in many fields. As an analogy, DSP can be compared to a previous technological revolution: *electronics*. While still the realm of electrical engineering, nearly every scientist and engineer has some background in basic circuit design. Without it, they would be lost in the technological world. DSP has the same future.

This recent history is more than a curiosity; it has a tremendous impact on *your* ability to learn and use DSP. Suppose you encounter a DSP problem, and turn to textbooks or other publications to find a solution. What you will typically find is page after page of equations, obscure mathematical symbols, and unfamiliar terminology. It's a nightmare! Much of the DSP literature is baffling even to those experienced in the field. It's not that there is anything wrong with this material, it is just intended for a very specialized audience. State-of-the-art researchers need this kind of detailed mathematics to understand the theoretical implications of the work.

A basic premise of this book is that most practical DSP techniques can be learned and used without the traditional barriers of detailed mathematics and theory. *The Scientist and Engineer's Guide to Digital Signal Processing* is written for those who want to use DSP as a *tool*, not a new *career*.

The remainder of this chapter illustrates areas where DSP has produced revolutionary changes. As you go through each application, notice that DSP is very *interdisciplinary*, relying on the technical work in many adjacent fields. As Fig. 1-2 suggests, the borders between DSP and other technical disciplines are not sharp and well defined, but rather fuzzy and overlapping. If you want to specialize in DSP, these are the allied areas you will also need to study.

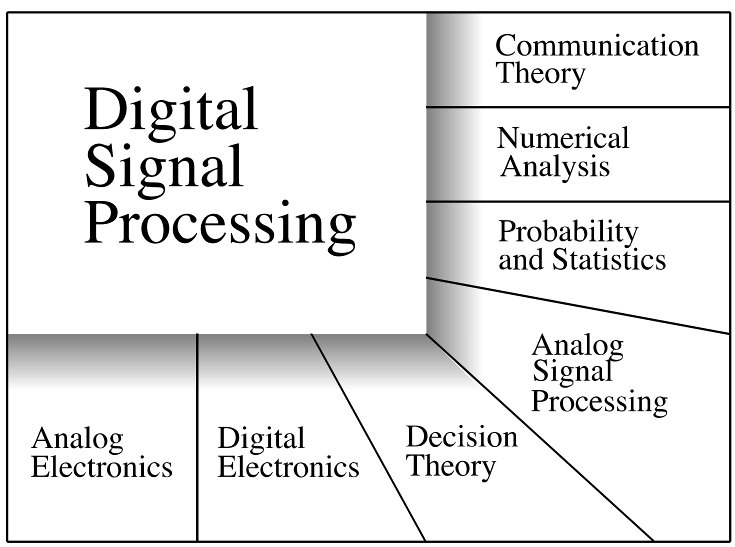

FIGURE 1-2
Digital Signal Processing has fuzzy and overlapping borders with many other areas of science, engineering and mathematics.

Telecommunications

Telecommunications is about transferring information from one location to another. This includes many forms of information: telephone conversations, television signals, computer files, and other types of data. To transfer the information, you need a *channel* between the two locations. This may be a wire pair, radio signal, optical fiber, etc. Telecommunications companies receive *payment* for transferring their customer's information, while they must *pay* to establish and maintain the channel. The financial bottom line is simple: the more information they can pass through a single channel, the more money they make. DSP has revolutionized the telecommunications industry in many areas: signaling tone generation and detection, frequency band shifting, filtering to remove power line hum, etc. Three specific examples from the telephone network will be discussed here: multiplexing, compression, and echo control.

Multiplexing

There are approximately *one billion* telephones in the world. At the press of a few buttons, switching networks allow any one of these to be connected to any other in only a few seconds. The immensity of this task is mind boggling! Until the 1960s, a connection between two telephones required passing the analog voice signals through mechanical switches and amplifiers. One connection required one pair of wires. In comparison, DSP converts audio signals into a stream of serial digital data. Since bits can be easily intertwined and later separated, many telephone conversations can be transmitted on a single channel. For example, a telephone standard known as the *T-carrier system* can simultaneously transmit 24 voice signals. Each voice signal is sampled 8000 times per second using an 8 bit companded (logarithmic compressed) analog-to-digital conversion. This results in each voice signal being represented as 64,000 bits/sec, and all 24 channels being contained in 1.544 megabits/sec. This signal can be transmitted about 6000 feet using ordinary telephone lines of 22 gauge copper wire, a typical interconnection distance. The financial advantage of digital transmission is enormous. Wire and analog switches are expensive; digital logic gates are cheap.

Compression

When a voice signal is digitized at 8000 samples/sec, most of the digital information is *redundant*. That is, the information carried by any one sample is largely duplicated by the neighboring samples. Dozens of DSP algorithms have been developed to convert digitized voice signals into data streams that require fewer bits/sec. These are called **data compression** algorithms. Matching **uncompression** algorithms are used to restore the signal to its original form. These algorithms vary in the amount of compression achieved and the resulting sound quality. In general, reducing the data rate from 64 kilobits/sec to 32 kilobits/sec results in no loss of sound quality. When compressed to a data rate of 8 kilobits/sec, the sound is noticeably affected, but still usable for long distance telephone networks. The highest achievable compression is about 2 kilobits/sec, resulting in