Cisco Systems

Telecommunications Technologies Reference

A comprehensive guide to North American and international telecommunications standards

Bradley Dunsmore
Toby Skandier

ciscopress.com

Telecommunications Technologies Reference

Brad Dunsmore

Toby Skandier

Christian Martin, Joel T. McKelvey, and Tim Woods

Published by:
Cisco Press
201 West 103rd Street
Indianapolis, IN 46290 USA

Printed in the United States of America 1 2 3 4 5 6 7 8 9 0

First Printing September 2002

Library of Congress Cataloging-in-Publication Number: 2001090440

ISBN: 1-58705-036-6

Warning and Disclaimer

This book is designed to provide information about telecommunications technologies. Every effort has been made to make this book as complete and as accurate as possible, but no warranty or fitness is implied.

The information is provided on an “as is” basis. The author, Cisco Press, and Cisco Systems, Inc. shall have neither liability nor responsibility to any person or entity with respect to any loss or damages arising from the information contained in this book or from the use of the discs or programs that may accompany it.

The opinions expressed in this book belong to the author and are not necessarily those of Cisco Systems, Inc.

Trademark Acknowledgments

All terms mentioned in this book that are known to be trademarks or service marks have been appropriately capitalized. Cisco Press or Cisco Systems, Inc., cannot attest to the accuracy of this information. Use of a term in this book should not be regarded as affecting the validity of any trademark or service mark.

Feedback Information

At Cisco Press, our goal is to create in-depth technical books of the highest quality and value. Each book is crafted with care and precision, undergoing rigorous development that involves the unique expertise of members from the professional technical community.

Readers’ feedback is a natural continuation of this process. If you have any comments regarding how we could improve the quality of this book, or otherwise alter it to better suit your needs, you can contact us through e-mail at feedback@ciscopress.com. Please make sure to include the book title and ISBN in your message.

We greatly appreciate your assistance.

Publisher	John Wait
Editor-in-Chief	John Kane
Cisco Systems Management	Michael Hakkert
	Tom Geitner
Production Manager	Patrick Kanouse
Acquisitions Editor	Michelle Grandin
Development Editor	Andrew Cupp
Project Editor	Eric T. Schroeder
Copy Editor	Cris Mattison
Technical Editors	Ron Hranac
	Grady Neely
	Laurent Nicolas
	Nancy Roth
	Martin Walshaw
Team Coordinator	Tammi Ross
Book Designer	Gina Rexrode
Cover Designer	Louisa Klucznik
Composition	Mark Shirar
Indexer	Tim Wright

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
Cisco Systems Europe
11 Rue Camille Desmoulins
92782 Issy-les-Moulineaux
Cedex 9
France
http://www-
europe.cisco.com
Tel: 33 1 58 04 60 00
Fax: 33 1 58 04 61 00

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters
Cisco Systems Australia,
Pty., Ltd
Level 17, 99 Walker Street
North Sydney
NSW 2059 Australia
http://www.cisco.com
Tel: +61 2 8448 7100
Fax: +61 2 9957 4350

Cisco Systems has more than 200 offices in the following countries. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at www.cisco.com/go/offices

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong • Hungary • India • Indonesia • Ireland Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain Sweden • Switzerland • Taiwan • Thailand • Turkey • Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

Introduction

Over the past several years, markets have opened up around the world that require a broader understanding of service provider networks and telecommunications technologies. In that time, it has become difficult to find a single comprehensive reference guide for both North American and international technology.

This book is designed to serve as a reference guide for a broad range of technology that is found in modern telecommunications networks. The coverage in this book is not limited to the technologies in North America, as it details many of the international communication methods as well. The intention is to be informative of service provider technology regardless of your location.

There are many books on the market that cover specific aspects of telecommunication technologies. Many of them are technology specific or cover only North American methods. This book was created to act as a comprehensive reference guide for individuals whose knowledge must span both international boundaries and various technologies. You will learn about telecommunication methodology as it is deployed around the world, as well as how it operates.

Who Should Read This Book

This book is written for individuals who wish to learn about the technologies used throughout the world for data and voice communication. It is written in such a way that someone with little background in telecommunications can read and understand the material, but it is also technical enough for a field engineer to use as a reference guide.

It is recommended that the reader have the following prerequisite knowledge:

- Fundamental understanding of WAN communication
- CCNA or equivalent experience

Expert Whiteboards

Expert Whiteboards are sections that have been included in most chapters. The purpose of these sections is to discuss an advanced topic of the technology or to detail a specific deployment solution. These sections are written by a variety of engineers to add real-world applications to technologies discussed in each chapter.

Chapter Summaries and Review Questions

The "Summary" and "Review Questions" sections at the end of each chapter are designed to reinforce some of the most important topics that are discussed in that chapter. Use them to assess your understanding of the chapter's subject and then review as necessary. The review questions are in multiple choice, lab-based, and fill in the blank format. The answer for each question is discussed in detail and explained for complete understanding. You can find the answers to the review questions in Appendix A.

CHAPTER 3

Analog-to-Digital Conversion

Understanding Digital Communication

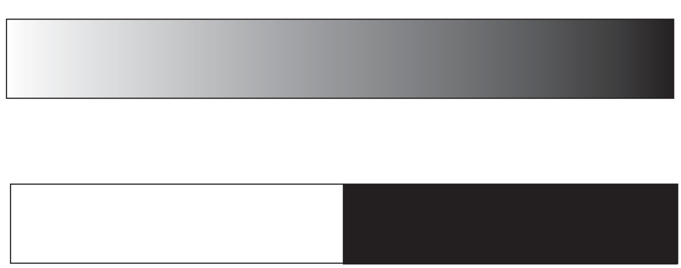
Digital communication is the transmission of information by discrete pulses of electricity or light impulses. The simplicity in digital communication belies its vast power. Often, you can get away with paying attention only to whether a pulse exists in a particular moment of time. Contrast this with the need to monitor every infinite change in signal strength to correctly duplicate a waveform, as in the case of analog communication.

In digital communication, you need concern yourself only with on or off, yes or no, true or false. Only two states exist, which greatly simplifies the information that needs to be transmitted across a communications medium. This alone greatly improves the quality of what you transmit, by eliminating endless "shades of gray" and concentrating only on "black and white."

Motivation for Digital Communication

Just imagine being tasked with the chore of recognizing every shade of gray possible. This presents, literally, an infinite number of possibilities. Contrast this with having to recognize only black and white (see Figure 3-1). All of a sudden, the job gets a lot easier. Furthermore, the chance of committing an error in recognition diminishes, as does the possibility of, for example, a little dust introducing some form of recognition difficulty. Dingy white is still white, and faded black is still black, as long as black and white are the only two choices. As you will see shortly, this reduction in the possibility of error and this loose tolerance for recognition are also benefits of digital communication.

Figure 3-1 *Shades of Gray Analogy*



With this analogy in mind, consider that the main impetus for the development of digital communication was the need for a better way to store musical information, with fewer errors and more quality. In the 1920s, for example, music had to be stored as analog waveforms etched in wax, which could then be read only a limited number of times.

Evolution of Digital Communication

The idea was that analog information could be represented as distinct pulses of information, like Morse code, where only a few simple combinations are combined to represent a larger, more complex set of symbols. After conversion, analog information is stored and retrieved with more accuracy and quality than by directly recording analog information. The idea arose early on to use a bi-state mechanism as the combining element for digital communication. This was the dawn of what is now known as binary digital communication, where 1s and 0s are used exclusively in the transmission and storage of information.

This might be a good time to clarify a fact that is sometimes overlooked when talking about binary digital information. It is important to realize that the 1s and 0s referred to in digital communication are simply human interpretations of what is being stored or communicated. For example, a 1 might indicate that a memory location is holding a charge, or that a communications line is electrified at that moment in time, and a 0 might indicate that no such charge or electrification exists. The use of 1s and 0s is just a way to work with large quantities of binary information and to mathematically predict or analyze various trends and situations.

Improvements Over Analog Communication

Recall the earlier analogy of being tasked with recognizing shades of gray, as opposed to just black and white. The point was made that dust could settle on the object, making it more difficult to correctly identify exactly which shade of gray you were looking at, especially if the shades were infinitesimally close to begin with. Quality assurance in this task would be tricky, at best.

However, if you had to identify the objects only as black or white, it would take quite a bit of dust to obscure the underlying blackness or whiteness. This is also the case with digital signals. So much tolerance is built into the receiving mechanism, that common interference does not alter the perceived value of the digital signal. A slightly stronger or weaker pulse is still read as a pulse. A slight to medium disturbance in the signal, where no pulse should be, is still perceived as the lack of a pulse.

In contrast, analog transmission systems accumulate interference and impairments. After being introduced, these anomalies cannot be easily eliminated. Any change to the original signal permanently alters the quality of the signal, sometimes distorting it to an unrecognizable extent. The effect is heightened when you amplify the analog signal to compensate for signal loss. Any interference that the signal has already picked up is amplified, along with the original signal.

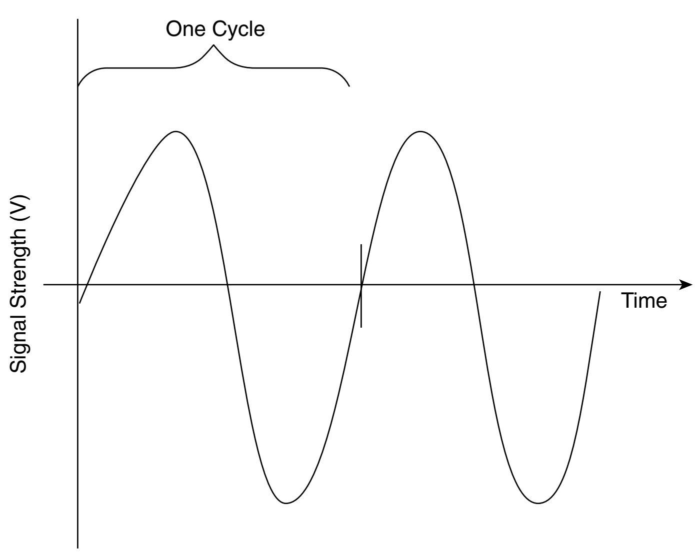
If you go back to the black and white analogy, you can see that if you were asked to perform the recognition process in a slightly darkened room, you would have more trouble identifying the shades of gray, but the identification of black or white would still be relatively simple. This is an example of signal loss. Because colors and grayscale are transmitted by relative reflection of light, less light means less signal arrives at your eye. In fact, a complete signal loss (complete darkness) makes even the recognition of black and white lean entirely to the black (which translates as no optical data or "0"). In this case, it doesn't matter if you're being asked to recognize shades of gray or just black and white. You can't see or recognize anything. The point is, perceived as all 0s, a complete loss of signal is as equally catastrophic in digital transmission, as it is in analog.

In the case of analog and digital communication, you are hindered more by the loss and amplification of the analog signal than by the loss and regeneration of the digital signal. It is a simple task to restore a digital pulse to its original strength and shape, as compared to restoring a potentially distorted analog signal to its original form. A distorted pulse is still a pulse. This is because the digital signal is distorted by analog interference, which is easily ignored. This same analog interference hides within an analog signal, which makes it all but impossible to reverse.

The Signal Conversion Process

By way of review, an analog signal is one in which each point on the waveform is meaningful (a continuous waveform). Analog signals are graphically depicted as sinusoidal waves (see Figure 3-2). This is also the motion that a guitar string exhibits when plucked. The string moves through and past the origin (the x-axis showing time, here), which is nearly an equivalent distance toward the guitar as away from the guitar.

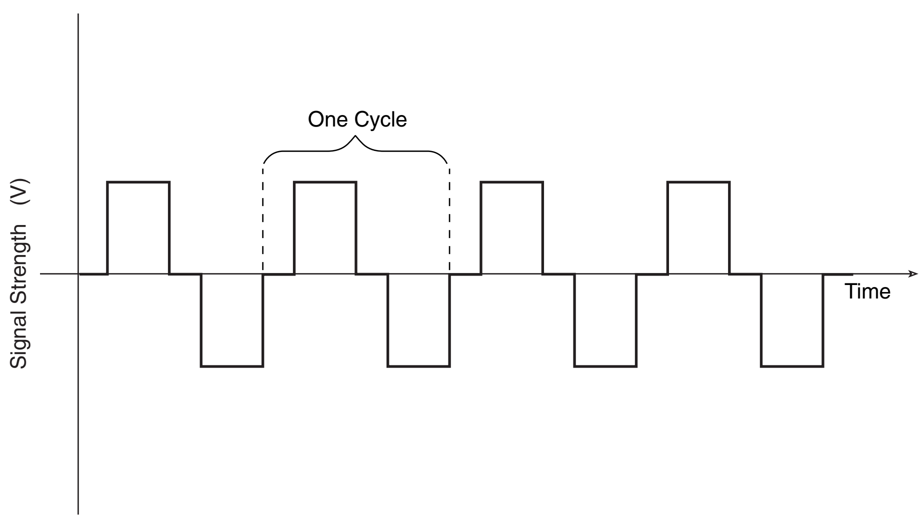
Figure 3-2 *Sinusoidal Analog Waveform*



Although the positive amplitude is often illustrated to be the same as the negative amplitude within the same cycle, in reality, because the signal is usually trailing off or attenuating (losing signal strength) as time goes on, these depictions can be slightly inaccurate. The amplitude of a crest in the waveform often is smaller than that of the previous crest. The concept of a cycle, in graphical terms, which might be easier to understand than the technical definition, is the portion of a waveform that begins and ends at the horizontal axis that is heading in the same direction at the beginning and end. This means that one upsweeping and one downsweeping crest makes up a cycle. Figure 3-2 shows a waveform with two cycles.

In contrast, a digital signal contains discrete pulses, in which only the existence or absence of a pulse, and sometimes the rising or falling disposition of one or more edges of the pulse (as in Manchester encoding), is meaningful. Only the amplitude of the pulse, not the voltage levels attained throughout the creation of the pulse, is important. Digital signals are graphically depicted as square waves (see Figure 3-3). Digital waveforms are also discussed as having cycles, although the physics of analog waves do not govern digital waves.

Figure 3-3 *Square Digital Waveform*



The physical difference between digital waves and analog waves is that unless you employ some form of bipolar line coding scheme (positive and negative representation of the 0s and 1s in a form designed for the transmission medium), such as alternate mark inversion (AMI) or bipolar 8-zero substitution (B8ZS), and transmit all 1s, you might not answer every positive crest with a negative, as in the case of an analog waveform. Line coding schemes such as AMI and B8ZS are covered in more detail later in this book during the discussion of T1. After being coded into digital, the analog source can be represented by a series of bits that have few 1s over various stretches of the bit stream. These 0s are often depicted as no pulse, hence no crest in the wave. Nevertheless, even two consecutive 0s, because of the time slots assigned to the lack of electrical activity, can be considered to make up one cycle in the digital waveform.

Figure 3-3 shows a digital pulse stream that represents some form of bipolar return to zero (BRZ) technology, which is transmitting all 1s in this case. The half-pulse width transmission of zero for each pulse helps keep true to the zero-voltage reference, especially during the transmission of all 1s. This can occur for an extended period during testing or in certain equipment failure conditions to keep the line up. In this case, an extended period without a return to zero might allow the signal to wander, which causes the positive or negative pulses to read as 0s, depending on which way the signal wandered. Digital line coding schemes are covered in detail in Chapter 5, "T1 Technology."

Converting an analog signal into a digital form that adequately represents the original waveform, after its conversion back to analog, is accomplished by a series of four time-tested processes:

- Filtering
- Sampling
- Quantizing
- Encoding

When grouped together, these processes are known collectively as *pulse code modulation (*PCM), which is documented in the International Telecommunication Union Telecommunication Standardization Sector (ITU-T, formerly the CCITT) standard G.711. The sections that follow discuss all four processes in greater detail.

Filtering

Filtering can be thought of as the process of isolating only the contiguous frequencies that you are interested in digitizing. One of the simplest examples is that of an analog voice source, which is intended for digitization and subsequent transmission over digital circuits.

In the case of an analog voice source, the contiguous range of frequencies of interest that communicate across a circuit is the 3100 Hz (often expressed 3.1 kHz) range, from 300 Hz to 3400 Hz, inclusive. This is the most common range of spoken voice.

If frequency, which is expressed in hertz (Hz) or cycles per second, is foreign to you, think of it as pitch or tone. A higher frequency produces a higher pitch, which some might consider more treble, whereas a lower frequency produces a lower, more bass pitch.

Filtering, in this example, is the exclusion of any frequencies below 300 Hz and above 3400 Hz. For the purpose of computing sampling frequency (which is discussed next), the maximum frequency is 4 kHz. This provides a more than adequate sampling rate for what is known as voice frequency (VF).

Sampling

While working for AT&T in 1928, a scientist by the name of Harry Nyquist published the paper "Certain Topics in Telegraph Transmission Theory," in which he outlined what is now referred to as the Nyquist Theorem. This paper was many years ahead of its time. Decades passed before the equipment was available to enable this theorem for digitizing voice for storage or transmission.

Before getting into the Nyquist Theorem, you need to understand the term sampling. *Sampling*, in this case, is the measuring of the amplitude of the analog waveform at regular (equal) intervals. These equal intervals are computed by using the Nyquist Theorem.

The Nyquist Theorem states that to adequately represent an analog wave in digital form, you must sample the analog waveform at a rate at least twice that of the highest frequency to be transmitted. As noted earlier, the maximum voice frequency, for the purpose of computing the sampling frequency of the analog waveform, is 4 kHz.

Using the Nyquist Theorem, you can determine that you need to sample the analog signal that is being transmitted at a rate of 8000 times per second. The Nyquist Theorem dictates that you multiply the maximum frequency of 4000 Hz by two, which yields 8000 samples per second. Figure 3-4 illustrates this mathematical relationship, including units of measure. This numeric value is important to consider in the encoding phase of PCM, which is discussed later in this chapter.

Figure 3-4 *Nyquist Theorem Calculations*

4000 ~~Cycles~~ / Second (Maximum Frequency)

x 2 Samples / ~~Cycle~~ (Nyquist Multiplier)

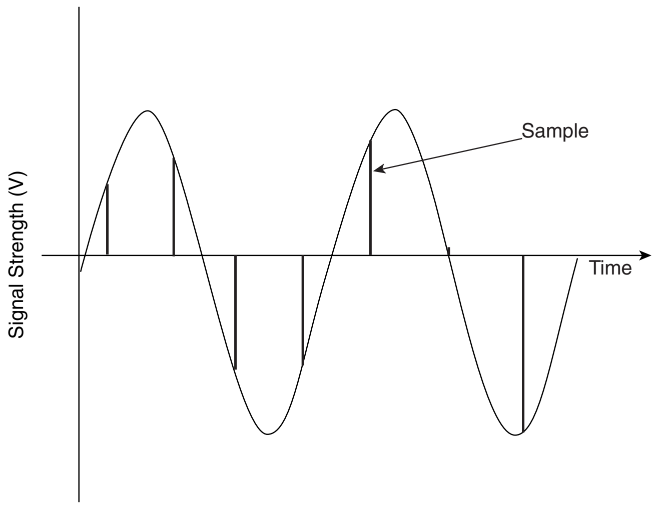
8000 Samples / Second

The equipment responsible for capturing these samples 8000 times per second uses two inputs. One of these inputs is the constant stream of analog information from the source. The other input is a clock signal that occurs 8000 times a second. The result is that only the source signal that exists each time the clock pulse arrives at the gate is captured (sampled).

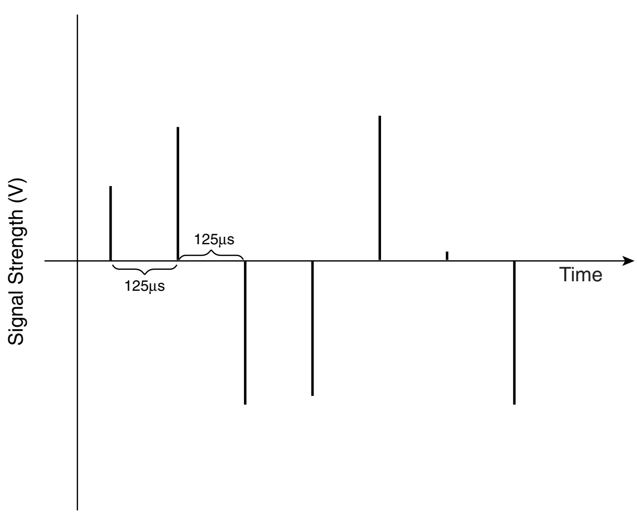
These 8000 samples become the only part of the original analog waveform to remain after the sampling process. If you think of these samples as pulses of varying amplitude, you have what is known as a pulse stream (or pulse train). These pulses are modulated (or varied), based on how the original analog waveform varies. By definition, this is pulse amplitude modulation (PAM). You can consider a PAM signal as the result of the sampling process and the input for the quantizing process.

It's a simple task to compute the equal interval between samples. By splitting a second 8000 ways, you come up with the value 0.000125 seconds (or 125 μs). Therefore, every 125 millionths of a second, regardless of the frequency or amplitude of the analog waveform, a sample is taken. Figure 3-5 illustrates the sampling process.

Figure 3-5 *The Sampling Process*



a. Samples at Regular Intervals



b. The PAM Pulse Train with the Waveform Removed

As shown in Figure 3-5, samples taken at adequate intervals provide a guideline to aid receiving equipment in the reconstruction of the original waveform. The receiving equipment performs a sort of "connect-the-dots" with the pulses in the PAM pulse train.

This, together with specialized circuitry, provides a waveform that is indistinguishable from the original version by the human ear.

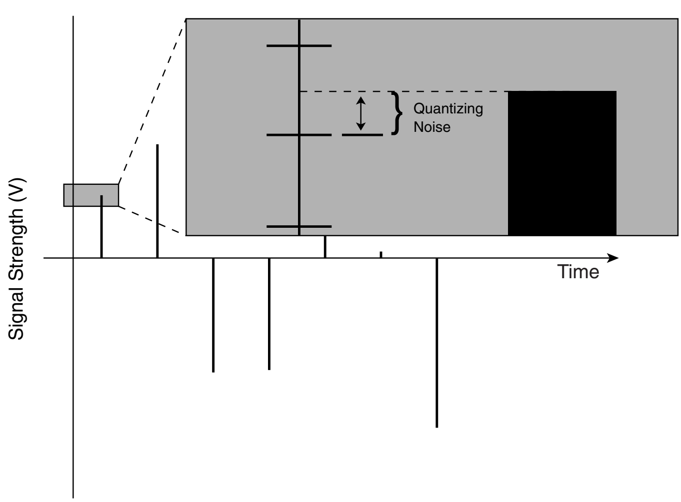
Looking at the pulse stream in Part b of Figure 3-5, you might assume that the digital conversion is well in hand, but consider the fact that the amplitude of each pulse in the train is but one in an infinite range of possible amplitudes. By definition, this is still an analog signal. Nevertheless, only one step stands between the PAM signal and being ready to encode a true digital bit stream—quantizing or quantization.

Quantizing

Now that you have a PAM signal in the form of a pulse train, you need to evaluate the voltage levels of the individual pulses, based upon a standard scale. Only by adjusting each pulse's amplitude to match a value from a finite set can you hope to use a finite series of digital bit patterns to turn each sample into a portion of a bit stream that can reconstruct the pulse train at the receiving end. This is the object of the encoding process that is discussed in the next section.

From the preceding discussion, you might have already surmised that the result of the quantizing phase is still not the final digital signal, though the quantized PAM signal can be thought of as digital. Because 256 discrete voltage levels are more difficult to transmit and receive with as few errors as only two or three levels (in the case of bipolar line coding), further refinement of the quantized digital signal occurs next, in the encoding phase. This portion of the conversion process might well have the least distinguishable output. Nevertheless, it is a crucial step towards digitization. Figure 3-6 clarifies the fairly abstract nature of the quantizing phase.

Figure 3-6 *Imperfections in the Quantizing Process*



Because the original PAM signal is made up of pulses that can have amplitudes within an infinite range, it is necessary to use a finite scale to prepare the PAM signal for the encoding phase, at which point each pulse in the train is converted to a series of 0s and 1s. You cannot expect each original PAM pulse to fall exactly on one of the finite points of the scale, which means that you end up with some altered pulses, with no accompanying information to guide you back to the original pulse. These discrepancies are referred to as *quantizing errors* or *quantizing noise*. These errors do not produce audible differences to the human ear. In fact, minimizing the effect of quantizing errors is the subject of an upcoming discussion on a process called companding.

The PAM pulse is rounded to the nearest point on the scale, regardless of whether that point is higher or lower than the actual sample. This is the first step toward minimizing quantizing errors. The biggest reduction in quantizing noise comes from the use of a non-linear companding law.

Two algorithms are in use today for error reduction in the quantizing phase:

- **μ-Law (pronounced mu-law—also known as μ-255)**—Used in North America and Japan
- **A-Law**—Used in Europe and the rest of the world

These are known as *companding algorithms* because they effectively compress the PAM signal for error reduction on the transmitting end and expand the signal back to normal on the receiving end.

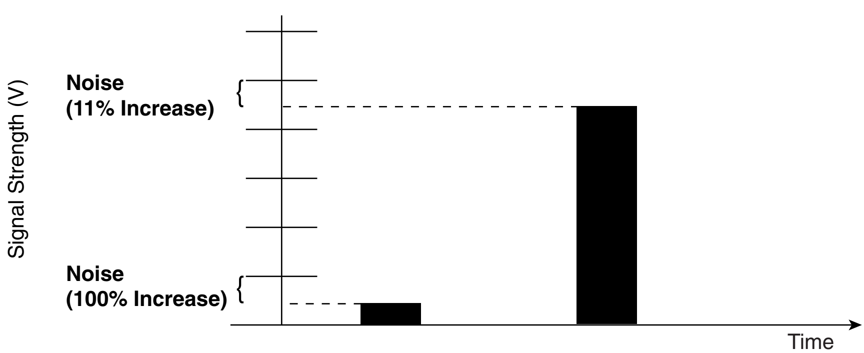
Analog information that has been quantized by one algorithm becomes incompatible with equipment that uses the other. It is common to convert between the two standards for communication between conflicting equipment. The digital signal level 0 (DS-0) created by North American or Japanese equipment can be converted to the DS-0 that is common in other parts of the world. Then, traffic that might have been multiplexed into a T1 or other T-carrier circuit can be multiplexed into an E1 or other related circuit. The responsibility of conversion usually falls on the party using μ-Law companding; meaning that international communication takes place with A-Law companding. A DS-0 can be defined as the 64-kbps bit stream that has not yet been multiplexed and is the direct product of the PCM process.

Remember that the process of quantizing exists solely to prepare the analog waveform for digitization during the encoding phase. You are limited to 8 bits per sample when encoding. If you take 8000 samples per second, you wind up with 64,000 bits per second (64 kbps) for each original analog waveform. Eight bits can vary 2^8 (256) ways, which means that you are allowed only 256 discrete points on the y-axis during the quantizing phase. If this logic is not apparent to you, consider that the word "bit" comes from a concatenation of the words "binary digit." This implies the use of the binary (or base-2) numbering system. Because there are only two possible values in this numbering system (0 and 1), the total number of possible values able to be represented by eight binary digits is found by raising two (the number of discrete values in the numbering system to which each digit can be set) to the power of eight (the number of digits being considered at one time—eight bits per sample). To further dilute the effect, these 256 points must be divided evenly between positive and negative pulses.

Depending upon whom you ask, you are liable to hear that toll-quality voice needs 4000 or more points on the vertical scale to reproduce the original signal on the other end in an acceptable manner. This assumes a linear relationship between the original PAM signal and the PAM signal that you use for encoding, which is a technical way of saying the two PAM signals are the same. Unfortunately, this approach would require 12 or more bits per sample during the encoding phase, which would result in higher bit rates, which in turn would result in higher circuit frequencies, which would lead to shorter circuit run lengths because of the increased attenuation of higher frequencies. Companding offers a compromise. Remember that companding would not be necessary if you could encode the information with 12 or more bits, instead of just 8.

The principle of companding is based primarily on the idea that lower amplitude PAM pulses are more sensitive to quantizing noise (a higher signal-to-noise ratio) than are higher amplitude pulses and, to a lesser degree, on the statistical probability that analog traffic presents with lower amplitudes (lower volume) most of the time. This simply means that in a linear quantizing arrangement (all intervals between points on the vertical axis are equal), the quantizing noise can represent a larger percentage increase/decrease in lower-amplitude pulses than in higher ones, which greatly affects the reproduced fidelity of low-volume information. As Figure 3-7 illustrates, an equal amount of quantizing noise, in absolute terms, winds up being an appreciably larger increase for the lower-amplitude pulse, in relative terms. This is the same concept as the perception of time to humans. As you get older, a year (although the same as every other year, in absolute terms) seems shorter because it represents a relatively shorter period, compared to the years of life you have stored away in your memory. To the toddler, however, that same absolute year seems like forever because it is a relatively larger portion of their existence.

Figure 3-7 *The Greater Effect on Low-amplitude Pulses*



Because the noise is relatively greater at these more popular and vulnerable amplitudes, concentrate on reducing the quantizing error at these amplitudes by placing the points on the vertical axis closer at the lower amplitudes than at the higher ones. This way, you have excellent quality for the majority of the traffic and surprisingly good quality for the rest. For those that would like to follow along mathematically, Figure 3-8 shows the formulas for computing the quantized PAM signal (y) from the original PAM signal (x), for both μ-Law and A-Law. For many of us, it suffices to merely understand that the two technologies are different and that conversion is required between them.

Figure 3-8 *Companding Equations*

μ-Law

$$y = \operatorname{sgn}(x) \; \frac{\ln(1+\mu\, |x|)}{\ln(1 = \mu)}, \; -1 <= x <= 1$$

sgn (x) : + or -, based on x
μ = 255 (hence the name μ-255)

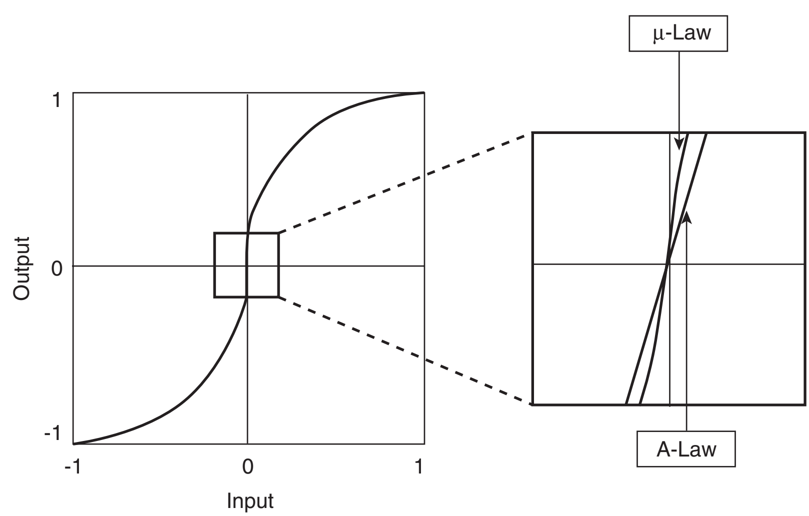
A-Law

$$y = \begin{cases} \operatorname{sgn}(x) \; \dfrac{A\, |x|}{1+\ln A}, & 0 <= |x| <= 1/A \\[2ex] \operatorname{sgn}(x) \; \dfrac{1+\ \ln A\, |x|}{1+\ln A}, & 1/A <= x <= 1 \end{cases}$$

A = 87.6

The ranges for PAM signals (x and y) are normalized to ±1 volt, for the purposes of these equations. Figure 3-9 shows these equations graphed. Pay special attention to the logarithmic S curve.

Figure 3-9 *Graphs Comparing A-Law and* μ*-Law*



In Figure 3-9, you also see that the A-Law curve is straighter near (0,0) than is the μ-Law curve. In reality, these algorithms approximate logarithmic behavior by specifying segments, within which quantized values are actually linear. Although technically there are 16 segments in these pseudo-logarithmic curves, eight in the positive quadrant and eight in the negative, toward the origin, segments are effectively combined because of the collinear nature of these segments.

The difference in the graphs of the two companding algorithms stems from the fact that A-Law combines more linear segments than does μ-Law. In fact, whereas μ-Law is considered a 15-segment curve (two are combined to form one), A-Law is credited with only 13 segments (four are combined). Although the difference in the equations causes the (x, y) coordinates to plot differently, both algorithms are made up of 16 quantum values (quantized values) in each of 16 linear segments, for the 256 total values represented by an 8-bit code.

The relationship between consecutive linear segments is fairly straightforward. Each segment exhibits half the slope of the preceding segment (it splits the difference between the previous segment and horizontal). Each consecutive segment also doubles the range of amplitudes covered by the previous segment. This also implies that the resolution is cut in half (the quantizing error can as much as double for a sample), because each segment has a fixed 16 quantum values that increasingly must occupy larger ranges of amplitude on the vertical axis.

In practice, both algorithms employ a scheme of 128 positive decimal quantum values (0 to 127) and 128 negative values (–0 to –127) in their quantizing scales, although these values map differently. This difference follows to the encoding phase. In addition, a few mathematical tricks are performed, in the case of encoding A-Law, all of which contribute to the incompatibility between the companding algorithms, and between the resulting PAM signals and encoded bit streams. One-way transmission is a common ramification of mismatched companding algorithms at opposite ends of a circuit.

NOTE There are 256 discrete values, 0 to 127 and –0 to –127. The actual zero signal level is not represented. The first bit is the sign bit, which allows for a value of 0 with a negative sign bit, basically –0, as the first value below the x-axis.

Encoding

The final phase of the conversion process is one that this chapter has been alluding to for a while. This is what it's all about. After this phase, you have a stream of binary digits that is the digital traffic for transmission across the digital circuit. First, you need to understand that the term *encoding*, as it is used here, does not mean the same thing as encoding when it applies to transforming a bit stream into pulses of electricity. That type of encoding is discussed in chapters relating to digital circuits.

This type of encoding refers to taking the adjusted (quantized) PAM signal and converting each sample into a stream of 8 bits, based on that sample's pulse amplitude. Similar to the quantizing scheme, the encoding method is based on the companding algorithm in use. As you will see, the 8-bit codes that each algorithm generates for the same quantum value are completely different.

Figure 3-10 shows a graphical representation of the segmented approach to encoding µ-Law quantum values. Although the bit patterns are opposite from what you might expect in the seven least significant bits (minimum is all 1s, whereas maximum is all 0s), it is easier to diagram than the A-Law process. Even though the A-Law algorithm equates values in a more logical way (zero basically means zero), the final product is the result of performing an exclusive OR (XOR) function with 0x55 (01010101). This process has its roots in days gone by but remains as an artifact of the original technology. Basically, during the transmission of a low-amplitude (silent) signal, the XOR operation ensures that pulses are still encoded, as opposed to the encoding of mostly 0s, which can jeopardize synchronization.

Figure 3-10 *The Segmented Encoding Process for µ-Law*

Bit 1 (Most Significant Bit)
Sample Polarity

Positive	1
Negative	0

Bits 2, 3, 4
Companding Segment #

0	1	1	1
1	1	1	0
2	1	0	1
3	1	0	0
4	0	1	1
5	0	1	0
6	0	0	1
7	0	0	0

Bits 5, 6, 7, 8
Quantizing Step

0	1	1	1	1
1	1	1	1	0
2	1	1	0	1
3	1	1	0	0
4	1	0	1	1
5	1	0	1	0
6	1	0	0	1
7	1	0	0	0
8	0	1	1	1
9	0	1	1	0
10	0	1	0	1
11	0	1	0	0
12	0	0	1	1
13	0	0	1	0
14	0	0	0	1
15	0	0	0	0

The segmented encoding process follows the same structure for both companding algorithms, but differs in the encoded values for similar sample-pulse amplitudes. The common structure is illustrated in Figure 3-10, but the values for bits two through eight are decidedly not the same. The first, or most significant, bit represents the PAM pulse's polarity. The next 3 bits represent the eight possible segments, based on the polarity already discussed. Yes, there are 16 segments in all, but recall that these are distributed as eight for each of the two polarities. Therefore, you obtain the unique values for the 16 segments when you consider the first 4 bits together (polarity and segment number). The last 4 bits represent the 16 linear steps in the segment indicated.

The conversion of the encoded value into a binary quantum value that ranges between the decimal values 0 and 255 can be accomplished by leaving bit one alone, the polarity, and inverting bits two through eight. For example, referring to Figure 3-10, an encoded value of 11000101 represents a quantized pulse in the fourth positive segment (marked segment 3) that falls closest to the eleventh quantizing step (marked 10 in the diagram). To turn this into a value between 0 and 255 in binary, leave the first bit alone and invert the remaining 7 bits. This results in 10111010, which is a decimal 186. On the positive scale, encode quantum values between 128 and 255, inclusive. The quantum value 186 is near the halfway point on the positive scale, which is where the description of the encoded value 11000101 would have placed the pulse.

What the preceding paragraph is talking about is μ-Law encoding. A-Law, although more difficult to illustrate, is fairly simple to explain. In fact, the quantum value matches the initially encoded value, so that no inversion of bits two through eight is necessary. The trick comes when the initially encoded value is obtained. Before a true A-Law bit stream is realized, you must XOR the initially encoded value with the hexadecimal value 55 (decimal 85), which in binary is 01010101. As you can see, the even bits are inverted in the final product. Figure 3-11 illustrates the use of the XOR Boolean operation.

Figure 3-11 *The XOR Operation*

	0		1		0		1
XOR	0	XOR	0	XOR	1	XOR	1
	0		1		1		0

In English, Figure 3-11 says that a single operand (and only a single operand) must be a 1 to get a yes (1) answer. Otherwise a no (0) is the result, even when both operands are 1s, which would return a yes answer if you were using either the OR or the AND operation, but not with XOR. Whenever a 0 is XORed with the original data, the same value as the original results, but when it is a 1, the opposite value is the solution. This is the phenomenon that causes the XOR of 0x55 to invert only the even bits for samples quantized and encoded by using the A-Law algorithm.

So, for the same quantum value as in the previous example, which was 186, the initial A-Law encoded value is actually decimal 186 or binary 10111010. This is read as the eleventh step in the fourth positive segment. Sound familiar? Remember that just because both μ-Law and A-Law describe a quantum value of 186 the same way, the two algorithms place this step of this segment at slightly different points on the scale. A quantized pulse that maps to 186 in one companding algorithm does not map to 186 in the other.

What this means is that, contrary to what you might think, the conversion process is not a simple numerical replacement. In fact, one of the simpler methods requires two conversions. The μ-Law information must first be converted to a 16-bit code, which provides sufficiently generic information. Then this 16-bit code can be converted into A-Law.

Never mind conversion headaches. You're simply trying to produce an A-Law byte from a sample. However, one more step remains. The 10111010 value must be XORed with 01010101. This produces 11101111, a value that does not directly resemble either the quantum value or the initially encoded byte. The one detail that has remained throughout the manipulation is the sample's polarity. The most significant bit has not been altered. For this reason, the polarity (most significant) bit has the same values in the same situations for both μ-Law and A-Law. The same cannot be said for bits two through eight.

Variations on the PCM Theme

PCM is great when you have an application that requires 56 to 64 kbps of digital bandwidth. Sometimes analog traffic can handle a little more loss of integrity. As long as you go in realizing integrity might be jeopardized, you can do quite a few things to preserve bandwidth and get the most out of your digital circuits. Some modern techniques make it difficult for the average human ear to detect anything different from the quality afforded by good old-fashioned PCM. This section introduces two fairly well established technologies that provide an alternative to standard PCM and allow more devices to communicate over the same medium, with minimal to no loss. Additional standards should not be difficult to grasp, after you understand the two presented here. They are differential pulse code modulation (DPCM) and adaptive differential pulse code modulation (ADPCM).

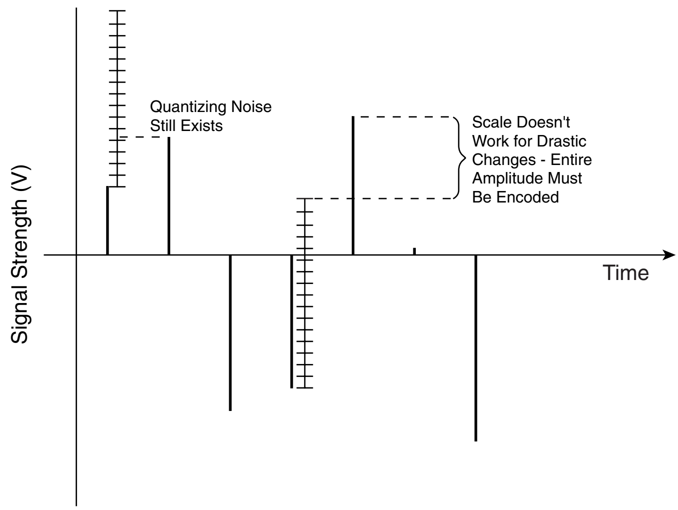
DPCM

DPCM capitalizes on the fact that the amplitudes of samples of analog information that are captured at the Nyquist rate (twice the limited maximum frequency of the analog source), or higher, tend to be close to one another a high percentage of the time. If this is the case, why not simply look at this difference between the amplitudes of adjacent samples, rather than the amplitudes of the samples themselves? Noting only this differential provides the ability to represent the samples in a way that uses fewer bits than if you were to directly encode the amplitudes of the samples, as in standard PCM. Figure 3-12 illustrates the concept of measuring only the difference between amplitudes of adjacent samples.

An important difference between these differential methods and standard PCM is that quantizing in DPCM and ADPCM is not performed on the actual sample pulses, but instead on the difference between the pulses. If the difference is small, as expected, the quantizing noise can actually be less, especially at higher amplitudes, than in the case of PCM. So with potentially less noise and fewer bits required for transmission (often only 3 or 4 bits per sample, as opposed to 8), it's no wonder these technologies are so popular.

There's no magic to this technology. It's mathematical prediction. Like any prediction, however, the results can be off sometimes, though more so with DPCM than ADPCM. DPCM has a simple prediction mechanism. It starts with a fully described initial value, as PCM does for all samples. It then predicts the next sample will be the same as the last. The fact that it is not the same is not a problem, because it only encodes the error in the prediction. The receiving end predicts by using exactly the same algorithm and circuitry as the transmitting end, so it understands how to reconstruct the actual pulse from the prediction error that is transmitted. The problem arises when there are not enough bits in the scheme to represent the prediction error. In other words, the change between sample amplitudes is off the scale.

Figure 3-12 *DPCM's Focus*

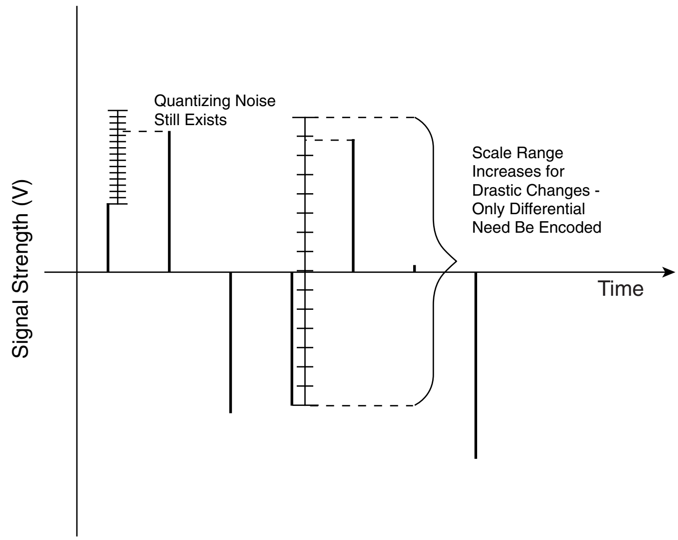


ADPCM

What happens during the less common occurrences when the change between amplitudes of consecutive samples is not close enough to be represented adequately with DPCM? Basically, the quantizing noise can be great. With DPCM, there is no mechanism to circumvent this situation, aside from encoding the actual sample, rather than what might have become an inaccurate prediction. Enter ADPCM. The adaptive part of ADPCM is an enhancement to DPCM. In general, everything works the same, but with ADPCM the range that the 3 or 4 bits represent can change, as necessary (see Figure 3-13).

With this ability for the encoding circuitry to alter the range, when the distance between amplitudes increases beyond what the current range can handle, the range can grow to accommodate the distance, although quantizing noise might increase. But the converse is also true. If the distance decreases, the range can decrease, likely reducing quantizing noise in the process. These alterations in noise occur because regardless of the range, with a fixed number of bits representing the difference in amplitudes, the number of points in the adaptive range remains the same. So the same fact that causes the wider ranges to be more loss-inducing causes the narrower ranges to be less so.

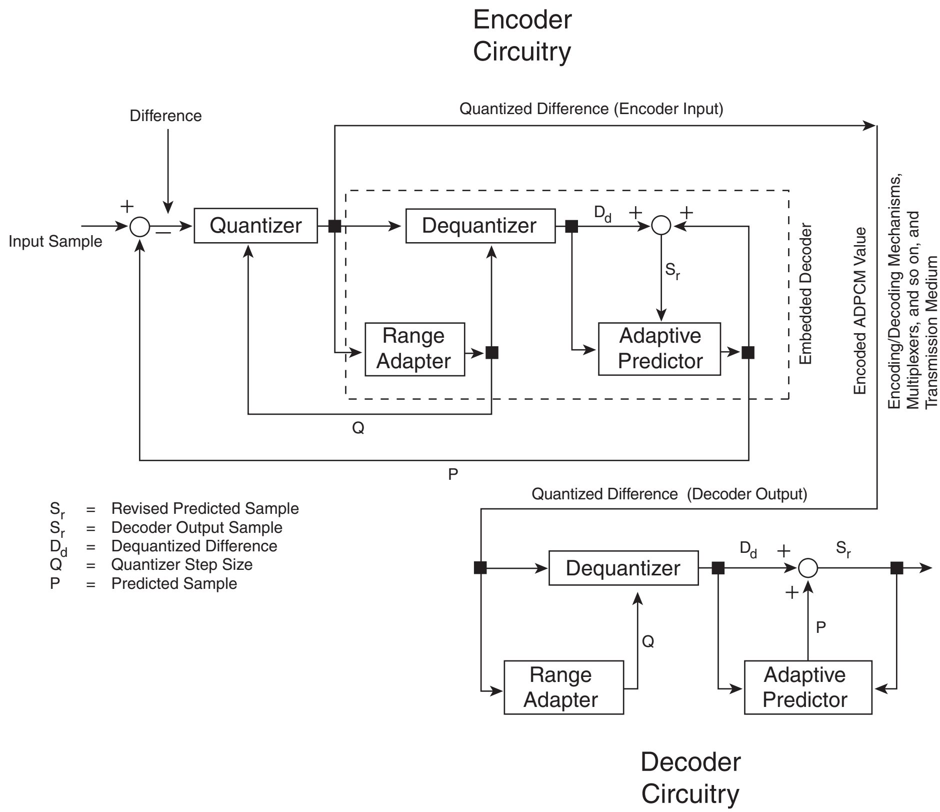
Figure 3-13 *ADPCM's Adjustable Range*



The prediction mechanism that ADPCM uses is a bit more complex than that of DPCM. In general, ADPCM keeps a running average of a set number of previous differences that were encoded. It keeps its prediction current by also incorporating two or more of the most recent predictions, which forms a weighted average. A good example of this is ITU-T standard G.721. In the case of G.721, the adaptive predictor forms an average by using the last six difference values before they are quantized (or more appropriately, dequantized by the embedded decoder circuitry) and the last two predicted values. Proprietary versions of ADPCM can work differently, although the principles are the same. In fact, G.721 (also see more current G.726) specifies only a 32-kbps ADPCM algorithm, although other implementations specify rates such as 40 kbps, 24 kbps, and 16 kbps.

What's this about the encoder containing a decoder? Well, one way to make sure the decoder at the receiving end knows what step size (what range of quantizing steps) the transmitting device is using is if the encoder in the transmitting device is in synchronization with what the decoder at the receiving end is expecting. This way, if the transmitter changes its quantizing range, the receiving end uses the same data to change its dequantizing range. Figure 3-14 shows a block diagram of a sample ADPCM encoder and decoder pair. The encoder contains the decoder's blocks as a subset. The encoder circuitry has updated its predictions and quantizing steps (range) for the next round, by using the same binary code that it is sending to the decoder. This ensures that the identical circuitry of the decoder uses this same information to arrive upon the same results as the encoder's embedded decoder.

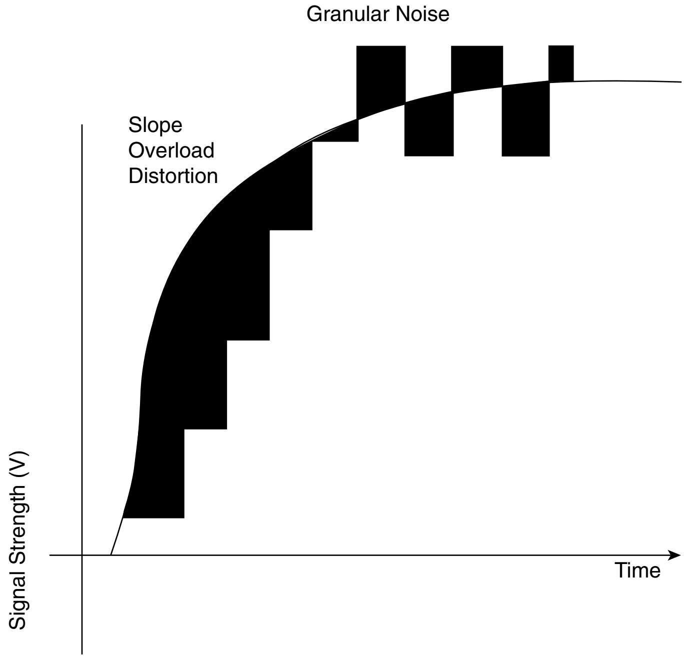
Figure 3-14 *Block Diagram of an ADPCM Encoder/Decoder Pair*



Other Modulation Schemes

Besides the popular versions of ADPCM (G.721 is definitely not the only one), several other encoding methods are in use today for various applications. One of these is Delta Modulation (DM). DM works on the basis that the relative change in the direction of the amplitude from one sample to the next is adequate information to successfully reconstruct the analog waveform at the receiving end. As a result, DM requires only a single bit to represent this directional information. The next sample's amplitude is greater than or less than the current sample's amplitude. It's as simple as that. What's not so simple is the solution for the unfortunate situations when the fixed step size assumed by the decoder on the receiving end becomes unfaithful to the reality of steeper or flatter slopes than the norm. Two forms of distortion error occur in these circumstances, slope overload and granular noise (see Figure 3-15).

Figure 3-15 *Delta Modulation Distortion*



With DM, each bit represents an entire sample. As a result, each one of these samples encoded as a bit is represented in a primitive way, with a relatively small amount of information being transmitted about the sample. This is the main reason that the noise distortion discussed here can be so damaging to fidelity. One way of keeping both types of noise within acceptable limits is to make the step size fairly small, while sampling at a much higher rate than the Nyquist rate. Because there is only one bit per sample, the bit rate matches the sampling rate. Therefore, if you are producing 32 kbps of traffic, you are using a sampling rate of 32,000 samples per second. You must take care not to completely erode the compression benefits, in the form of bandwidth savings, that this technique is designed to offer. Thus, DM alone cannot always effectively overcome these issues, but there is another technique based on DM that can. It's called Continuously Variable Slope DM (CVSD). CVSD allows for the monitoring of a flow-control bit stream, which can come in the form of all 1s to increase the step size to avoid slope overload or all 0s to decrease the step size to allow for a less steep slope, thus curtailing the effects of granular noise.

Other adaptive forms of DM simply watch for trends by remembering previous directions. When the same direction is followed for many samples, the prediction is made that the slope might be fairly flat. So, to decrease the effects of granular noise, the step size is decreased. Conversely, if the direction has been alternating between positive and negative, the indication is that the slope is greater than expected. In this case, the step size is increased to avoid slope overload distortion.

In addition to the time-domain methods discussed in this chapter, frequency-domain approaches also exist, with certain advantages. With Sub-Band ADPCM (SB-ADPCM), the input speech is split into several frequency bands, or sub-bands, and each is coded independently by using ADPCM on the isolated frequencies. At the receiving end, the bits are decoded and the sub-band frequencies are recombined, which yields the reconstructed speech signal. The advantages of doing this come from the fact that the noise in each sub-band is dependent only on the coding in that particular sub-band. As a result, you can use tighter encoding schemes and quantizing steps with sub-bands that are perceived as more important to the human listener. In this way, the noise in these frequency regions is low, whereas in other sub-bands you can allow a higher level of noise, as these frequencies are perceived as less important. Sub-band *codecs* (coder/decoder) produce toll-quality speech by using only 16-32 kbps. Because of the filtering necessary to split the speech into sub-bands, these codecs are more complex than simple DPCM encoders and introduce more coding delay. The complexity and delay are still relatively low, when compared to most hybrid codecs.

Summary

In this chapter, you learned the history and motivation behind the development of digital transmission and storage technology. You also learned why digital formats improve upon the quality offered by comparable analog technologies.

You were presented with each stage of the conversion process for producing a digital bit stream from an analog source. These stages are as follows:

- **Filtering**—Where only the standardized voice frequency is allowed to enter the process
- **Sampling**—Where the analog waveform is observed 8000 times per second, producing another form of analog signal known as the PAM signal
- **Quantizing**—Where each sample's pulse in the PAM signal is adjusted to match one of 256 discrete levels on the vertical axis, at which time the companding process is applied to the PAM signal, an event that must be reversed at the receiving end
- **Encoding**—Where the adjusted PAM signal is converted into a bit stream, by assigning an eight-bit code to each of the 256 quantization levels from the previous stage

Finally, this chapter presented alternate technologies that have their roots based in the analog-to-digital conversion technology outlined in the beginning of this chapter. These technologies include DPCM, ADPCM, DM, CVSD, and SB-ADPCM.

Review Questions

Give or select the best answer or answers to the following questions. The answers to these questions are in Appendix A, "Answers to Review Questions."

1 Which of the following are examples of the difference between analog and digital communication? (choose two)

- **a** The term frequency only applies to digital communication.
- **b** The term bit rate only applies to digital communication.
- **c** The quality of an analog signal is generally better than that of a digital signal.
- **d** The quality of a digital signal is generally better than that of an analog signal.
- **e** Circuits and sending and receiving equipment are identical in analog and digital technologies.

2 Which of the following is *not* a stage in the analog-to-digital conversion process?

- **a** Sampling
- **b** Encoding
- **c** Filtering
- **d** Pulse conversion
- **e** Quantizing

3 What is the earliest stage in the conversion process at which the signal technically can be considered digital?

- **a** Sampling
- **b** Quantizing
- **c** Filtering
- **d** Encoding
- **e** Pulse conversion

4 Which of the following ITU-T recommendations covers PCM in general?

a G.711

b G.721

c G.726

d G.729

5 Which of the following advanced processes adapts the quantizing scale to more closely match the wider or narrower variation between adjacent pulses?

a PCM

b DPCM

c ADPCM

d DM

e CVSD

f SB-ADPCM

INDEX

Symbols & Numerics

μ-Law, 88–91

1s density, 112
2B1Q line coding, 244
48-octet ATM payload, 300
4B3T line coding, 245
64-kbps loop rate (DSS), 115

A

AAL (ATM adaptation layer), B-ISDN model, 319, 350
  AAL0, 351
  AAL1, 351–353
  AAL2, 353–355
  AAL3/4, 355–360
  AAL5, 361–363
  CS, 319
  SAR, 320
AAL-indicate bit, 310
ABR (available bit rate), 337–341
  closed-loop flow control, 338
  service category, 366
AC (access control) field
  DQDB slots, 380
  SIP Layer 2 PDUs, 394
access classes, SMDS, 376
access signaling specifications, Frame Relay, 273
AD (Adjunct), 64
adaptation layer, 317
adaptive QAM (quadrature amplitude modulation), CAP, 416
Address Complete messages (TUP), 58
address field (DXI), 404
Address messages (DUP), 55
address octets (Q.921), 221
address screening, 390
addressing (SMDS), 376
  subscriptions, 388–390
admission control, 479
ADMs (add/drop multiplexers), 498
ADPCM (adaptive differential pulse code modulation), 95–96
  SB-ADPCM, 99
ADSL (asymmetric digital subscriber line), 432
  components of, 419–420
AERM (alignment error rate monitor), 43
AF (ATM Forum), 300
  Traffic Management Specification, service categories, 365–366
  UNI 3.x/4.0 specification, 304
  UNI IAs, 303
alarm conditions (T1), troubleshooting, 146–147
A-Law, 89–91
alias point codes, 38
alignment process, Q.921, 224–227
A-links (access links), 36
AMI (alternate mark inversion) encoding, 42
  1s density, 112
  DSS implementation, 110–111
  pulse stuffing, 134
amplifier cascades, 448
amplitude, 4
analog communication, 3
  amplitude, 4
  attenuation, 6–9
  DACs, 10
  distortion, 5–6
  E&M signaling, 13
  EMI, 9
  frequency, 5
  GR-303-CORE standard, 17–18
  ground start signaling, 13
  IDLCs (Integrated Digital Loop Carriers), 14–16
  interference, 81
  loop start signaling, 13
  noise, 7–9

R2, 169
reverse battery signaling, 13
sampling, 85
sinusoidal waves, 82
tip and ring circuits, 11–14
versus digital, 3, 81–82
wavelength, 5
ANSI (American National Standards Institute)
signaling links versus ITU signaling links, 35
ANSI T1.410-1992. *See* DDS
AO-DI (Always On Dynamic ISDN), 250
APCs (Adjacent Point Codes), 38
applications, SDH/SONET, 522–523
architectures
ISDN, 212
circuit switching, 213–215
local loop, 213–216
reference points, 218–219
DQDB AU, 378
DSL
CPE, 437
NAP, 437
NSP, 437–440
SMDS, 374
AS (address screening), 376
AS bearer channels (ADSL), 427
associated signaling, 29
asynchronous circuits, T3, 182
asynchronous technologies versus synchronous, 105
AT (access tandem) switches, 25
ATM (Asynchronous Transfer Mode)
AAL, 350
AAL0, 351
AAL1, 351–353
AAL2, 353, 355
AAL3/4, 355–360
AAL5, 361–363
B-ISDN model, 316
AAL, 319–320
ATM layer, 319
C-plane, 317
M-plane, 317
physical layer, 318–319
U-plane, 317
development of
B-ISDN origins, 301
consensus on 48-octet payload, 300
standards bodies, 301–302
error control modes, 314
header fields, 305
CLP, 313
GFC, 306
HEC, 313, 316
PT, 309–313
VCI, 306–308
VPI, 306
interfaces, 304
QoS
CAC, 320
CER, 325
CLR, 323
maxCTD, 323–325
peak-to-peak CDV, 325
SECBR, 326
reference points, 304
service classes, 331
ABR, 337–341
CBR, 333–334
GFR, 343
rt-VBR, 334–336
UBR, 341–342
traffic management, 344
continuous-state leaky bucket algorithm, 347
GCRA, 345–346
PCR algorithm, 349
policing, 345
SCR algorithm, 350
traffic parameters, 327
BT, 329
CDVT, 329–331
connection-traffic descriptors, 328
MBS, 329
MCR, 329
MFS, 329
PCR, 328
SCR, 328
versus STM, 303

ATM layer (B-ISDN model), 319
attenuation, 6–9, 448
  in DSL circuits, 421
ATU-R (ATU-Remote), 419
AUs (access units), 377–379

B

B (Bearer) channels, 213
B8ZS (bipolar 8-zero substitution), 133
BACP (bandwidth allocation control protocol), 243
bandwidth reservation, CAC, 320–321
BAP (bandwidth allocation protocol), 243
baseline privacy, 479
BASize field (SIP Layer 3 PDUs), 387
Bc (committed burst size), 282
BE (Beginning-End) Tag fields, SIP Layer 3 PDUs, 386
Be (excess burst size), 282
bearer capability IE (Q.931 messages), 234–236
bearer channels, DSL, 427
bearer traffic, 22
Bellcore, role in SMDS development, 373–374
BERT (bit error rate test), 143–144
B-ICI (Broadband Inter-Carrier Interface), 304
billing methods, ISDN BRI, 254–255
bipolar line coding schemes, 83
bipolar signals versus unipolar, 108
B-ISDN model, 300, 316
  AAL, 319, 350
    AAL0, 351
    AAL1, 351–353
    AAL2, 353–355
    AAL3/4, 355–360
    AAL5, 361–363
    CS, 319
    SAR, 320
  ATM layer, 319
  C-plane, 317
  M-plane, 317
  physical layer, 318
    PMD sublayer, 318
    TC sublayer, 318–319
  U-plane, 317
bit meanings of DLCIs, 280–282
bit stuffing, 43, 185
  DS3 PLCP frames, 400
BITS (Building Integrated Timing Supply), 494–495
B-links (bridge links), 36
BLSR (bidirectional line-switched ring), 523
blue alarms (T1), 147
BNRZ (bipolar non-return to zero), 133
BOD (bandwidth on demand), 243
boosting DSL power levels, 429
boundary detection, ATM cells, 318
BPVs (bipolar violations), 112, 134
BRI (Basic Rate Interface)
  AO-DI, 250
  billing methods, 254–255
  circuit provisioning, 248–249
  configuring, 250–252
  echo cancellation, 216–217
  features, 243–244
  FX, 249
  Japanese implementations, 244
  line coding
    2B1Q, 244
    4B3T, 245
  NIDs, 217
  operation, 242–243
  reference points, 218–219
  SPIDs, 248
  troubleshooting, 252–254
bridge taps, 131, 434
broadcast television
  frequency allocation, 451–453
  standards, 451
BRZ (bipolar return to zero), 84
BT (burst tolerance), 329
bursty traffic
  AAL3/4, 355–356
  errors, 424
buses (DQDB), HoB, 379
byte-interleaved multiplexing, 492, 504

C

cable IP network networks, 484
DPT/RPR architectures, 486
Ethernet aggregation architectures, 486
future of, 487
high-speed optical core, 487
OCx architectures, 486
cabling
home run, 254
length, effect on DSL performance, 435

cable modem initialization process
DHCP request/response, 471–472
frequency scanning, 468
online state, 473
power ranging, 470–471
registration request, 472–473
TOD, 472
UCD receipt, 470
cable plant, 447
frequency allocation, 451–453
headend, 450–451
hubs, 451
quality requirements, 453–454
RF signals, 455
upstream noise, 455–456
cable throws, 253
CAC (connection admission control), 320–321
calculating
cell arrival times, 346
CMR, 327
noise margin in DSL circuits, 421
Call Accept Message (DUP), 55
call control
Frame Relay, 292
R2 signaling, 167
call flow, ISDN, 232–233
call reference length indication octet (Q.931 messages), 230
Call Rejected Message (DUP), 55
called/calling number IE (Q.931 messages), 238–240
Calling Line Identity message (DUP), 55
CAP (carrierless amplitude and phase modulation), 416
capabilities of ISDN BRI, 243–244
capability point code, 38
capacitance, 111
CAS (channel associated signaling), 22, 28, 229
CATV, 446
cable plant, 447
frequency allocation, 451–453
headend, 450–451
hubs, 451
quality requirements, 453–454
RF signals, 455
upstream noise, 455–456
DOCSIS specification, 456
baseline privacy, 479
binary configuration files, 473–478
downstream signaling, 465
FEC, 459–460
fiber node combining, 482–484
interleaving, 460
MAC domains, 463–464
modulation, 458–459
MPEG framing, 461
security features, 480–482
throughput, 461–463
upstream admission control, 479
upstream signaling, 466–467
versus Euro-DOCSIS, 457
headend, 447, 450
HFC networks, 449–450
signal attenuation, 448
C-bits, 114, 195–197
CBR (constant bit rate), 333
conformance definition, 334
service category, 365
CCAF (Call Control Agent Function), 62
CCS (common channel signaling), 22, 29, 229. *See also* SS7
CDVT (cell delay variation tolerance), 327–331
CE (carrier equipment), SMDS, 374–375
cell-rate decoupling, 319

cells, 303
  arrival times, calculating, 346
  boundary detection, 318
  clumping, 330
  discarding
    EPD, 342
    SD, 342
  entry/exit events, 323
  header fields, 305
    CLP, 313
    GFC, 306
    HEC, 313, 316
    PT, 309–313
    VCI, 306–308
    VPI, 306
CER (cell error ratio), 325
CFAs (carrier failure alarms), 146
channel associated signaling (CAS), 22, 28, 229
channel banks, 136
  D1, 137–139
  D1D, 140
  D3, 140
  D4, 140
  D5, 140–142
channel ID IE (Q.931 messages), 237–238
channelized T3, 182–184
  Cisco 1000 ESR configuration, 198–199
Charging Message (TUP), 58
check bytes, 422
CICS (Customer Information Control Systems), 35, 57
CIR (committed information rate), 282
circuit provisioning, 248–249
Circuit State Message (DUP), 55
circuit switching, ISDN, 213–215
  reference points, 218–219
circuit testing patterns, 117–118
circuits (DSL)
  attenuation, 421
  bridge taps, 434
  bursty errors, 424
  cable length, 435
  coding gain, 425
  crosstalk, 435
  dribbling errors, 425
  Fastpath, 425
  FEC, 422–424
  interfering signals, 434
  loading coils, 434
  noise margin, 421–422
  resistance, 436
  RFI, 434
Cisco routers, DSS/SW56 configuration, 118–119
Class 5 SSPs, 25
Clear messages (DUP), 55
CLI (Cumulative Leakage Index), 455
C-links (cross links), 36
cloned MAC addresses, detecting, 481
closed-loop flow control model (ABR), 338
closed topology (DQDB), 379
CLP (cell loss priority) field, 313
CLR (cell loss ratio), 323
clusters, 39
CMI (coded mark inversion), 175
CMTS (cable modem termination system), CPE database, 482
coaxial cable networks
  HFC, 449–450
  signal attenuation, 448
coding gain, 425
co-directional interfaces, 156
combining, 482–484
common channel signaling (CCS), 22, 29, 229
Common Part (AAL3/4), 356
common path distortion, 456
companding algorithms, 88–91
concatenation, 467
  SDH versus SONET, 504
configuring
  DSS/SW56, 118–119
  E1 controllers, 163–166, 173–174
  E3 interfaces, 205–206
  Frame Relay on Cisco devices, 292–295
  ISDN BRI, 250–252
  ISDN PRI, T1/E1, 258–259
  SMDS, 405–406

T1 controllers, 147
T1 hairpin connections, 148
T3 on Cisco 1000 ESR, 198–199
conformance definitions
CBR, 334
nrt-VBR, 337
rt-VBR, 336
congestion avoidance mechanisms, Frame Relay, 286–289
congestion control
closed-loop flow control, 338
dual leaky bucket model, 336
EPRCA, 339
GCRA, virtual scheduling algorithm, 346–347
connection status, Frame Relay, 289–291
connection-traffic descriptors, components of, 328
connectors, RJ-48X, 129
constellations, 415
Trellis coding, 429–430
control field (DXI), 404
control octets, 222–223
controlled frame slip, 126
controllers (T1), configuring, 147
conversion process
encoding, 92–93
sampling, 85–87
converters, D/A, 9–10
core access specifications, Frame Relay, 272
corrupted frames, 323
COs (central offices), 14
cost benefits of SW56, 115
COTs (Continuity Tests), 56
CPCS (common part convergence sublayer), 320
functions, 359–361
CPE (customer premises equipment)
addressing, 376
database, 482
SMDS, 374
C-plane (B-ISDN model), 317
CRC-4 framing, E1 circuits, 160
CRC-32 field (SIP Layer 3 PDUs), 391
crosstalk, 8, 435
CS (convergence sublayer), 319
CS-1 (Capability Set-1), 65–66
CS-2 (Capability Set-2), 66–68
CSES (consecutive severely errored seconds), 146
CSU (customer service unit), 107, 117–118
CSU/DSU (channel service unit/data service unit), 128
CTD (cell transfer delay), 323–324
customer premises
ISDN installation, 254
NIDs, 217
CVSD (Continuously Variable Slope DM), 98–99

D

D (Data channels), 213
D1 channel banks, 137–139
D1D channel banks, 140
D3 channel banks, 140
D4 channel banks, 140
D5 channel banks, 140–142
DA (Destination Address) field (SIP Layer 3 PDUs), 387
DACs (digital-to-analog converters), 10
DACS (Digital Access and Crossconnect System), 497
data-link layer functionality, Q.921 specification, 220–224
alignment process, 224–227
timers, 227–228
DB-15 connectors, 157
DBR (deterministic bit rate), 333
dBs (decibels), 7
logarithmic signal gain/loss, 131
mid-span repeaters, 216
DDS (digital dataphone service). *See also* SW56
circuit deployment, 105
circuit testing patterns, 117–118
Cisco router configuration, 118–119
CSU, 107
DSU, 108
line coding
64-kbps loop rate, 115

        - AMI, 110–111
        - naming conventions, 112
        - secondary channels, 113
    - local loop, 106
    - loop rates, secondary channel, 106
    - nailed architecture, 109
    - OCU-DP, 108–109
    - point-to-multipoint connections, 109
    - timing sources, 112
    - transmission rates, 106
    - transmit filter, 106
- default Q.921 timer values, 227
- deferred blocking, GR-303-CORE standard, 17
- delay parameters (QoS)
    - CER, 325
    - maxCTD, 323, 325
    - peak-to-peak CDV, 325
    - SECBR, 326
- dense combining, 482
- dependability parameters (QoS), CLR, 323
- deploying
    - DDS circuits, 105
    - T3, options, 193
- detection state, 314
- development
    - of ATM
        - B-ISDN origins, 301
        - consensus on 48-octet payload, 300
        - standards bodies, 301–302
    - of Frame Relay, 266–270
    - of SMDS
        - DQDB, 377
        - time-to-market delay, 385
- devices
    - ATM, error control modes, 314
    - cable modems, intialization sequence, 468–473
    - Cisco routers, configuring for DSS/SW56, 118–119
    - cloning, 480
    - CSU/DSU, circuit testing patterns, 117–118
    - digital repeaters, 215
    - DSL modems, trainup, 426
    - DSS
        - CSU, 107
        - DSU, 108
        - OCU-DP, 108–109
    - echo cancellers, ISDN BRI deployment, 216–217
    - IMTs, 27
    - ISDN BRI hardware, 242
    - mid-span repeaters, 215–216
    - network nodes, clusters, 39
    - NIDs, 11, 217
    - NT12s, 218
    - optical networking components, 495
        - LTE, 498–499
        - optical routers, 500
        - PTE, 496
    - STE, 500
    - POTS splitters, 419
    - required for SMDS CE, 375
    - SLT (Signaling Link Terminal), 43
    - SS7
        - SCPs, 27
        - SSPs, 23–25
        - STPs, 27
    - sub-band codecs, 99
    - TE1, 219
- digital communication
    - black and white analogy, 81
    - conversion process
        - encoding, 92–93
        - quantizing process, 87–91
        - sampling, 85–87
    - evolution of, 81
    - impetus for, 80
    - pulse streams, 85
    - quantizing process, noise reduction, 88–91
    - sampling, 87
    - signal conversion process, 83–84
    - versus analog, 81–82
- digital repeaters, 215
- digital signaling
    - R2, 169
    - versus analog, 3

discarding cells, SD, 342
discards (Frame Relay), 284–286
discrete pulses, 83–84
discrete values, 91
distance-insensitive transport technologies, 372
distributed architecture, INs, 62
divide and conquer, 456
DLCIs, (data-link connection identifiers), 276–278
    bit meanings, 280–282
    Inverse ARP, 279–280
D-links (diagonal links), 36
DM (Delta Modulation), 97
DMT (discrete multi-tone modulation), 417–418
DOCSIS cable specification, 456–457
    baseline privacy, 479
    binary configuration files, 473–474
        DOCSIS 1.0, 474–475
        DOCSIS 1.1, 475–478
    cable modem initialization process
        DHCP request/response, 471–472
        frequency scanning, 468
        online state, 473
        power ranging, 470–471
        registration request, 472–473
        TOD, 472
        UCD receipt, 470
    cable plant quality requirements, 453–454
    CMTS, CPE database, 482
    FEC, 459–460
    fiber node combining, 482, 484
    interleaving, 460
    MAC domains, 463
        downstream signaling, 465
        SID, 464
        upstream signaling, 466–467
    modulation, 458–459
    MPEG framing, 461
    security features, 480, 482
    throughput, 461–463
    upstream admission control, 479
DPC (Destination Point Code), 38
DPCM (differential pulse code modulation), 94–95
DQDB (Distributed Queue Dial Bus), 377
    AU, 377–379
    DMPDUs, format, 384
    IMPDUs, SAR, 382, 385
    MAC layer functionality, 380–382
    topologies, 379
dribbling errors, 425
drop and insert, 164
DS0 (digital signal level zero), 88, 109, 127
DS1 (digital signal level one) media, PLCP frames, 397–399
DS2 (digital signal level two) media
    E12 multiplexing, 200
    frame structure, 187
    M-Frame, bit stuffing, 188
DS3 (digital signal level three) media
    bit stuffing, 400
    circuits, 185
    M12 multiplexing, 185
    PLCP frames, 399–400
DS4 (digital signal level four), 206
DSL (digital subscriber line)
    architectures
        CPE, 437
        NAP, 437
        NSP, 437–440
    bridge taps, 434
    cable length, effect on performance, 435
    CAP, 416
    circuit attributes
        attenuation, 421
        bursty errors, 424
        coding gain, 425
        dribbling errors, 425
        Fastpath, 425
        FEC, 422–424
        noise margin, 421–422
    crosstalk, 435
    DMT, 417–418
    flavors
        ADSL, 432
        EtherLoop, 432
        G.Lite, 432

HDSL, 432
HDSL2, 433
IDSL, 433
RADSL, 433
SDSL, 433
VDSL, 433
framing modes, 427–428
higher-layer protocol transport mechanisms, 439–440
history of, 414
interfering signals, 434
loading coils, 434
modem trainup, 426
power levels, 428
reducing/boosting, 429
Trellis coding, 429–430
QAM, 415–416
resistance, effect on performance, 436
RFI, 434
wholesale services, 441
DSLAM (DSL access multiplexer), 419
DSU (data service unit), 108
circuit testing patterns, 117–118
DTMF (dual tone multifrequency), 12, 170
SW56 implementation, 116
DTP/RPR architectures, 486
dual leaky bucket congestion control, 336, 348
dual-latency modem, DSL data transmission, 427
DUP (Data User Part), 54–55
DWDM (dense wavelength division multiplexing), 521
DXI (Data Exchange Interface) protocol, 375, 403
framing, 404–405
heartbeat polling, 405–406
troubleshooting, 407

E

E&M signaling, 13
E.164 numbers, 213
E1 circuits, 154. *See also* T1 circuits
controllers
configuring, 163–166
R2 signaling, 173–174
CRC-4 framing, 160
line coding, 161
NO-CRC-4 framing, 158
physical characteristics, 155–157
PLCP frames, 401
PRI configuration, 258–259
subrate offerings, 157
E3 circuits, 199–200
E12 multiplexing, 200
E23 multiplexing, 202
G.804 framing, 205
G.823 framing, 202–204
interface configuraiton, 205–206
E12 multiplexing, 200
E23 multiplexing, 202
echo cancellation, 216–217
Elastic Networks, EtherLoop, 432
E-links (extended links), 36
EMI (electromagnetic interference), 8
in ISDN deployments, 215
en bloc signaling, 59
encapsulation, SMDS, 374
encoding schemes, 92–93. *See also* line coding
DSL
CAP, 416
DMT, 417–418
QAM, 415–416
encryption, 479
end-to-end signaling, 59
entry events, 323
EPD (early packet discard), 342
EPRCA (enhanced proportional rate control algorithm), 339
ERM (explicit rate marking), 340
error detection, Frame Relay FCS field, 285
error detection code field (OAM), 311
error mitigation
FEC, 459–460
interleaving, 460
errors, T1, 145–146
ES (errored second), 146

ESF (Extended Superframe), 136
 D5 channel banks, 140–142
estimating DSL power levels, 428
EtherLoop, 432
Euro-DOCSIS, 457
 modulation, 458–459
 throughput, 461–463
exit events, 323
expected cell arrival times, calculating, 346
extreme-dense combining, 482

F

fast byte, 427
Fastpath, 425
FCC First Report & Order, 69
FCS (frame-check sequence) field
 DXI, 404
 Frame Relay packets, 285
FDM (frequency-division multiplexing), 127, 450
FEBEs (far-end block errors), 161,
FEC (forward error correction), 422–423, 459–460
 interleaving, 423–424
FEs (functional entities), 62
FEXT (far-end crosstalk), 9, 435
F-GCRA (Frame-GCRA), 349
fiber-node combining, 482–484
fiber-optic cabling, 501, 504, 522
fields
 of ATM headers, 305
  CLP, 313
  GFC, 306
  HEC, 313, 316
  PT, 309–313
  VCI, 306–308
  VPI, 306
 of DXI frames, 404–405
 of Q.931 messages, 230
 of SIP Layer 1 PDUs, 396
 of SIP Layer 2 PDUs, 393
  Access Control field, 394
  MID field, 395
  NCI field, 394
  Payload CRC field, 396
  Payload Length field, 395
  Segment Type field, 394
  Segmentation Unit field, 395
  Sequence Number field, 395
 of SIP Layer 3 PDUs, 386
  BASize field, 387
  BE Tag fields, 386
  CRC-32 field, 391
  DA field, 387
  HE field, 391–392
  HLPID field, 390
  Information field, 392
  Length field, 387
  PAD field, 390
  QoS field, 391
  Reserved field, 386
  SA field, 387–388
filtering, 84
FISU (Fill-In Signal Unit), 43–45
fixed-length slots (DQDB), 380–382
flag field (DXI), 404
flavors of DSL
 ADSL, 432
 DSLVDSL, 433
 EtherLoop, 432
 G.Lite, 432
 HDSL, 432
 HDSL2, 433
 IDSL, 433
 RADSL, 433
 SDSL, 433
F-links (fully associated links), 36
FLR (frame loss ratio), 323
format
 of DMPDUs, 384
 of Q.931 messages, 230
format errors, T1, 145–146
fragmentation, 467
Frame Relay
 access signaling specifications, 273
 ATM interworking, 302
 configuring on Cisco devices, 292–295

congestion avoidance mechanisms, 286–289
core access description specifications, 272
DCLI bit meanings, 280, 282
development of, 266–270
discards, 284–286
DLCI, 276–280
framework specifications, 271
LMI specification, 289–291
service description specifications, 271–272
signaling, 286–292
SLAs, contractual values, 282
VCs, 275–276
framework specifications, Frame Relay, 271
framing
DSL, 427–428
DSS, bit values, 114
DXI, 404–405
Q.921, octets, 221–222
SDH, 519–520
T1, 135
D1 channel banks, 137–139
D1D channel banks, 140
D3 channel banks, 140
D4 channel banks, 140
D5 channel banks, 140–142
T3, 195–197
unchannelized E3
G.804, 205
G.832, 202–204
unmarked frames, 343
frequency, 5, 84
FRF (Frame Relay Forum), 268– 270
ATM/Frame Relay interworking, 302
IAs, 273–274
full async framing (DSL), 427
full booking, 321
full overhead framing (DSL), 427
full sync framing (DSL), 427
function-specific field (OAM), 311
future of cable IP networks, 487
FX (foreign exchange), 249

G

G.804 framing, 205
G.832 framing, 202–204
G.Lite, 432
GCRA (generic cell rate algorithm), 345
F-GCRA, 349
PCR algorithm, 349
SCR algorithm, 350
General Forward Setup Information message (TUP), 58
GFC (generic flow control) field, 306
GFR (guaranteed frame rate), 343
GPS (global positioning system) timing distribution, 493
GR-303-CORE standard, 17–18
ground start signaling, 13

H

hairpin connections, configuring
E1, 164
T1, 148
hardware
optical network components, 495
LTE, 498–499
optical routers, 500
PTE, 496
STE, 500
ISDN, 242
HDB3 line coding, 258
HDSL (high data-rate DSL), 432
HDSL2, 433
HE (Header Extension) field (SIP Layer 3 PDUs), 391–392
headend, 450–451
heartbeat polling (DXI), 405
HEC (header error correction) field, 313, 316
hertz, 84
HFC (hybrid-fiber coaxial) networks, 449–450
hierarchy of national SSPs, 25
higher-layer transport mechanisms, 439–440

high-order address octets (Q.921), 221
high-speed optical core networks, 487
history
of DSL, 414
of Frame Relay, 266–270
of NADH, 181
of SMDS, 372
alignment with 802.6, 373, 385
SIG, 374
HLPID (Higher Layer Protocol Identifier) field (SIP Layer 3 PDUs), 390
HoB (head of bus), 379
home run, 254
hubs, 451
hunt state, 315
hybrid SSPs, 23

I

IACR (initial allowed cell rate), 338
IAs (Implementation Agreements), 268, 273–274, 300
B-ICI 2.0, 304
UNI-related, 303
IDSL (ISDN-based DSL), 433
IETF (Internet Engineering Task Force) RFC 1209, 374
IMPDUs (Initial Meeting Access Control PDUs), SAR, 382, 385
impulse noise, 455
IMTs (Inter-Machine Trunks), 27
INs (intelligent networks)
CS-1, 65–66
CS-2, 66–68
FEs, 62
PEs, 63–64
information elements, Q.931 messages, 233–240
information field
DXI, 404
SIP Layer 3 PDUs, 392
ingress noise, 455
interfaces, ATM, 304
interference
analog communication, 81
EMI
ISDN carriers, 215
effect on DSL performance, 434
interleaving, error mitigation, 460
international digital hierarchies, JDH, 174–176
international split plans, 452
international SSPs, 23, 25
international television standards, 451
interregister signaling, R2 signaling, 170–173
interworking
ATM/Frame Relay, 302
SMDS and Layer 3 protocols, 377
Inverse ARP, 279–280
IP (Intelligent Peripheral), 64
ISDN
architecture, 212
BRI
2B1Q line coding, 244
4B3T line coding, 245
AO-DI, 250
billing methods, 254–255
circuit provisioning, 248–249
configuring, 250, 252
features, 243–244
FX, 249
Japanese implementations, 244
operation, 242–243
SPIDs, 248
troubleshooting, 252–254
circuit switching, 213–215
E.164 numbers, 213
echo cancellation, 216–217
local loop, 213–215
mid-span repeaters, 216
NID, 217
NT12s, 218
PRI, 255–256
NFAS, 257
Q.Sig, 260
T1/E1 configuration, 258–259
troubleshooting, 259–260

Q.921 specification, SAPs, 221
reference points, 218–219
specifications
ITU Q.921, 220–228
ITU Q.931, 229–233
physical layer, 219
S/T bus, 218–219
TE1 devices, 219
ISUP (ISDN user part), 58
messages, 59–60
signaling methods, 59
ITU (International Telecommunications Union) digital hierarchy
E3, 199–200
E12 multiplexing, 200
E23 multiplexing, 202
G.804 framing, 205
G.832 framing, 202–204
interface configuration, 205–206
point code format, 40
signaling links, 35
ITU-T (International Telecommunications Union Telecommunications Standardization Sector)
ATM recommendations, 302
recommendation I.121, B-ISDN model, 316–320

J

Japanese BRI services, 244
JDH (Japanese Digital Hierarchy), 174–176
jitter, 146, 330, 467

K

KEK (key encryption key), 480
Kentrox, development of SMDS access solution, 386

L

LAPD (Link Access Procedure on the D channel), 220–224
Q.921 alignment, 224–227
Q.921 timers, 227–228
laser clipping, 456
latency
interleaving delay, 424
real-time polling, 467
Layer 3 protocols, interworking with SMDS, 377
layer management (M-plane), 317
layers of AAL
AAL0, 351
AAL1, 351–353
AAL2, 353–355
AAL3/4, 355–360
AAL5, 361–363
layers of MTP protocol
DUP, 54–55
ISUP, 58–59
MTP1, 42–43
MTP2, 43
FISU, 43–45
link alignment, 48–52
LSSU, 45–47
MTP3, 52–54
TUP, 56–58
LBO (long-haul line build-out), 130
LEDs (light emitting diodes), multimode fiber, 502
legislation, Telecommunication Act of 1996, 68
Length field (SIP Layer 3 PDUs), 387
line coding, 258
AMI
1s density, 112
DSS implementation, 110–111
CMI, 175
DSS
64-kbps loop rate, 115
naming conventions, 112
secondary channels, 113
E1, 161
ISDN BRI, 244

2B1Q, 244
4B3T, 245
T1, 132–135
T3, 194
line equalization, 130
line noise, 8
line overhead, STS-1 frames, 510–513
line protection switched topologies, 523
line signaling, R2, 167
link alignment, 48–52
link-by-link signaling, 59
link sets, 34
LMI (Local Management Interface) specification, 289–291
LMU (line-monitoring unit), 142
LNP (local number portability), 69–70
call flow, 71–72
LNPAs (local number portability administrators), 69
load sharing, MTP3, 53
loading coils, 131, 434
local loop technologies
DDS, 106
ISDN, 213–216
local number portability, 69–72
location portability, 72
logic errors, T1, 145–146
loop rates, DDS
secondary channel, 106
64-kbps, 115
loop start signaling, 13
lost frames, 323
low-order address octets (Q.921), 221
LS bearer channels, 427
LSSU (Link Status Signal Unit), 45–47
LTE (line termination equipment), 498–499

M

M12 multiplexing, DS3, 185
M13 multiplexing, 193
M23 framing (T3), 195–197
M23 multiplexing, 189, 192
MAC domains (DOCSIS), 463
downstream signaling, 465
SID, 464
upstream signaling, 466–467
MAC layer functionality, DQDB, 380–382
magnetic coils, 254
management cells, 312
MANs (metropolitan-area networks), 501
MAP messages, 470
marked frames, 343
mated pairs, 27
maxCTD (cell transfer delay), 323–325
MBR (maximum burst rate), 282
MBS (maximum burst size), 329
MCNS (Multimedia Cable Network System Partners Ltd.), 447
MCR (minumum cell rate), 329
message mode (common part), 356
messages
DUP, 55
ISUP, 59–60
Q.931, 230, 233
information elements, 233–240
TUP, 56–58
metasignaling, 307
M-Frame (DS2), bit stuffing), 188
MFS (maximum frame size), 329
microfilters, 419
MID field (SIP Layer 2 PDUs), 395
MIDs (message identification pages), 379
mid-span repeaters, 215–216
MNs (matched nodes), 499
modems, DSL
framing modes, 427–428
trainup, 426
modulation schemes
ADPCM, 95–96
CVSD, 98–99
DM, 97
DOCSIS, 458–459
DPCM, 94–95
MPEG framing, 461
M-plane (B-ISDN model), 317

MSB (most significant bit), cell payload values, 310
MSOs (multiple service operators), 446
MSU (Message Signal Unit), Service Information Field, 50–52
MT (multi-tone) signaling, interregister signaling, 170–173
MTP (Message Transfer Part), 41
    MTP1, 42–43
    MTP2, 43
        FISU, 43–45
        link alignment, 48–52
        LSSU, 45–47
        timers, 49
    MTP3, 37, 52–54
multimode distortion, 502
multimode fiber, 502
multiplexing, 109
    M13, 193
    M23, 189, 192
    SRDM, 108
    STDM, 267
    TDM, 109

N

NADH (North American Digital Hierarchy)
    DS2 frame structure, 187
    DS3
        M13 multiplexing, 193
        M23 multiplexing, 189, 192
    DS4, 206
    evolution of, 181
    signal levels, 184–185
    T3 framing, 195–197
nailed architecture, 109
naming conventions, DSS line coding schemes, 112
NANP (North American Numbering Plan), 59
NAPs (Network Access Points), 63
national SSPs, 23–25
NCI (Network Control Information) field, SIP Layer 2 PDUs, 394
network indicators, 24
network layer functionality, Q.931 specification, 229–233
network layer protocols, interworking with SMDS, 377
network nodes, clusters, 39
NEXT (near-end crosstalk), 9, 435
NFAS (non-facility associated signaling), 155, 257
NIDs (network interface devices), 11, 217, 419
    home run, 254
NNI (Network-to-Network Interface), 304
    header fields, 305
        CLP, 313
        HEC, 313, 316
        PT, 309–313
        VCI, 306–308
        VPI, 306
NO-CRC-4 framing, E1 circuits, 158
noise, 7–9
noise funneling, 455
noise margin, 421–422
non-associated signaling, 31
non-compelled R2 signaling, 167
non-deferred blocking, GR-303-CORE, 17
nonreal-time service classes, 331
    ABR, 337–341
    GFR, 343
    nrt-VBR, conformance definition, 336
    UBR, 341–342
North American Digital Hierarchy (NADH), 125
North American point code format, 39
nrt-VBR (real-time variable bit rate), 336
nrt-VBR service category, 365
NT12s, 218
NTSC (National Television Systems Committee), 451
number portability, 69–70
    call flow, 71–72
    location portability, 72
Nyquist Theorem, 85

O

OAM (Operations, Administrations, and Maintenance) cells, 309–313
network management levels, 312
OAM&P (operations, administration, maintenance and provisioning), 521
obsolesence of SMDS, 409
OC designation, SONET, 505–506
OCU-DP (office channel unit-data port), 108–109
OCx architectures, 486
OOF (out-of-frame), 137
OPC (Originating Point Code), 38
open topology, DQDB, 379
optical networks
DWDM, 521
fiber-optic cabling, 501, 504
hardware, 495
LTE, 498–499
optical routers, 500
PTE, 496
STE, 500
rings, 522
line protection switched topologies, 523
path protection switched topologies, 524
SDH
applications, 522–523
frame structure, 519–520
VCs, 517, 519
SONET, 505
applications, 522–523
STS-1 frame overhead, 508–516
STS-1 frame structure, 506–507
WDM, 521
optical routers, 500
OSI reference model, equivalent SS7 layers, 41
overbooking, 321
overlap signaling, 59
overlapped PAM (pulse amplitude modulation), 430

P

packets, fragmentation, 467
packet-switching technologies, Frame Relay, 266–268
access signaling specifications, 273
congestion avoidance mechanisms, 286–289
contractual SLA values, 282
core access specifications, 272
discards, 284–286
DLCI, 276–282
framework specifications, 271
LMI specification, connection status, 289–291
service description specifications, 271–272
signaling, 286–292
standards bodies, 268–270
VCs, 275–276
PAD field (SIP Layer 3 PDUs), 390
pair gain, 14
PAL (Phase Alternating Line) standard, 451
PAM (pulse amplitude modulation), 85–91
path overhead, STS-1 frames, 513
path protection switched topologies, 524
Payload CRC field (SIP Layer 2 PDUs), 396
Payload Length field (SIP Layer 2 PDUs), 395
PCM (pulse code modulation), 84
AMI line coding, 110–111
PCR (peak cell rate), 328
PDH (Plesiochronous Digital Hierarchy), 125–126, 181
peak-to-peak CDV, 325
PEs (physical entities), 63–64
physical characteristics
E1 circuits, 155–157
T1, 128–131
physical layer (B-ISDN model), 318
PMD sublayer, 318
TC sublayer, 318–319
physical layer specifications, ISDN, 219
plane management (M-plane), 317
PLAR (private line, automatic ringdown), 167

PLCP (physical layer convergence procedure)
functions, SIP Layer 1, 396
- DS1 media, 397–399
- DS3 media, 399–400
- E1 media, 401

Plesiochronous Digital Hierarchy (PDH), 125–126, 181
plesiochronous timing versus hierarchical synchronous timing, 492
PLL (Phase Lock Loop), 112
PMD (physical medium dependent) sublayer functions, SIP Layer 1 318, 396
PNNIs (Private Network-Network Interfaces), 304
point codes, 37–38
- ANSI format, 39
- network indicators, 24
- UTI format, 40–41

point-to-multipoint connections, DDS, 109
polarity states, 108
- BPVs, 112
- capacitance, 111

policing, 344–345. *See also* UPC
portability, 69–72
POTS splitters, 419
power cross, 107
power levels of DSL services, 428
- reducing/boosting, 429
- Trellis coding, 429–430

power ranging, 470–471
prediction mechanism, DPCM, 94
presync state, 315
PRI (Primary Rate Interface), 255–256
- NFAS, 257
- Q.Sig, 260
- T1/E1 configuration, 258–259
- troubleshooting, 259–260

probability density model (CTD), 324
protected fibers, 522
protocol discriminator (Q.931 messages), 230
proving period, link alignment process, 49
PSD (power spectral density), 428
pseudo-LAN conflict (SMDS), troubleshooting, 409
PT (payload type) field, 309–313
PTE (path termination equipment), 496
pulse signaling, R2, 169
pulse streams, 85
pulse stuffing, 134
PVCs (permanent virtual circuits), 275

Q

Q.921 specification, 220–224
- alignment process, 224–227
- control octets, 222–223
- SAPs, 221
- timers, 227–228

Q.931 specification, 229–230
- call flow, 232–233
- messages, 230, 233
  - information elements, 233–240

Q.Sig, 260
QAM (quadrature amplitude modulation), 415–416, 458
- Trellis coding, 429–430

QoS (quality of service)
- CAC, 320
- CER, 325
- CLR, 323
- maxCTD, 323–325
- peak-to-peak CDV, 325
- SECBR, 326

QoS field (SIP Layer 3 PDUs), 391
QPSK (quadrature phase shift keying), 458
quantiles, 325
quantizing process, 87–91
quasi-associated signaling, 30
queue-arbitration algorithm (DQDB), 380–382
quick train, 426

R

R reference points, 219
R2 signaling
    call control, 167
    characteristics, 166–167
    interregister signaling, 170–173
    line signaling, 167
RADSL (rate adaptive DSL), 421, 433
RARP (Reverse ARP), 279
RBE (routing bridge encapsulation), RFC 1483, 438
RBS (robbed-bit signaling), 17, 28, 138
real-time polling, 467
real-time service classes, 331
    CBR, 333–334
    rt-VBR, 334–336
recovering discarded Frame Relay packets, 284–286
red alarms (T1), 146
reduced merged framing (DSL), 428
reduced overhead framing (DSL), 427
reduced separate framing (DSL), 427
reducing
    DSL power levels, 429
    noise during quantizing phase, 88–91
redundancy bytes, 422
reference points, 218–219
    ATM, 304
    CTD, 323
reflections, 455
regenerators, 517
repeaters, mid-span, 215–216
requirements of signaling data link setup, 42
Reserved field (SIP Layer 3 PDUs), 386
resistance, effect on DSL performance, 436
reverse battery signaling, 13
RF (radio frequency) signals, noise funneling, 455
RFC 1209, 374
RFC 1483 bridging, 438
RFI (radio frequency interference), effect on DSL performance, 434
rings (optical), self-healing, 522
    line protection switched topologies, 523
    path protection switched topologies, 524
RJ-48X connectors, 129
routers, optical, 500
rt-VBR (real-time variable bit rate), 334
    conformance definition, 336
    service category, 365
RZ (return-to-zero) encoding, DDS AMI implementation, 110–111

S

S reference point, 218
S/T bus, 218–219
SA (Source Address) field, SIP Layer 3 PDUs, 387–388
sampling, 85–87
SANC (signaling area network code), 40
SAPs (service access points), 221
SAR (segmentation and reassembly), IMPDUs, 382, 385
SAR (segmentation and reassembly) sublayer, 320
SAV (source address verification), 376
SB-ADPCM (Sub-Band ADPCM), 99
SCEF (Service Creation Environment Function), 63
SCF (Service Control Function), 63
SCPs (Service Control Points), 27
SCR (sustained cell rate), 328
screening SMDS addresses, 390
SD (selective discard), 342
SDF (Service Data Function), 63
SDH (Synchronous Digital Hierarchy)
    applications, 522–523
    frame structure, 519–520
    OAM&P, 521
    VCs, 517–519
    versus SONET, 504
SDP (Service Data Point), 64
SDSL, 433
SECAM (Systeme Electronique Couleur Avec Memoire), 451
SECBR (severely errored cell block ratio), 326
secondary channels , DDS, 106, 113

security
  cloned devices, 480
  DOCSIS, 480–482
Segment Type field (SIP Layer 2 PDUs), 394
Segmentation Unit field (SIP Layer 2 PDUs), 395
self-healing rings, 523
  line protection switched topologies, 523
  path protection switched topologies, 524
semi-compelled R2 signaling, 167
Sequence Number field (SIP Layer 2 PDUs), 395
serial-link layer protocols, DXI, 403–405
service classes (ATM), 331
  nonreal-time
    ABR, 337–341
    GFR, 343
    nrt-VBR, 336
    UBR, 341–342
  real-time
    CBR, 333–334
    rt-VBR, 334–336
service description specifications, Frame Relay, 271–272
service providers
  portability, 69–70
    local number portability call flow, 71–72
    location portability, 72
  wholesale, DSL, 441
services
  CS-2, 66–68
  Type A, 65
  Type B, 65
SEUM (signal unit error rate monitor), 43
SF (D4 Superframe), 136
SFID (service flow identifier), 464
shades of gray analogy, analog communication, 81
SIB (Status Indicator Busy), 46–47
SIE (Status Indicator Emergency), 46–48
SIF (Service Information Field), MSU, 50–52
SIG (SMDS Interest Group), 374
signal levels of NADH, 184–185
signal unit identification (FISU), 44
signaling, 22
  analog, 3
    amplitude, 4
    attenuation, 6–9
    DACs, 10
    distortion, 5–6
    frequency, 5
    GR-303-CORE standard, 17–18
    IDLCs, 14, 16
    noise, 7, 9
    tip and ring circuits, 11–14
    versus digital, 3
    wavelength, 5
  associated, 29
  CAS, 28
  CCS, 29
  conversion process, 83–84
  Frame Relay, 286–289
    call control, 292
    connection status, 289–292
  non-associated signaling, 31
  polarity states, 108
  quasi-associated signaling, 30
  SS7
    DUP, 54–55
    ISUP, 58–59
    link sets, 34
    links, 36
    MTP, 41–54
    need for, 32–33
    point codes, 37–41
    SCPs, 27
    signaling links, 33–34
    SSPs, 23, 25
    STPs, 27
    TUP, 56–58
  traffic management, 53
signaling links, 33–34
Simple Unsuccessful Backward Setup Information Message (TUP), 58
SIN (Status Indicator Normal), 46–48
single leaky bucket method, congestion control, 334
single mode fiber, 502
single-ended services, 65

single-latency mode, DSL data transmission, 427
sinusoidal waves, 82
SIO (Status Indicator Out of Alignment), 47
SIOS (Status Indicator Out of Service), 46
SIP (SMDS Interface Protocol), 375–377
Layer 1
PLCP functions, 396–401
PDU fields, 396
Layer 2 PDU fields, 393–396
Layer 3 PDU fields, 386–392
SIP (SMDS Interface Protocol), 375
SIPO (Status Indicator Processor Outage), 46–47
SLAs (service-level agreements), contractual values, 282
SLCs (Signaling Link Code), 15, 34
slots, DQDB, 380
IMPDUs, SAR, 382, 385
SLS (Signaling Link Selection), 34
SLT (Signaling Link Terminal), 43
SMAF (Service Management Access Function), 63
smart-jacks, 128
SMC (SONET Minimum Clock), 493
SMDS (Switched Multimegabit Data Service), 355–356
access-classes, 376
access speeds, 376
addressing, 376
screening, 390
subscription, 388–390
alignment with 802.6, 373, 385
architecture, 374
Bellcore's role in development, 373–374
configuring, 405–406
DQDB, 377
DXI protocol, 375
troubleshooting, 407
features, 372
history, 372–374
Layer 3 protocol interworking, 377
obsolesence of, 409
psuedo-LAN conflict, troubleshooting, 409
SIP, 375–377
Layer 1 PDU, PLCP functions, 396–401
Layer 2 PDU fields, 393–396
Layer 3 PDU fields, 386–392
SNI, troubleshooting, 408
source address validation, 390
time-to-market delay, 385
troubleshooting, 407
SMF (Service Management Function), 63
SN (Service Node), 64
SNI (Subscriber Network Interface), troubleshooting, 408
SONET (Synchronous Optical Network), 505
applications, 522–523
fiber-optical cabling, 501, 504
OAM&P, 521
OC designation, 505–506
STS-1 frame overhead, 508–516
STS-1 frame structure, 506–507
versus SDH, 504
source address validation, SMDS, 390
source devices (ATM), traffic parameters, 327
BT, 329
CDVT, 329, 331
MBS, 329
MCR, 329
MFS, 329
PCR, 328
SCR, 328
source-traffic descriptors, 327
sparse combining, 482
specifications
Frame Relay
access signaling, 273
core access, 272
framework, 271
service description, 271–272
ISDN
physical layer, 219
Q.921, 220–228
Q.931, 229–233
SPIDs (service profile identifiers), 40, 248
split plan, 452
SPVCs (soft PVCs), 275
SRDM (Subrate Data Multiplexer), 108

SRF (Special Resource Function), 63
SS7 (Signaling System 7). *See also* INs
  DUP, 54–55
  equivalent OSI layers, 41
  ISUP, 58–59
  links, 36
  MTP, 41
    MTP1, 42–43
    MTP2, 43–52
    MTP3, 52–54
  need for, 32–33
  point codes, 37–38
    ANSI format, 39
    UT format, 40–41
  SCPs, 27
  signaling links, 33–34
  SSPs, 23–25
  STPs, 27
  TUP, 56–58
SSCF (service specific coordination function), 320
SSCS (service specific convergence sublayer), 320
SSF/CCF (Service Switching Function/Call Control Function), 62
SSPs (Service Switching Points), 23–25
  national SSPs, Class 5, 25
  SLS, 34
SSU (Synchronization Supply Unit), 494–495
standard train, 426
standards bodies
  Frame Relay, FRF IAs, 273–274
  governing ATM development, 301–302
STDM (statistical TDM), 267
STE, 500
steaming mode (common part), 356
STM (Synchronous Transport Module) versus ATM, 303
STPs (Signal Transfer Points), 27
Stratum clocks, 493
STS (Synchronous Transport Signal), 505
STS-1 frame structure
  overhead, 508–516
  SONET, 506–507
sub-band codecs, 99
sublayers of AAL, 319–320
subrate offerings, E1, 157
Subsequent Address Message (TUP), 57
super headends, 451
SVCs (switched virtual circuits), 275
SW56, 115
  Cisco router configuration, 118–119
  DTMF, 116
sync byte, 427
sync state, 315
synchronous technologies versus asynchronous, 105
synchronous timing
  BITS, 494–495
  Stratum clocks, 493
SYNTRAN (synchronous transmission), 182, 505

T

T reference point, 218
T1 circuits
  alarm conditions, troubleshooting, 146–147
  controller configuration, 147
  ESF frame format, D5 channel banks, 140–142
  framing formats, 135
    D1 channel banks, 137–139
    D1D channel banks, 140
    D3 channel banks, 140
    D4 channel banks, 140
    D5 channel banks, 140–142
  hairpin connections, configuring, 148
  line-coding schemes, 132–135
  physical characteristics, 128–131
  PRI configuration, 258–259
  troubleshooting, 142
    BERT, 143–144
    format errors, 145–146
    logic errors, 145–146

T3 circuits, 180
- channelized, 183, 198–199
- deployment options, 193
- framing, 195–197
- line coding, 194
- synchronous circuits, 182
- unchannelized, 183

T3Plus, development of HSSI, 402
tandem switches, 25
Tc (committed rate measurement interval), 282
TC sublayer, 318–319
TDM (time-division multiplexing), 14, 22, 109, 126
TE1 (terminal equipment type 1) devices, 219
TE2 (terminal equipment type 2) devices, 219
TEI checks, 225
TEK (traffic encryption key), 480
Telecommunication Act of 1996, 68
television, international standards, 451
test patterns, DDS, 117–118
tilt effect of signal strength, 448
timers
- for link alignment, 49
- Q.921, 227–228

timing sources of DDS, 112
tip and ring circuits, 11–14
TMs (terminal multiplexers), 496
toll free number lookup, 67
tone-on-idle signaling, 168
topologies, DQDB, 379
traffic, congestion control
- closed-loop flow control, 338
- continuous-state leaky bucket model, 347
- dual leaky bucket model, 336
- EPRCA, 339
- single leaky bucket model, 334

traffic management, 344
- ABR service category, 366
- CBR service category, 365
- CER, 325
- CLR, 323
- GCRA, 345–346
- maxCTD, 323, 325
- nrt-VBR service category, 365
- PCR algorithm, 349
- peak-to-peak CDV, 325
- policing, 345
- rt-VBR service category, 365
- SCR algorithm, 350
- SECBR, 326
- UBR service category, 365

traffic parameters (ATM), 327. *See also* service classes
- BT, 329
- CDVT, 329–331
- MBS, 329
- MCR, 329
- MFS, 329
- PCR, 328
- SCR, 328

traffic shaping, 330
trainup, 426
transmission frame adaptation, 319
transmission rates, DDS, 106
transmit filter, DDS, 106
transport mechanisms, DSL, 439–440
Trellis coding, 429–430
tributaries, 516
troubleshooting
- ISDN BRI, 252–254
- ISDN PRI, 259–260
- SMDS, 407
  - DXI, 407
  - pseudo-LAN conflict, 409
  - SNI, 408
- T1, 142
  - alarm conditions, 146–147
  - BERT, 143–144
  - format errors, 145–146
  - logic errors, 145–146

trunk cable, 447
trunks, IMTs, 27
TUP (Telephone User Part) messages, 56–58
Type A frames (Q.921), 221
Type A services, 65
Type B frames (Q.921), 221
Type B services, 65

U

U interface (ISDN), 218
UBR (unspecified bit rate), 341–342, 365
UCDs (upstream channel descriptors), 469
UGS (Unsolicited Grant Service), 466
unassigned cells, 319
unchannelized T3, 182–184
UNI (User-Network Interface), 304
  header fields, 305
    CLP, 313
    GFC, 306
    HEC, 313, 316
    PT, 309–313
    VCI, 306–308
    VPI, 306
  PNNIs, 304
  related IAs, 303
UNI 3.x/4.0 specification, 304
unipolar signals versus bipolar, 108
unmarked frames, 343
unprotected fiber, 522
UPC (usage parameter control)
  continuous-state leaky bucket algorithm, 347
  GCRA (generic cell rate algorithm), 345–346
  PCR algorithm, 349
  SCR algorithm, 350
U-plane (B-ISDN model), 317
upper-layer protocols (SS7)
  DUP, 54–55
  ISUP, 58–59
  TUP, 56–58
UPSR (unidirectional path switched ring), 524
upstream admission control, 479
upstream noise, 455–456

V

VCCs (virtual channel connections), 306
VCI (virtual channel identifier) field, 306–308
  effect on PT, 312–313
VCs (virtual circuits), 275–276
VCs (virtual containers), 517–519
VDSL (very high-rate DSL), 433
virtual channel switching, 308
virtual path switching, 308
virtual scheduling algorithm, 346
voltage
  1s density, 112
  capacitance, 111
VPCs (virtual path connections), 306
VPI (virtual path identifier) field, 306
  effect on PT, 312–313

W

wavelength, 5
WDM (wavelength-division multiplexing), 127, 521
wholesale services, DSL, 441
wink mode (E&M signaling), 13

X-Y-Z

XOR (exclusive OR) function, 92–93

yellow alarms, T1, 146

zero substitution, DSS line coding schemes, 113
zone ID (ITU point codes), 40