Quantization Level

The number of quantization levels for an N bit converter is 2N.

From: The Circuit Designer's Companion (Third Edition), 2012

Related terms:

Pulse Code Modulation, Amplitudes, Analog Signal, Analog-to-Digital Converter, Electric Potential, Quantisation, Quantization Error, Quantizer

View all Topics

The Big Picture

John Semmlow, in Circuits, Signals and Systems for Bioengineers (Third Edition), 2018

1.2.3.2.2 Quantization

Slicing the signal amplitude in discrete levels, quantization, is shown in Figure 1.6. The equivalent number can only approximate the level of the analog signal, and the degree of approximation depends on the range of numbers used and the amplitude of the analog signal. For example, if the signal is converted into an 8-bit binary number, the range of numbers is 2^8 or 256 discrete values. If the analog signal amplitude ranges between 0.0 and 5.0 V, then the quantization interval is 5/256 or 0.0195 V. If the analog signal is continuous in value, as is usually the case, it can only be approximated by a series of binary numbers representing the approximate analog signal level at discrete points in time as seen in Figure 1.6. The errors associated with quantization are described in Chapter 3.

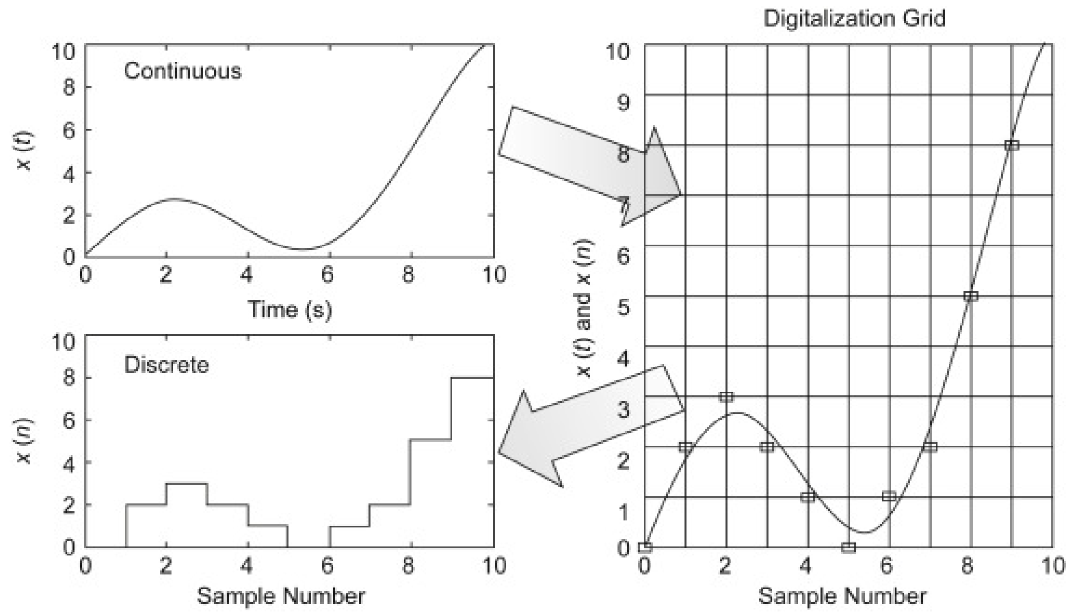

Figure 1.6. Digitizing a continuous signal, upper left, requires slicing the signal in time and amplitude, right. The result, lower left, is a series of numbers that approximate the original signal as a series of discrete levels at discrete time values. This digitizing operation is also known as analog-to-digital conversion.

Example 1.1

A 12-bit analog-to-digital converter (ADC) advertises an accuracy of ± the least significant bit (LSB). If the input range of the ADC is 0 to 10 V, what is the accuracy of the ADC in analog volts?

Solution: If the input range is 10 V, then the analog voltage represented by the LSB is:

Hence the accuracy would be ±.0024 V.

It is relatively easy, and common, to convert between the analog and digital domains using electronic circuits specially designed for this purpose. Many medical devices acquire the physiological information as an analog signal, then convert it to digital format using an “analog-to-digital converter” (“ADC”) for subsequent computer processing. For example, the electrical activity produced by the heart can be detected using properly placed electrodes, and the resulting signal, the electrocardiogram (ECG), is an analog encoded signal. This signal might undergo some “preprocessing” or “conditioning” using analog electronics, but would eventually be converted to a digital signal using an ADC for more complex, computer-based processing and storage. In fact, conversion to digital format is usually done even when the data are only stored for later use.

Transformation from the digital to the analog domain is possible using a “digital-to-analog” converter (“DAC”). Most PCs include both ADCs and DACs as part of a sound card. This circuitry is specifically designed for the conversion of audio signals, but can be used for other analog signals. Data transformation cards and USB-driven devices designed as general-purpose ADCs and DACs are readily available and offer greater flexibility in sampling rates and conversion gains. These devices generally provide multichannel ADCs (usually 8–16 channels) and several DAC channels.

In this text, the basic concepts that involve signals are often introduced or discussed in terms of analog signals, but most of these concepts apply equally well to the digital domain, assuming that the digital representation of the original analog signal is accurate. The equivalent digital domain equation is presented alongside the analog equation to emphasize the equivalence. Many of the problems and examples use a computer, so they are necessarily implemented in the digital domain even if they are presented as analog domain problems.

Is a signal that has been transformed from the continuous to the discrete domain the same? Clearly not; just compare the two different signals in Figure 1.5. Yet in signal analysis we often operate on discrete signals converted from an analog signal with the expectation (or assumption) that the discrete version is essentially the same as the original continuous signal. If they are not the same, is there at least some meaningful relationship between the two? The definitive answer is, maybe. The conditions necessary for the existence of a meaningful relationship between a continuous signal and its discrete version are described in Chapter 4. For now we will assume that all computer-based signals used in examples and problems are accurate representations of their associated continuous signals. In Chapter 4 we look at the consequences of the analog-to-digital conversion process in more detail and establish rules for when a digitized signal can be taken as a truthful representation of the original analog signal.

> Read full chapter

Biomedical signals and systems

Sri Krishnan, in Biomedical Signal Analysis for Connected Healthcare, 2021

2.2.2 Signal power:

Therefore,

The *SNRq* improves as the number of quantization levels increases, given that there are fixed number of levels and they could be represented by bits, and therefore, the bit rate requirements also increase. It should be noted that each increase in bit size leads to a 6 dB increase in *SNRq*.

Fig. 3.6 shows a simple example of sampling a continuous-time signal with a sampling frequency of 5 Hz and with a quantization of four levels. The bit rate calculation is also shown to give an idea on how to calculate bandwidth and storage requirements of biomedical signals for computing and communication needs in a connected healthcare setup.

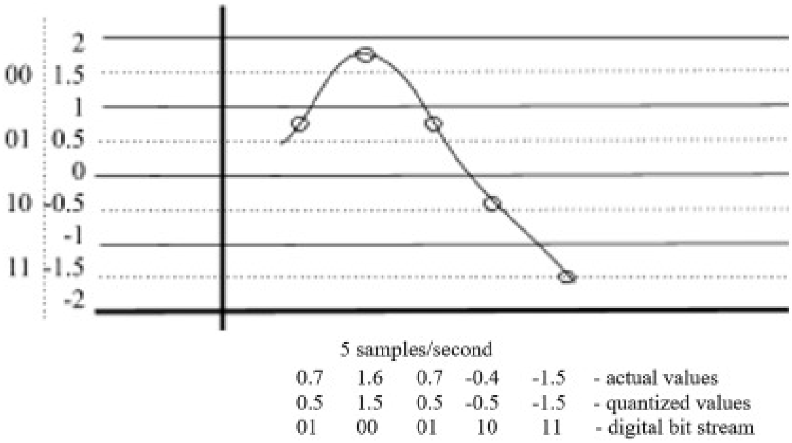

Figure 3.6. A simple example showing analog signal to digital bit stream conversion (sampling and quantization processes).

- In this case, there are 4 levels, and therefore, 2 bits are needed. The bit rate is calculated as

The process of quantization described here is a uniform scalar quantization process, and there are various extensions of quantization process that are possible and they include nonuniform quantization and vector quantization. Biomedical signals are typically quantized in a range of 8 bits/sample to 16 bits/sample, and such a range needs to be optimized if the connected healthcare system poses bandwidth or storage limitations. Digitized biomedical signals provide an appropriate domain for computer-based processing and analysis of signals. The underlying theory in processing and analyzing the signals would be covered in the subsequent sections.

> Read full chapter

Digital Signals

Ian Sinclair, in Electronics Simplified (Third Edition), 2011

Conversions

No advantages are ever obtained without paying some sort of price, and the price to be paid for the advantages of digital recording, processing, and reproduction of signals consists of the problems of converting between analog and digital signal systems, and the increased rate of processing of data. For example, a sound wave is not a digital signal, so that its electrical counterpart must be converted into digital form. This must be done at some stage where the electrical signal is of reasonable amplitude, several volts, so that any noise that is caused will be negligible in comparison to the signal amplitude. That in itself is no great problem, but the nature of the conversion is.

What we have to do is to represent each part of the wave by a number whose value is proportional to the voltage of the waveform at that point. This involves us right away in the two main problems of digital systems: resolution and response time. Since the conversion to and from sound waves is the most difficult challenge for digital systems, we shall concentrate on it here. By comparison, radar, and even television, are systems that were almost digital in form even from the start. For example, the television line waveform consists of the electrical voltage generated from a set of samples of brightness of a line of a scanned image, and it is as easy to make a digital number to represent each voltage level as it is to work with the levels as a waveform.

To see just how much of a problem the conversion of sound waves is, imagine a system that used only the numbers –2 to +2, on a signal of 4 V total peak-to-peak amplitude. If this were used to code a waveform (shown as a triangular wave for simplicity) as in Figure 9.1(a) then since no half-digits can exist, any level between –0.5 V and +0.5 V would be coded as 0, any signal between +0.5 V and +1.5 V as 1 and so on, using ordinary denary numbers rather than binary numbers here to make the principle clearer. In other words, each part of the wave is represented by an integer (whole) number between –2 and +2, and if we plotted these numbers on the same graph scale then the result would look as in Figure 9.1(b). This is a ‘block’ shape of wave, but recognizably a wave which if heavily smoothed would be something like the original one.

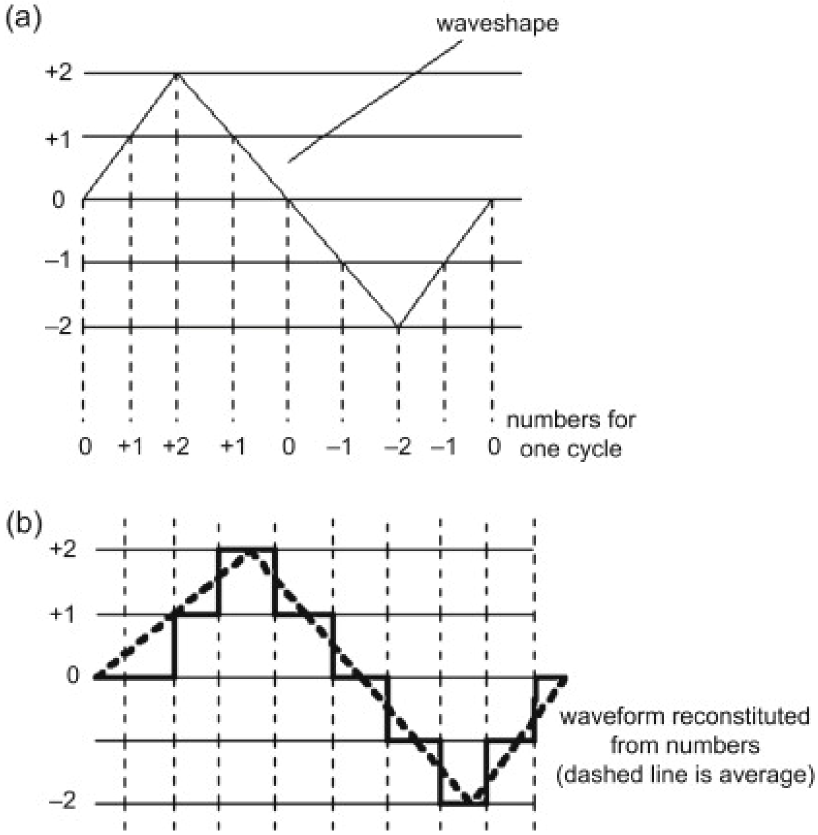

Figure 9.1. Quantizing a waveform. Each new level of voltage is represented by the number for that level, so that the waveform is coded as a stream of numbers. Reversing the process produces a shape that when smoothed (averaged) provides a recognizable copy of the input even for this very crude five-level system

We could say that this is a five-level quantization of the wave, meaning that the infinite number of voltage levels of the original wave has been reduced to just five separate levels. This is a very crude quantization, and the shape of a wave that has been quantized to a larger number of levels is a much better approximation to the original. The larger the number of levels, the closer the wave comes to its original pattern, though we are cheating in a sense by using a sinewave as an illustration, since this is the simplest type of wave to convert in each direction; we need know only one number, the peak amplitude, to specify a sinewave. Nevertheless, it is clear that the greater the number of levels that can be expressed as different numbers then the better is the fidelity of the sample.

Definition

Quantization means the sampling of a waveform so that the amplitude of each sample can be represented by a number. It is the essential first step in converting from analog form to digital form.

In case you feel that all this is a gross distortion of a wave, consider what happens when an audio wave of 10 kHz is transmitted by medium-wave radio, using a carrier wave of 500 kHz. One audio wave will occupy the time of 50 radio waves, which means in effect that the shape of the audio wave is represented by the amplitudes of the peaks of 50 radio waves, a 50-level quantization. You might also like to consider what sort of quantization is involved when an analog tape system uses a bias frequency of only 110 kHz, as many do. Compare this with the 65,536 levels used for a CD.

The idea of carrying an audio wave by making use of samples is not in any way new, and is inherent in amplitude modulation (AM) radio systems which were considered reasonably good for many years. It is equally inherent in frequency modulation (FM), and it is only the use of a fairly large amount of frequency change (the peak deviation) that avoids this type of quantization becoming too crude. Of all the quantized ways of carrying an audio signal, in fact, FM is probably the most satisfactory, and FM methods are often adopted for digital recording, using one frequency to represent a 0 and another to represent a 1. Another option that we shall look at later is changing the phase and amplitude of a wave, with each different phase and amplitude representing a different set of digital bits.

Summary

The conversion of a waveform into a set of digital signals starts with quantization of the wave to produce a set of numbers. The greater the number of quantization levels, the more precise the digital representation, but excessive quantization is wasteful in terms of the time required.

This brings us to the second problem, however. Because the conversion of an audio wave into a set of digits involves sampling the voltage of the wave at a large number of intervals, the digital signal consists of a large set of numbers. Suppose that the highest frequency of audio signal is sampled four times per cycle. This would mean that the highest audio frequency of 20 kHz would require a sampling rate of 80 kHz. This is not exactly an easy frequency to record even if it were in the form of a sinewave, and the whole point of digital waveforms is that they are not sinewaves but steep-sided pulses which are considerably more difficult to record. From this alone, it is not difficult to see that digital recording of sound must involve rather more than analog recording.

The next point is the form of the numbers. We have seen already that numbers are used in binary form in order to allow for the use of only the two values of 0 and 1. The binary code that has been illustrated in this chapter is called 8-4-2-1 binary, because the position of a digit represents the powers of two that follow this type of sequence. There are, however, other ways of representing numbers in terms of 0 and 1, and the main advantage of the 8-4-2-1 system is that both coding and decoding are relatively simple. Whatever method is used, however, we cannot get away from the size that the binary number must have. It is generally agreed that modern digital audio for music should use a 16-bit number to represent each wave amplitude, so that the wave amplitude can be any of up to 65,536 values. For each sample that we take of a wave, then, we have to record 16 digital signals, each 0 or 1, and all 16 **bits** will be needed in order to reconstitute the original wave. We refer to this as a **bit-depth** of 16. Bit-depths higher than this are used for professional equipment; a bit-depth of 24 is typical.

Definition

A bit is short for a binary digit, a 0 or 1 signal. By convention, bits are usually gathered into a set of eight, called a **byte**. A pair of bytes, 16 bits, is called a **word**. Unfortunately, the same term is also used for higher number groupings such as 32, 64, 128, etc.

> Read full chapter

Digital video and audio coding

Richard Brice, in Newnes Guide to Digital TV (Second Edition), 2003

Dither

When a quantizer converts a very large signal that crosses many quantization levels, the resulting errors from literally thousands of very slightly wrong values do indeed create a noise signal that is random in nature. Hence the misnomer quantization noise. But when a digital system records a very small signal, which only crosses a few quantization thresholds, the errors cease to be random. Instead, the errors become correlated with the signal and, because they are correlated with (or related to) the signal itself, they are far more noticeable than would be an equivalent random source of noise.

In 1984, Vanderkooy and Lipshitz proposed an ingenious and inspired answer to this problem. They demonstrated that it is possible to avoid quantization errors completely by adding a very small amount of noise to the original analogue signal prior to the analogue-to-digital converter integrated circuit. They showed that a small amount of noise is enough to break up any patterns the brain might otherwise be able to spot by shifting the signal constantly above and below the lowest quantizing thresholds. This explains the block in Figure 3.1 marked ‘ditherer’, which is shown as summing with the input signal prior to the sampler. In the pioneering days of digital audio and video, the design of ADCs and DACs consumed a vast amount of the available engineering effort. Today's engineer is much luckier. Many 'one-chip' solutions exist which undertake everything but a few ancillary filtering duties.

> Read full chapter

Bits 'n' Pieces – Digital Audio

Richard Brice, in Music Engineering (Second Edition), 2001

Dither

When a quantiser converts a very large signal that crosses many quantisation levels, the resulting errors from literally thousands of very slightly wrong values do indeed create a noise signal which is random in nature. Hence the misnomer quantisation noise. But when a digital system records a very small signal, which only crosses a few quantisation thresholds, the errors cease to be random. Instead the errors become correlated with the signal. And because they are correlated with (or related to) the signal itself, they are far more noticeable than would be an equivalent random source of noise.

In 1984 Vanderkooy and Lipshitz proposed an ingenious and inspired answer to this problem. They demonstrated that it is possible to avoid quantisation errors completely, by adding a very small amount of noise to the original analogue signal prior to the analogue to digital converter integrated circuit. They showed that a small amount of noise can cause low-level information to be coded as a kind of pulse-width modulation of the least significant bit, as illustrated in Figure 10.8. This explains the block in Figure 10.3 marked dither noise generator which is shown as summing with the input signal prior to the sampler. In the pioneering days of digital audio and video, the design of ADC's and DAC's consumed a vast amount of the available engineering effort. Today's engineer is much luckier. Many 'one-chip' solutions exist which undertake everything but a few ancillary filtering duties. Circuits for a high-quality, commercial audio ADC and DAC are given in Figure 10.9a–c. These utilise one-chip solutions from Crystal Semiconductor. A commercial DAC unit is illustrated in Figure 10.10.

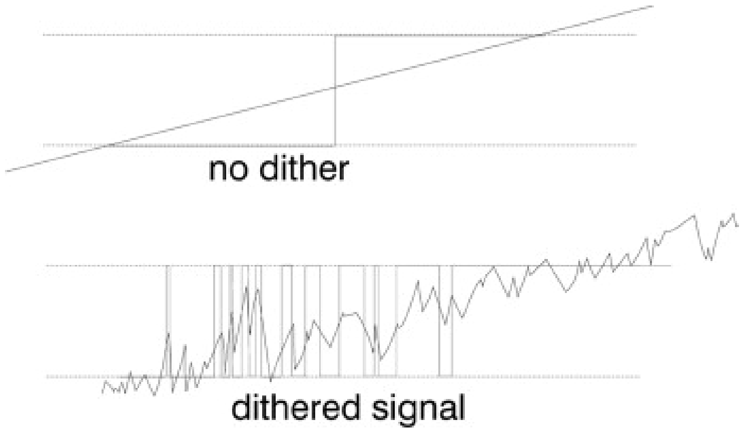

Figure 10.8. How ‘dither’ noise codes low-level information

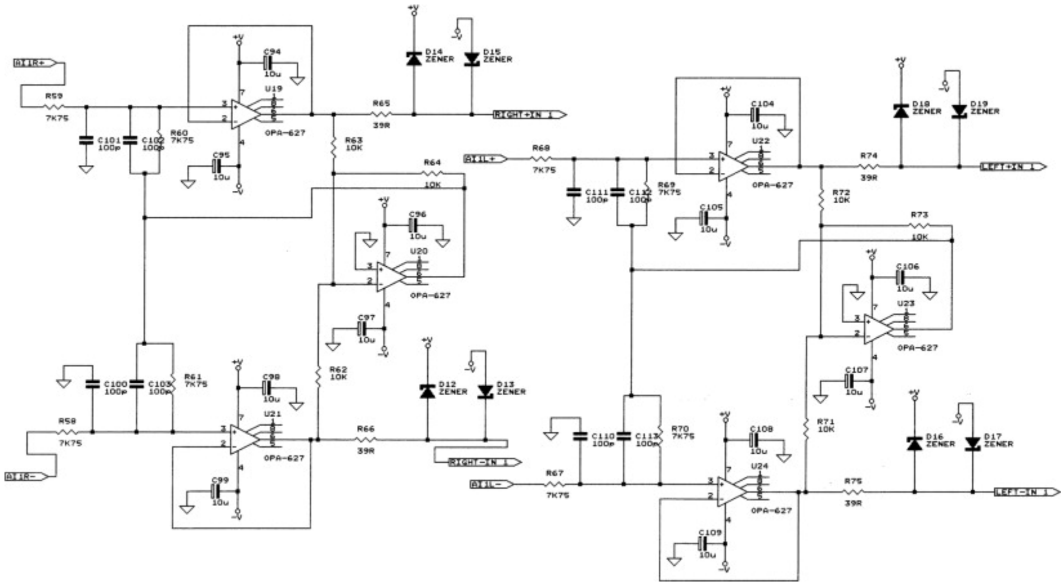
Figure 10.9a. Circuits for ADC and DAC

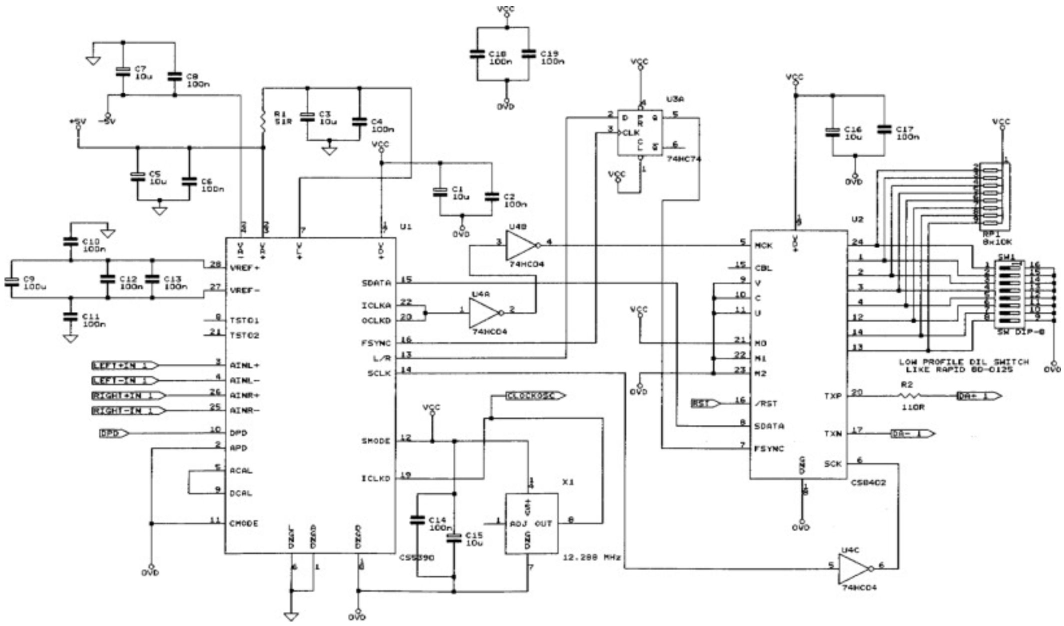
U1
CS5390
U2
CS8402
U3A
74HC74
U4A
74HC04
U4B
74HC04
U4C
74HC04
X1
12.288 MHz
RP1
8x10K
SW1
SW DIP-8
LOW PROFILE DIL SWITCH
LIKE RAPID 80-0125
VCC
DVD
+5V
-5V
VREF+
VREF-
TSTO1
TSTO2
AINL+
AINL-
AINR+
AINR-
DPD
APD
ACAL
DCAL
CMODE
SDATA
ICLKA
OCLKD
FSYNC
L/R
SCLK
SMODE
ICLKD
MCK
CBL
/RST
TXP
TXN
SCK
R2
110R
CLOCKOSC

Figure 10.9b.

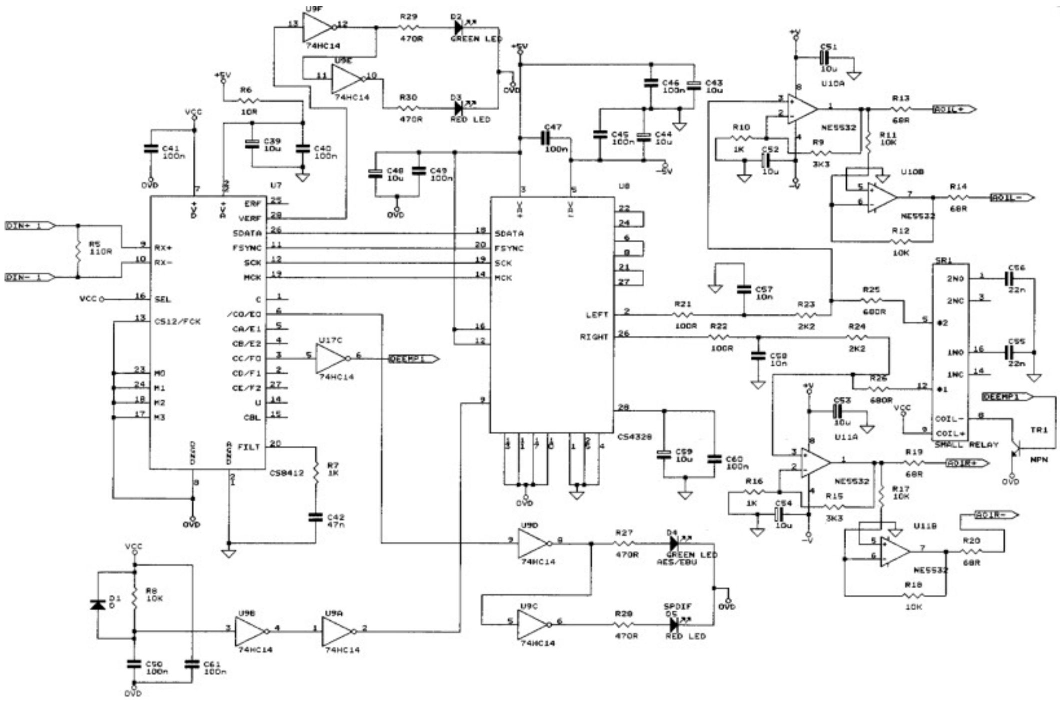

Figure 10.9c.

Figure 10.10. Commercial high-quality digital to analogue converter

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Distributed Video Coding: Basics, Codecs, and Performance

Fernando Pereira, ... João Ascenso, in Distributed Source Coding, 2009

8.6.1.2 Quantization

Different (decoded) quality can be achieved by changing the number of quantization levels, M_k, used for each DCT band b_k. In this section, eight rate-distortion (RD) points are considered, corresponding to the various 4 × 4 quantization matrices depicted in Figure 8.7. Within a 4 × 4 quantization matrix, the value at position k in Figure 8.7 (position numbering within the matrix is made in a zigzag scanning order) indicates the number of quantization levels associated with the DCT coefficients band b_k. The value 0 means that no Wyner–Ziv bits are transmitted for the corresponding band. In the following, the various matrices will be referred to as Q_i with $i = 1, \ldots, 8$; the higher is Q_i, the higher are the bit rate and the quality.

16	8	0	0
8	0	0	0
0	0	0	0
0	0	0	0

(a)

32	8	0	0
8	0	0	0
0	0	0	0
0	0	0	0

(b)

32	8	4	0
8	4	0	0
4	0	0	0
0	0	0	0

(c)

32	16	8	4
16	8	4	0
8	4	0	0
4	0	0	0

(d)

32	16	8	4
16	8	4	4
8	4	4	0
4	4	0	0

(e)

64	16	8	8
16	8	8	4
8	8	4	4
8	4	4	0

(f)

64	32	16	8
32	16	8	4
16	8	4	4
8	4	4	0

(g)

128	64	32	16
64	32	16	8
32	16	8	4
16	8	4	0

(h)

FIGURE 8.7. Eight quantization matrices associated with different RD points.

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Data Models

Robert A. SchowengerdtProfessor, in Remote Sensing (Third edition), 2007

National Imagery Interpretability Scale (NIIRS)

Spatial characteristics, such as *GIFOV* or *GSI*, and radiometric characteristics, such as quantization level or detector noise, are required for numerical design, comparison, and evaluation of imaging systems. However, they do not relate the system parameters to the tasks expected to be achieved from the imagery. A summary metric that attempts to make just such a connection is the *National Imagery Interpretability Scale (NIIRS)*. NIIRS was developed for military applications where imagery is interpreted visually by experienced and certified analysts. It is primarily dependent on spatial resolution capability, i.e., *GSI*, but also includes image *SNR*- and *PSF*-related influences. A 10-level NIIRS scale has been developed for military application; an abbreviated description of the panchromatic imagery NIIRS is given in Table 4-1; a more detailed description can be found in Leachtenauer *et al.* (1997) and IRARS (1996). A multispectral imagery NIIRS has also been developed (IRARS, 1995). These scales are applied to unrated imagery by presenting those images to trained ("NIIRS certified") interpreters and asking them to rate each according to the level of detail that can be discerned in the image. An example of applying the process to IKONOS 1m panchromatic imagery, for which an average NIIRS rating of 4.5 was obtained, is described in Ryan *et al.* (2003).

TABLE 4-1. Example of the National Image Interpretability Scale (NIIRS).

rating level	**example criterion**
0	cannot be interpreted due to clouds or poor quality
1	distinguish airport taxiways and runways
2	detect large buildings
3	identify large ship type
4	identify individual tracks in railroad yard
5	identify individual railcars by type
6	identify automobiles as sedans or station wagons
7	indentify individual railroad ties
8	indentify vehicle windshield wipers
9	detect individual railroad tie spikes

At first glance, the NIIRS does not seem particularly relevant to civilian remote sensing as emphasized in this book. However, with the trend to higher resolution multispectral systems, such as IKONOS, QuickBird, and OrbView, there is increasing crossover between the civilian and military communities in the use of the same sensors for different applications. Also, the overall goal expressed by the NIIRS of relating system characteristics to task performance is a useful and potentially valuable approach for quantitative remote sensing system analysis. For example, it is possible to relate sensor parameters mathematically to NIIRS and thereby predict whether a particular imaging system will perform adequately for specified tasks using the *General Image-Quality Equation (GIQE)* (Leachtenauer *et al.*, 1997). To apply this concept to civilian earth remote sensing systems, the set of tasks should be defined appropriately, e.g. mapping of the Anderson land cover and land use categories (Chapter 9). Another major change is that the tasks are not defined by visual analysis but by computer classification or other information extraction processes, whose performance needs to be described in a numerical way.

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Waveform Quantization and Compression

Lizhe Tan, Jean Jiang, in Digital Signal Processing (Third Edition), 2019

10.5 Summary

1. The linear midtread quantizer used in the PCM coding has an odd number of quantization levels, that is, $2n - 1$. It accommodates the same decoded magnitude range for quantizing the positive and negative voltages.
2. Analog or digital μ-law compression improves coding efficiency. 8-bit μ-law compression of speech is equivalent to 12-bit linear PCM coding, with no difference in the sound quality. These methods are widely used in the telecommunications industry and multimedia system applications.
3. DPCM encodes the difference between the input sample and predicted sample using a predictor to achieve the coding efficiency.
4. DM coding is essentially a 1-bit DPCM.
5. ADPCM is similar to DPCM except that the predictor transfer function has six zeros and two poles and is an adaptive filter. ADPCM is superior to 8-bit μ-law compression, since it provides the same sound quality with only 4 bits per code.
6. Data compression performance is measured in terms of the data CR and the bit rate.
7. The DCT decomposes a block of data to the DC coefficient (average) and AC coefficients (fluctuation) so that different numbers of bits are assigned to encode DC coefficients and AC coefficients to achieve data compression.
8. W-MDCT alleviates the block effects introduced by the DCT.
9. The MPEG-1 audio formats such as MP3 (MPEG-1, layer 3) include W-MDCT, filter banks, a psychoacoustic model, bit allocation, a nonlinear quantizer, and Huffman lossless coding.

> Read full chapter

Physics and Fundamental Theory

N. Miura, in Comprehensive Semiconductor Science and Technology, 2011

Magnetic fields quantize the energy levels of conduction bands and valence bands in semiconductors. The effects of the level quantization are visible in transport and optical phenomena as oscillatory structures or as prominent peaks in the spectra. By analyzing the spectra as a function of magnetic field or energy, we can obtain information of the Landau levels and the energy band structure. Especially, in high magnetic fields, the quantization effect becomes very distinct, as the Landau level spacing becomes large in comparison to the level broadening due to the carrier scattering. In two-dimensional electron systems, the effect of magnetic fields applied perpendicular to the two-dimensional plane is such that the quantum phenomena and the electron–electron interaction effects are conspicuously observed. In this chapter, we present a brief review of such magneto-spectroscopy. The topics include magneto-transport phenomena, such as the Shubnikov–de Haas effect, the magneto-tunneling effect, and the magnetophonon effect, magneto-optical effects such as interband magneto-absorption, the magneto-exciton spectra, Faraday rotation spectra, and cyclotron resonance.

> Read full chapter

Data Converters

Nihal Kularatna, in Modern Component Families and Circuit Block Design, 2000

3.2.2 ADC Resolution and Dynamic Range Requirements

Having discussed the sampling rate and filtering, we next discuss the effects of dividing the signal amplitude into a finite number of discrete quantization levels. Table 3-1 shows relative bit sizes for various resolution ADCs, for a full-scale input range chosen as approximately 2 V, which is popular for higherspeed ADCs. The bit size in determined by dividing the full-scale range (2.048 V) by 2_N.

Table 3-1. Bit Sizes, Quantization Noise, and Signal-to-Noise Ratio (SNR) for 2.048 V Full-Scale Converters

Resolution (N Bits)	1 LSB = q	%FS	Rms Quantization Noise,	Theoretical Full-Scale SNR(dB)
6	32 mV	1.56	9.2 mV	37.9
8	8 mV	0.39	2.3 mV	50.0
10	2 mV	0.098	580 µV	62.0

12	500 μV	0.024	144 μV	74.0
14	125 μV	0.0061	36 μV	86.0
16	31 μV	0.0015	13 μV	98.1

The selection process for determining the ADC resolution should begin by determining the ratio between the largest signal (full-scale) and smallest signals you wish the ADC to detect. Convert this ratio to dB and divide by 6. This is your minimum ADC resolution requirement for DC signals. You actually will need more resolution to account for extra signal headroom, since ADCs act as hard limiters at both ends of their range. Remember that this computation is for DC or low-frequency signals and that the ADC performance will degrade as the input signal slew rate increases. The final ADC resolution actually will be dictated by dynamic performance at high frequencies. This may lead to the selection of an ADC with more resolution at DC than is required.

Table 3-1 also indicates the theoretical rms quantization noise produced by a perfect *N*-bit ADC. In this calculation, the assumption is that quantization error is uncorrelated with the ADC input. With this assumption, the quantization noise appears as random noise spread uniformly over the Nyquist bandwidth, DC to $f_s/2$, and it has an rms value equal to Other cases may be different, and some practical explanation is given in Analog Devices (1995).

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