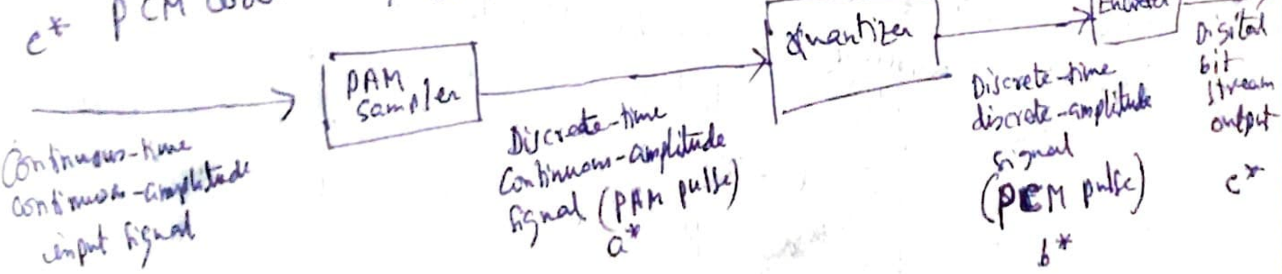
Then increasing the number of bits is not a practical solution.



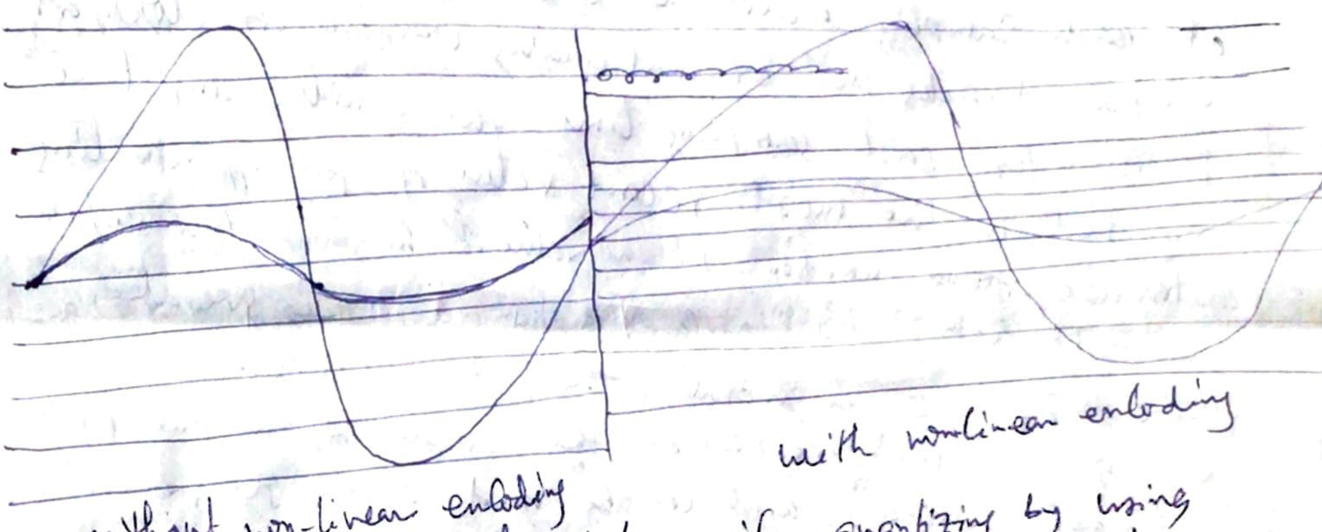
a* PAM value : 1 3.4 5.5 7.1 8.2 9 9.2 7.5 ......

b* Quantized Code number : 1 3 5 7 8 9 9 7 ......

c* PCM Code : 0001 0011 0101 0111 1000 1001 1001 0111 ....

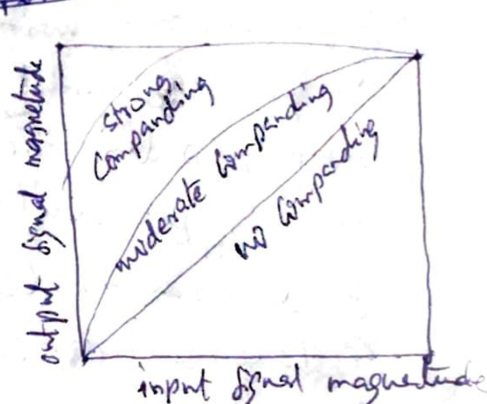


[illegible], PCM scheme is refined using a technique known as nonlinear encoding; in which the quantization levels are not equally spaced. The problem with equal spacing is that the mean absolute error for each sample is same, regardless of signal level. Consequently, lower amplitude values are relatively more distorted than the higher amplitude values. Thus, by allowing greater number of quantizing steps for signals of low amplitude and a smaller number of quantizing steps for signals of large amplitude, we can achieve a marked reduction in overall signal distortion.



The same effect can be achived by uniform quantizing by using companding - expanding techniques on the input signal.

Companding is a process of compressing intensity range of a signal by imparting more gain to weak signal than to strong signal on input. ~~At output, the reverse is performed.~~



Thus, with a fixed number of quantizing levels, more levels are available for lower level signals. At output, the compander expands ~~[illegible]~~ so that compressed values are restored to their original values.

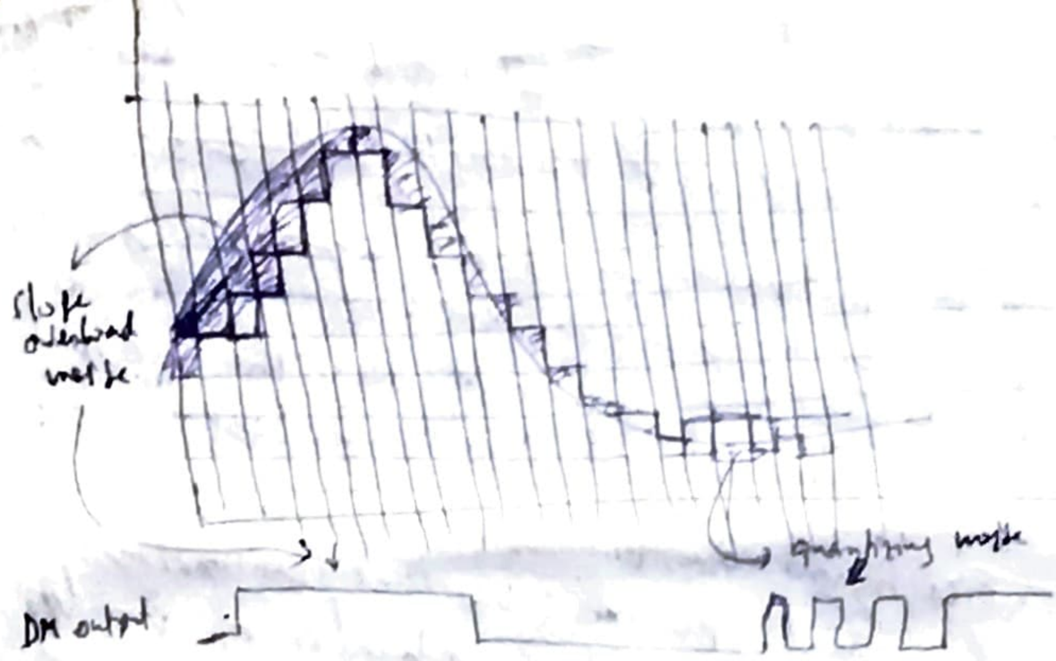
Non-linear encoding can significantly improve the PCM SNR ratio. For example, for voice signal it can improve upto 24 to 30 dB.

③ Delta modulation (DM):- One most popular alternatives to PCM is delta modulation which is used to improve the performance of PCM or to reduce its complexity.

With DM, an analog input is approximated by a staircase function that moves up or down by one quantization level (δ) at each sampling interval (T_s). Most important characteristic of this staircase function is that its behavior is binary (moves up or down a constant amount δ). ~~So it is~~ Hence, ~~so~~ the output of DM process is a bits stream. A 1 is generated if the staircase function moves up during next interval, a 0 is generated otherwise. The transition (up or down) at each sampling interval is chosen such that the staircase function tracks the original analog waveform as closely as possible. At each sampling time, the analog input is compared to the most recent value of the approximating staircase ~~value~~ function. If the value of the sampled waveform exceeds that of the staircase function, a 1 is generated, otherwise a 0 is generated. thus, the staircase is always changed in the direction of the input signal, and produces a binary sequence which can be used at the receiver to reconstruct the staircase function.

Two important parameters of DM are δ and sampling rate. When the analog waveform is changing very slowly, there will be quantizing noise. This noise increases as δ increases. On the other hand, when analog waveform is changing more rapidly than the staircase can follow, there will be slope overload noise: This noise increases as δ decreases. Hence δ must be chosen to produce a balance between these two ~~no~~ noises.

Though, accuracy of the scheme can be improved by increasing sampling rate, it will also increase the data rate of the output signal.



The principle advantage of DM over PCM is the simplicity of its implementation. However, in general, PCM exhibits better SNR characteristics than DM at the same data rate.

Performance

A very good voice reproduction via PCM can be achieved with ~~normally~~, 128 quantization levels or 7-bit coding ($2^7 = 128$). A voice signal occupies a bandwidth of 4 kHz. Thus, according to sampling theorem, samples should be taken at a rate of 8000 samples per second which implies $8000 \times 7 = 56$ kbps data rate for the PCM-encoded digital data. But by Nyquist theorem, this digital signal could require 28 kHz of bandwidth. Even severe differences are seen with higher bandwidth signals. In spite of that, digital techniques continue to grow in popularity for transmitting analog data

$C = 2B\log_2 M \Rightarrow 56 = 2 \cdot 28 \cdot 1$ [M=2]

mainly because:—

1. Repeaters are used instead of amplifiers, there is no cumulative noise.
2. Normally, TDM (time division multiplexing) is used for digital signals instead of FDM (frequency division multiplexing) used in analog signals. With TDM, there is no intermodulation noise, whereas this is a problem in FDM.
3. allows more efficient digital switching techniques.
4. both analog and digital data can be handle similarly.