Chapter 5

DSP Comes of Age: The 1970s

What was Happening in the 1970s

movies people were watching:
Woody Allen's *Annie Hall*
The Godfather with Marlon Brando
Saturday Night Fever

TV shows people were watching:
"All in the Family"
"M*A*S*H"
"Saturday Night Live"

music people were listening to:
Bruce Springsteen
disco music
punk rock

books people were reading:
Solzhenitsyn's *Gulag Archipelago*
Richard Adams's *Watership Down*

Some historians have seen in the 1970s the beginning of the Third Industrial Revolution. The First Industrial Revolution, characterized by steam power and factory production, began in the late 18th century. The Second Industrial Revolution, characterized by electric power, the internal combustion engine, and telegraph and telephone communications, began in the late 19th century. The defining technology of the Third Industrial Revolution is the computer, particularly the microprocessor.[1] Besides in computer technology itself, there were revolutionary changes in communications, instrumentation, and control systems, and all of these areas became much more important economically.

In the exploitation of the technologies of the First and Second Industrial Revolutions, the Soviet bloc had been able to compete with the West (though users of Soviet cars and telephones might have their doubts). With the new computer and communications technologies, however, the Communist states fell further and further behind.[2] Indeed, failure to participate fully in the Third Industrial Revolution was a contributing cause of Communist defeat in the Cold War.[3]

---

[1]J.A.S. Grenville, *A History of the World in the Twentieth Century* (Cambridge, MA: Harvard University Press, 1994), p. 927; and Walter B. Wriston, "Bits, bytes, and diplomacy" (*Foreign Affairs*, vol. 76 (1997), no. 5, pp. 172–182).

[2]Grenville *op. cit.*, p. 927.

[3]The historian Richard Bessel has written, "Indeed, one reason for the destabilization of eastern European socialist regimes during the 1980s was that their populations had come to expect the fruits of a consumer society which the state-socialist regimes were unable to deliver." [Richard Bessel, "European society in the Twentieth Century." In T.C.W. Blanning, ed., *The Oxford Illustrated History of Modern Europe* (Oxford: Oxford University Press, 1996), pp. 231–254; quotation on p. 241.]

Many people would remember the 1970s for *détente* (relaxation of tensions), highlighted by trips by President Nixon to China and the Soviet Union in 1972 and the Strategic Arms Limitation Treaty the same year. Even more memorable were the 1972 Watergate break-in (leading to Nixon's resignation in 1974); the 1973 oil embargo by the Arab oil-producing states (causing an energy crisis and contributing to the worldwide inflation of the mid 1970s); the end of the Vietnam war in 1975; the overthrow of the Shah of Iran in 1979; and the Soviet invasion of Afghanistan the same year. Prominent were environmental concerns (the Club of Rome published *The Limits of Growth* in 1972) and the women's movement (*Ms* magazine first appeared in 1972).

Engineers may remember new aircraft (such as the Boeing 747, the Airbus, and the Concorde), the first pocket electronic-calculators,[4] the blackout of New York City in 1977, and the beginning of bar codes and electronic scanners in supermarkets. In 1975 came the first personal computer, the Altair 8800, sold in kit form, and an assembled computer, the Apple II, was offered in 1977. In 1975 Sony and JVC began marketing video cassette recorders (JVC in VHS format, Sony in Betamax), and at about the same time Citizens Band radio burst into popularity.[5] The 1970s was the first decade of video games (Pong appeared in 1972), of word processing (on typewriters with the capability of storing text), and of ATMs (automatic teller machines). It was also a time when many people thought the world was changing too fast, as suggested by the popularity of Alvin Toffler's *Future Shock* (1970), which discussed the "bombardment of the senses" and "information overload".[6]

In the 1970s consumers began to be aware of digital signal processing. In Japan, the recordings of Nippon Columbia began to be digitally mastered in 1972.[7] The same year in Britain, the BBC began using PCM for high-quality

---

[4]It was the HP-35, the first with exponentiation and trig-function keys, that sold the engineering community on pocket calculators; Hewlett-Packard sold more than 300,000 of these in the three years following its marketing launch in 1972 [*IEEE Spectrum*, vol. 25 (1988), no. 11, p. 128]. When calculators with such capabilities became inexpensive, the slide rule became history.

[5]Citizens Band radio started in the United States in 1958, but first came to wide public attention with the gasoline shortage and truckers' strike of 1974. Suddenly millions of Americans wanted to have CB radio in their homes, cars, or boats. In 1976 approximately 11 million units were sold. [*The U.S. Consumer Electronics Industry in Review: 94 Edition* (Washington, DC: Electronic Industries Association, 1994), p. 20.]

[6]Alvin Toffler, *Future Shock* (New York: Random House, 1970).

[7]Peter J. Bloom, "High-quality digital audio in the entertainment industry: an overview of achievements and challenges" (*IEEE ASSP Magazine*, vol. 2 (1985), no. 4, pp. 2–25).

FIGURE 1. The Speak & Spell toy was introduced by Texas Instruments in 1978. (Texas Instruments photo reproduced by permission.)

sound distribution for radio and television;[8] and in its studios it began using an 8-track digital audio recorder with error correction.[9] In 1975 Tom Stockham showed how DSP could improve historical recordings of Enrico Caruso,[10] and digitally restored recordings began to appear the following year.[11] In 1978 Texas Instruments introduced a toy called Speak & Spell.[12] (See Figure 1.) It taught a child to spell by pronouncing a word and indicating whether an attempted spelling was correct. Two things made the toy practical: an efficient

---

[8]Bloom *op. cit.*

[9]Guy M. McNally, "Digital audio in broadcasting" (*IEEE ASSP Magazine*, vol. 2 (1985), no. 4, pp. 26–44). Professional audio equipment went digital before the consumer products: in 1975 real-time digital reverberation systems became available, and in 1977 several professional digital audio recorders were being sold [Bloom *op. cit.*].

[10]Thomas G. Stockham, Thomas M. Cannon, and Robert B. Ingebretsen, "Blind deconvolution through digital signal processing" (*Proceedings of the IEEE*, vol. 63 (1975), pp. 678–692).

[11]Bloom *op. cit.*

[12]'Speak & Spell' is a trademark of Texas Instruments Incorporated.

Maurice Bellanger: The multirate filtering idea came from our work on delta modulation. Since our PCM channel bank had a per-channel digital filter, we had to minimize the number of multiplications. The multiplier was the most area-consuming operation. A second reason was that, in order to perform filtering in that context, we had to start from a high sampling rate. The final sampling rate is the regular PCM rate, 8 kilohertz, and we had to start from at least 32 kilohertz. So we tried to optimize the conversion from 32 kilohertz to 8 kilohertz using digital filters, and that's where the concept of multirate came in. We noticed that two half-band filters drastically reduced the number of multipliers, and thought that the concept could be applied in different fields. We gave some presentations of the concept within Philips, and it was used in other areas, not only for sample-rate reduction but also for sample-rate increase, which was called decimation and interpolation.

Interviewer: Why did you have the higher initial sampling rate?

Bellanger: Because what we wanted to make was a telephone band filter from 0 hertz to 3400 hertz. But, as you know, a digital filter is a sampled system, so it has a frequency periodicity. And after the base band, which is what we really want to keep with digital filters, we have image bands around all the multiples of the sampling frequency, which have to be canceled by analog filters in any case, since the output has to be analog. But the higher these images, the easier it is to suppress them. Our goal was to be able to get rid of those image bands with just resistors and capacitors—no coil, no severe filtering. Doing that required at least 32 kilohertz of over-sampling. Sixty-four would probably have been better but was too expensive, so we chose the 32.[1]

Interviewer: Has computer hardware development influenced the direction of the FFT algorithms?

James Cooley: Definitely. It gave some incentive for designing these nice scheduling algorithms for breaking the FFT into blocks and scheduling the blocks of calculation. Whole papers have been written just on the scheduling of data through hierarchical storage.[2]

[1] Maurice Bellanger oral-history interview 22 April 1997, p. 5.

[2] James Cooley oral-history interview 11 March 1997, p. 10.

algorithm for speech synthesis and inexpensive integrated circuits to carry out the algorithm. It contained a speech-synthesis chip, a read-only memory chip, tone generators, noise generators, and variable electrical filters. Speech was recreated, not synthesized de novo; the 165 words the toy could pronounce had been spoken by a human and then encoded by LPC (linear predictive

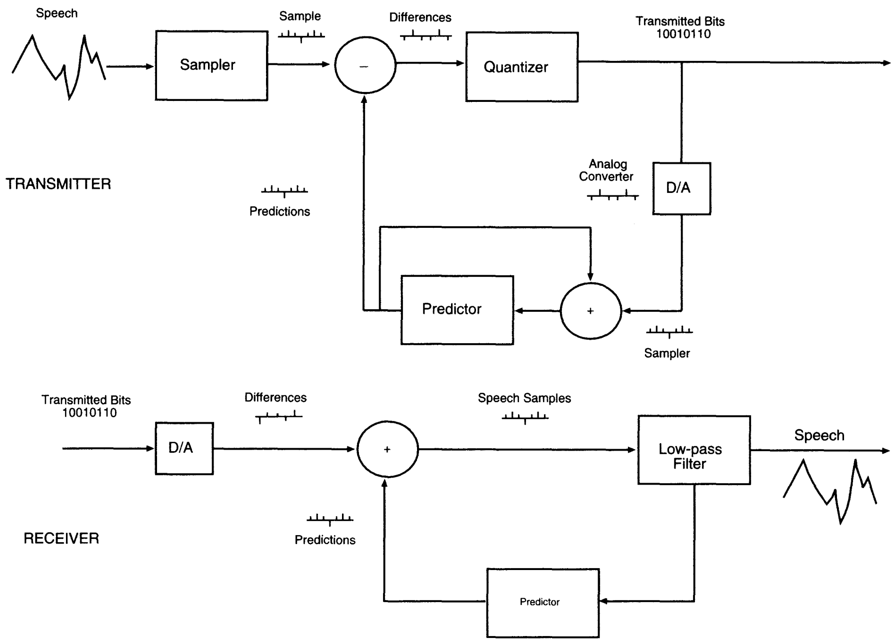

FIGURE 2. The idea of ADPCM (adaptive differential pulse code modulation) is to place identical predictors at transmitter and receiver, so that all that needs to be sent, in order to recreate the original speech, are the differences between the predicted speech samples and the actual samples. (Redrawn after figure on p. 247 of Robert W. Lucky, *Silicon Dreams: Information, Man, and Machine* (New York: St. Martin's Press, 1989).)

coding) for efficient storage.[13] (Speech synthesis de novo was performed by Kurzweil Reading Machine, a much more sophisticated device built in 1976 so that blind people could read ordinary text.)

Among the many advances in speech processing during this decade should be mentioned adaptive differential pulse code modulation (ADPCM), suggested by James Flanagan in 1973.[14] The idea was to exploit the fact that a small sample of speech can be predicted fairly accurately from preceding samples: place identical predictors at the transmitter and the receiver, and send only the difference between the actual sample and the prediction. (See Figure 2.) In 1984 ADPCM was adopted as a new standard for encoding speech; it could, in 32 kbps, achieve the same quality of

[13]Robert W. Lucky, *Silicon Dreams: Information, Man, and Machine* (New York: St. Martin's Press, 1989), pp. 202, 256.

[14]Sidney Millman, ed., *A History of Engineering and Science in the Bell System: Communication Sciences (1925–1980)* (AT&T Bell Telephone Laboratories, 1984), pp. 115–117.

Alfred Fettweis: How can we do something in the digital domain, which is the equivalent of these resonant transfer filters? I tried first to do it on the basis of voltages and currents, and then realized that this is not feasible. Only with mediocre properties was it feasible. So I had to do something different. I therefore tried to directly convert the equations of resonant transfer filters, thus to transpose them into equations that would be realizable by digital algorithms involving multiplications, additions, and delays, that is, the kind of operations commonly available in digital signal processing. And that's how I discovered these wave digital filters. I first obtained them indeed by means of the resonant transfer. I never published how I did that. It was very much more complicated than the approach I did publish. I did not even mention in the papers—the first papers certainly not—that I had obtained the results from resonant transfer. But, in fact, I did. So the resonant transfer work was fundamental for getting to this signal processing method.[1]

Alfred Fettweis: ... Digital signal processing, as opposed to analog signal processing, requires sequential ordering. You must carry out operations in sequence. You must be able to first carry out one operation, and the next one, and so on. This is very different in an analog circuit, where the Kirchhoff equations are all continuously satisfied, so to speak. There is no sequential ordering involved. In digital signal processing, however, you have an algorithm, by definition computable, so you must be able to assign a sequential order in which the operations have to be carried out. I realized this only later, I must confess, but that is really basic for these wave digital ideas. You have on the one hand passivity—that is, you can carry over the passivity of analog circuits. On the other hand, you have sequential ordering due to the fact that we base the analogy, not on voltages and currents, but on waves. And therefore you get sequential operations and satisfy the computability requirement. These are really the two important aspects: on the one hand computability, and on the other passivity and losslessness, which then give you the excellent sensitivity and robustness properties.

... I was quite confident that I could get access to such a Lyapunov function and therefore could also solve the stability and, more generally, robustness problem. I had to give the filters a name. They're digital filters, but they were based on the wave concept. That's why I called them wave digital filters.[2]

[1] Alfred Fettweis oral-history interview 24 April 1997, p. 41.

[2] Alfred Fettweis oral-history interview 24 April 1997, p. 45.

speech as the earlier standard could in 64 kbps.[15] In 1976 Ronald Crochiere, Susan A. Webber, and Flanagan showed that one could achieve moderate savings in digital rate through subband coding (SBC), which divided the signal spectrum into bands and adaptively quantized each independently.[16] In 1978 Larry Rabiner and Ron Schafer published the highly influential *Digital Processing of Speech Signals*, which showed how DSP pervaded the field of speech processing.[17]

The use of satellites for telephony increased the need for reducing echo. The older technique, echo suppression, was replaced by echo cancellation, a technique that, instead of attenuating the echo, synthesizes it and subtracts it from the total signal. (See Figure 3.) Proposed by M. Mohan Sondhi in 1967, the process became commercially feasible with large-scale integration technologies of the late 1970s. The first single-chip echo-canceler was designed by Donald L. Duttweiler in 1979.[18]

Speech processing researchers continued to innovate in digital-filter design, but the field was enlarged also by the work of researchers concerned with other types of signals. In 1969 Alfred Fettweis introduced a new type of digital filter, called the wave digital filter.[19] A seminal advance was the algorithm created by James McClellan and Thomas Parks for the design of equiripple finite-impulse response (FIR) filters; Larry Rabiner further developed the Parks-McClellan algorithm and applied it to a number of different types of filter.[20] Sidney Burrus and Thomas Parks showed how filters could

---

[15]Lucky *op. cit.*, p. 245.

[16]Millman *op. cit.*, p. 117.

[17]Lawrence R. Rabiner and Ronald W. Schafer, *Digital Processing of Speech Signals* (Englewood Cliffs, NJ: Prentice-Hall, 1978).

[18]Charles W.K. Gritton and David W. Lin, "Echo cancellation algorithms" (*IEEE ASSP Magazine*, vol. 1 (1984), no. 2, pp. 30–38), and Shoji Makino, "Acoustic echo cancellation" (*IEEE Signal Processing Magazine*, vol. 14 (1997), no. 5, pp. 39–41).

[19]The first full-length paper on the wave digital filter was Alfred Fettweis, "Digital filters related to classical filter networks" (*Archiv für elektrische Übertragung*, vol. 25 (1971), pp. 79–89), which was reprinted in Digital Signal Processing Committee, eds., *Selected Papers in Digital Signal Processing II* (New York: IEEE Press, 1976), pp. 510–524. A tutorial paper on the subject is Alfred Fettweis, "Wave digital filters: theory and practice" (*Proceedings of the IEEE*, vol. 74 (1986), pp. 270–327).

[20]Thomas W. Parks and James H. McClellan, "Chebyshev approximation for nonrecursive digital filters with linear phase" (*IEEE Transactions on Circuit Theory*, vol. 19 (1972), pp. 189–194); James H. McClellan, Thomas W. Parks, and Lawrence R. Rabiner, "A computer program for designing optimum FIR linear phase digital filters" (*IEEE Transactions on Audio and Electroacoustics*, vol. 21 (1973), pp. 506–526); and Millman *op. cit.*, p. 112. A digital filter whose output depends only upon previous input values is called a finite impulse response filter; in an infinite impulse response filter, the output depends upon previous output values as well as input values.

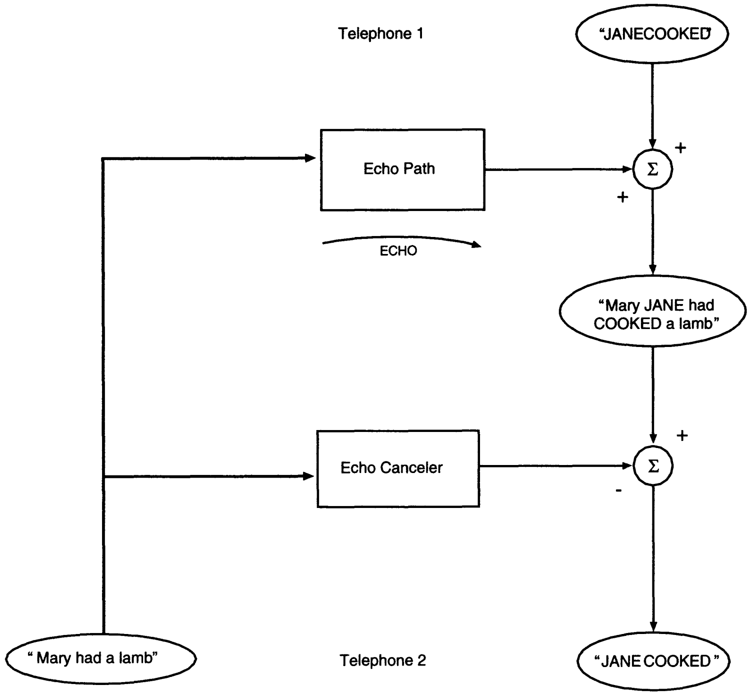

FIGURE 3. The echo canceler, placed at the near end of the line, does not block the echo, nor attenuate it. Instead, it synthesizes the echo and subtracts it, making the resultant signal free from echo. (Redrawn after figure on p. 31 of Charles W.K. Gritton and David W. Lin, "Echo cancellation algorithms" (*IEEE ASSP Magazine*, vol. 1 (1984), no. 2, pp. 30–38).)

be designed according to prescriptions in the time domain.[21] In 1976 Alain Croisier, Daniel Esteban, and Claude Galand introduced quadrature mirror filters (QMFs).[22] Sidney Darlington and Maurice Bellanger pioneered multirate filters.[23] Bede Liu, Toyohisa Kaneko, Alan Oppenheim, Hans Wilhelm Schuessler, Clifford Weinstein, and others made important contribu-

[21]C. Sidney Burrus and Thomas W. Parks, "Time domain design of recursive digital filters" (*IEEE Transactions on Audio and Electroacoustics*, vol. 18 (1970), pp. 137–141).

[22]Perinkolam P. Vaidyanathan, "Quadrature mirror filter banks, M-band extensions and perfect-reconstruction techniques" (*IEEE ASSP Magazine*, vol. 4 (1987), no. 3, pp. 4–20).

[23]Sidney Darlington, "On digital single-sideband modulators" (*IEEE Transactions on Circuit Theory*, vol. 17 (1970), pp. 409–414), and Maurice Bellanger, "Computation rate and storage estimation in multirate digital filtering with half-band filters" (*IEEE Transactions on Acoustics, Speech, and Signal Processing*, vol. 25 (1977), pp. 344–346).

tions in analyzing the accuracy of digital filters.[24] Thomas Huang pioneered in developing filters for image processing.[25] State-space methods and related mathematical techniques were developed, which were later introduced into fields such as filter design, array processing, image processing, and adaptive filtering.[26] FFT theory was extended to finite fields and used in areas such as coding theory.[27]

Work on digital filters and advanced DSP technologies was stimulated and disseminated by the second, third, and fourth Arden House Workshops, held in 1970, 1972, and 1974 respectively. These workshops attracted researchers from around the world, and the proceedings made up three special issues of the transactions of the IEEE Group on Audio and Electroacoustics.[28]

---

[24]See, for example, Bede Liu, "Effect of finite word length on the accuracy of digital filters—a review" (*IEEE Transactions on Circuit Theory*, vol. 18 (1971), pp. 670–677); Bede Liu and Toyohisa Kaneko, "Error analysis of digital filters realized with floating-point arithmetic" (*Proceedings of the IEEE*, vol. 57 (1969), pp. 1735–1747); Alan V. Oppenheim and Clifford J. Weinstein, "Effects of finite register length in digital filtering and the fast Fourier transform" (*Proceedings of the IEEE*, vol. 60 (1972), pp. 957–976); and Hans Wilhelm Schuessler, "On the approximation problem in the design of digital filters" (*Proceedings of the Fifth Annual Princeton Conference on Information Sciences and Systems 1971*, pp. 54–63). Other important papers in this area were Paul M. Ebert, James E. Mazo, and Michael G. Taylor, "Overflow oscillations in digital filters" (*Bell System Technical Journal*, vol. 48 (1969), pp. 2999–3020) and Leland B. Jackson, "Roundoff-noise analysis for fixed-point digital filters realized in cascade or parallel form" (*IEEE Transactions on Audio and Electroacoustics*, vol. 18 (1970), pp. 107–122).

[25]Thomas S. Huang, "Stability of two-dimensional recursive filters" (*IEEE Transactions on Audio and Electroacoustics*, vol. 20 (1972), pp. 158–163); Thomas S. Huang, "Two-dimensional windows" (*IEEE Transactions on Audio and Electroacoustics*, vol. 20 (1972)); and Thomas S. Huang, James Burnett, and Andrew Deczky, "The importance of phase in image processing filters" (*IEEE Transactions on Acoustics, Speech, and Signal Processing*, vol. 23 (1975), pp. 529–542).

[26]See, for example, Alfred Bruckstein and Thomas Kailath, "An inverse scattering framework for several problems in signal processing" (*IEEE ASSP Magazine*, vol. 4 (1987), no. 1, pp. 6–20); Patrick Dewilde, "Advanced digital filters" (Thomas Kailath, ed., *Modern Signal Processing* (Washington, DC: Hemisphere Publishing, 1985), pp. 169–209); and Richard A. Roberts and Clifford T. Mullis, *Digital Signal Processing* (Reading, MA: Addison-Wesley, 1987).

[27]See, for example, Richard E. Blahut, "Algebraic fields, signal processing, and error control" (*Proceedings of the IEEE*, vol. 73 (1985), pp. 874–893).

[28]Lawrence R. Rabiner, "The Acoustics, Speech, and Signal Processing Society—A Historical Perspective" (*IEEE ASSP Magazine*, January 1984, pp. 4–10). The special issues of the transactions appeared in June 1970, October 1972, and June 1975.

Thomas Huang: At Purdue [in the mid 1970s] we looked into some of the nonlinear filters, especially the so-called median filters. The median filter is for reducing noise in the image. The conventional way is to replace each point by the local average in the neighborhood to smooth out the noise. But it's not very effective when the noise is a spike type—salt and pepper—which is very common in digital transmission. It turns out that the median filter is much better. For a given point, you take the neighborhood around it, and instead of replacing the middle point by the mean, you replace it with the median gray level. It takes out spikes very easily and also has the nice property of keeping edges sharp. If you use the mean, the edges smear. We looked into this and found a very efficient algorithm. It became very popular; many of the software packages today use this algorithm.[1]

Fumitada Itakura: I continued my hobby of visiting mathematics libraries and accidentally found an interesting paper which transformed the autocorrelation function. It didn't say autocorrelation at all, but that a positive definite function could be expanded using line-spectrum type of transformation. The language was completely mathematical. But if I interpreted it in my engineering approach, the autocorrelation function could be expanded using minimum line-spectrum and the frequency and amplitude combination. That was the so-called LSP [line spectrum pair] theory. So I was quite lucky to find good mathematics—a very good mathematician paved the way for the speech scientist.[2]

[1]Thomas S. Huang oral-history interview 20 March 1997, p. 17.

[2]Fumitada Itakura oral-history interview 22 April 1997, p. 20.

The concepts of probability theory and statistics had long been part of signal processing, as, for example, in Bernard Widrow's work in the 1950s on quantization noise.[29] Throughout the 1960s, and later, Thomas Kailath and Enders Robinson published many papers that applied statistical theory, operator theory, and state-space techniques to a variety of different prob-

[29]Bernard Widrow, "A study of rough amplitude quantization by means of Nyquist sampling theory" (*IRE Transactions on Circuit Theory*, vol. CT-3 (1956), pp. 266–276), and Bernard Widrow oral-history interview 14 March 1997. Even earlier, Norbert Wiener's work on prediction and filtering involved statistical concepts.

lems.[30] In about 1970 the theory of Markov chains began to be applied to problems in speech processing, as in the work on speech recognition by James K. Baker and Fred Jelinek at IBM.[31] By the mid 1980s this type of statistical pattern recognition had found a wide range of speech applications.[32]

The 1970s saw the initiation of a great deal of work on facilitating communication between computers and people. At Bell Labs a system able to speak messages from a stored vocabulary was developed, and it found use in speaking the instructions for wiring telephone equipment (since the technician could work more rapidly and accurately if he did not need to look away from the work to read instructions).[33] (See Figure 4.) There was work on automatic speaker-recognition, either to identify speakers or to verify that a voice belongs to the person alleged. In the 1970s systems capable of high success rates were demonstrated, yet little practical use was found for automatic speaker-recognition, a situation which continued into the 1990s.[34]

The challenge of automatic speech recognition was also taken up, the hope being that people could dial a number by speaking it or even give oral instructions to computers. A landmark advance was Fumitado Itakura's introduction of a distance metric, between the utterance and templates of speech within the computer, that became widely used in the field.[35] In the early 1970s researchers at Lincoln Lab began a line of work destined to assume great importance when they invented techniques for sending speech over packet networks.[36]

---

[30]See, for example, Thomas Kailath, *Lectures on Kalman and Wiener Filtering Theory* (New York: Springer-Verlag, 1981); Thomas Kailath, *Linear Systems* (Englewood Cliffs, NJ: Prentice-Hall, 1980); Enders A. Robinson, *Random Wavelets and Cybernetic Systems* (London: Charles Griffin and Company, 1962); and Enders A. Robinson, *Statistical Communication and Detection with Special Reference to Digital Data Processing of Radar and Seismic Signals* (London: Charles Griffin and Company, 1967).

[31]Steve Young, "A review of large-vocabulary continuous-speech recognition" (*IEEE Signal Processing Magazine*, vol. 13 (1996), no. 5, pp. 45–57).

[32]Lawrence R. Rabiner and B.H. Juang, "An introduction to hidden Markov models" (*IEEE ASSP Magazine*, vol. 3 (1986), no. 1, pp. 4–16).

[33]Millman *op. cit.*, pp. 129–131, and Manfred Schroeder oral-history interview 2 August 1994, p. 35.

[34]Herbert Gish and Michael Schmidt, "Text-independent speaker recognition" (*IEEE Signal Processing Magazine*, vol. 11 (1994), no. 4, pp. 18–32), and Douglas O'Shaughnessy, "Speaker recognition" (*IEEE ASSP Magazine*, vol. 3 (1986), no. 4, pp. 4–17).

[35]Millman *op. cit.*, p. 132.

[36]Eva C. Freeman, ed., *MIT Lincoln Laboratory: Technology in the National Interest* (Lexington, MA: MIT Lincoln Laboratory, 1995), p. 231. An important publication in this area was Gold's 1977 paper on "Digital speech networks" (*Proceedings of the IEEE*, vol. 65 (1977), pp. 1636–1658).

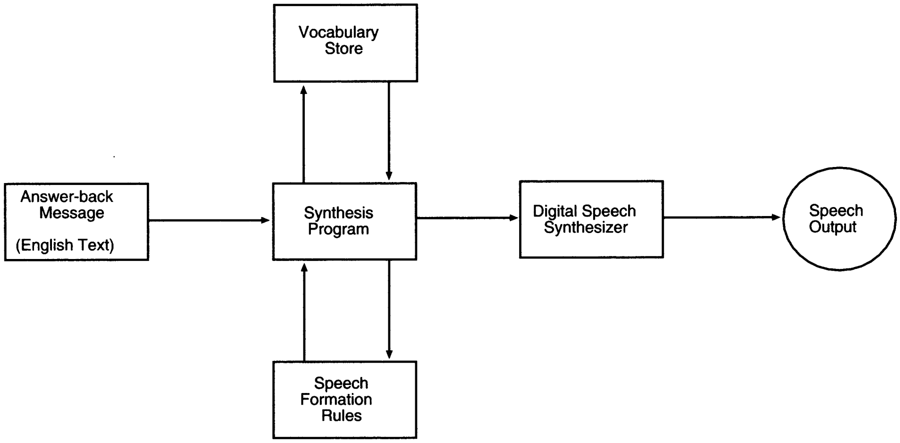

FIGURE 4. This diagram shows the basic design of the Bell Labs computer voice-response system that was used for speaking the instructions for wiring telephone equipment. (Redrawn after figure on p. 130 of Sidney Millman, ed., *A History of Engineering and Science in the Bell System: Communication Sciences (1925–1980)* (AT&T Bell Telephone Laboratories, 1984).)

The 1970s were a watershed in the history of signal processing for the development of signal-processing hardware. The decade opened with the completion of what may be the first real-time DSP computer: the Lincoln Fast Digital Processor (FDP).[37] (See Figure 5.) A special-purpose, parallel-processing computer, the FDP performed signal-processing tasks about a hundred times as fast as the general-purpose computers of the time.[38] There followed in 1974 the Lincoln Digital Voice Terminal, a computer built to carry out a variety of speech-compression algorithms.[39] A more powerful version, the Lincoln Digital Signal Processor (LDSP), was built in four copies and remained the main research tool for the Signal Processing Group at Lincoln Lab into the 1980s.[40]

The future, however, belonged to single-chip signal processors. Throughout the 1960s integrated-circuit techniques advanced, and, in the form of calculators and watches, integrated circuits came into the hands and onto the

[37]Don Johnson, "Rewarding the pioneers" (*IEEE Signal Processing Magazine*, vol. 14 (1997), no. 2, pp. 20–22).

[38]Freeman *op. cit.*, p. 228. The FDP contained 10,000 separate integrated-circuits and had a multiply time of 900 nanoseconds.

[39]Freeman *op. cit.*, p. 228.

[40]Ben Gold personal communication 30 January 1998.

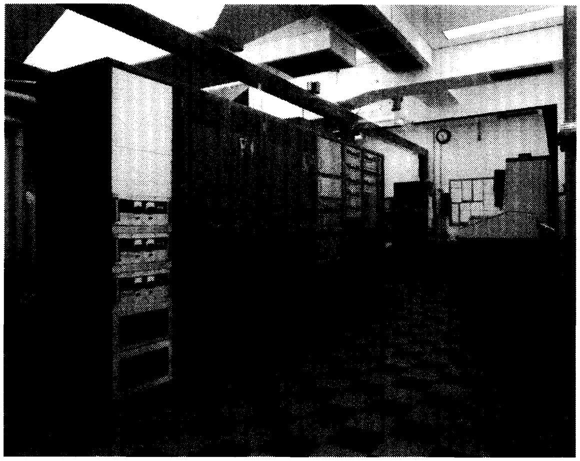

FIGURE 5. This photograph shows the Lincoln Fast Digital Processor (FDP), built under Charles Rader's direction at Lincoln Lab. (Lincoln Lab photo reproduced by permission.)

wrists of a great many people. Going beyond a calculator chip (having hard-wired functions), Intel produced the first (programmable) microprocessor chip, the Intel 4004 in 1971.[41] The following year Intel introduced the 8008, a microprocessor that operated on 8-bit, rather than 4-bit, words.[42] Microprocessors were used in some DSP devices, such as the LPC vocoder, completed at Lincoln Lab in 1977.[43] Because the usual signal-processing operations of filtering and transforming are calculation-intensive, involving huge numbers of multiplications, the introduction in 1976 by TRW of a 16-bit by 16-bit single-chip multiplier stimulated the development of real-time DSP machines.[44]

---

[41]At the time, the chip was called a 'micro-programmable computer on a chip'; the term 'microprocessor' came into use in 1972 [Ernest Braun and Stuart Macdonald, *Revolution in Miniature: The History and Impact of Semiconductor Electronics*, second edition (Cambridge: Cambridge University Press, 1982), p. 108]. The designer of Intel's first microprocessor, Marcian E. "Ted" Hoff, was a graduate student of signal-processing pioneer Bernard Widrow; Hoff was also the designer of the first Intel programmable DSP chip, the Intel 2920 (mentioned below) [Widrow personal communication 2 February 1998].

[42]Braun and Macdonald *op. cit.*, pp. 108–109.

[43]Freeman *op. cit.*, p. 230.

[44]Mark Kahrs, "Professional and consumer gear: hardware & software" (*IEEE Signal Processing Magazine*, vol. 14 (1997), no. 5, pp. 51–57). The TRW multiplier (the MPY-16) was introduced in 1976 and advertised in the 4 July 1976 issue of *Electronics* [Earl Swartzlander personal communication 30 January 1998].

Thomas Kailath: Through a student from industry who came to us, we got interested in an area of antenna array processing. You have signals coming from different directions. You have an antenna array, and you have got to separate these signals. The traditional methods for doing that are really equivalent to taking the FFT of the data. It's a spatial FFT rather than a temporal FFT, but it's similar. If you have sinusoidal waveforms in noise, you would tend to get peaks of the FFT at the sinusoidal frequencies. Similarly, when you take the spatial FFT of signals coming from different directions, you tend to get peaks in those directions. This is non-parametric, and it doesn't take account of the fact, for example, that these are plane waves coming in, or that there are only two or three of them.

The FFT doesn't care about those things. You have numbers, you take the FFT. So, one of our students, Ralph Schmidt, said, "Well, even if you have no noise, that method can't solve the problem exactly. But, if you add the information that you believe there are only, say, three plane waves coming in from a few directions, then we can get an exact solution without noise. And with noise present, we can deduce an efficient algorithm that gives good estimates of the direction." That idea launched a new field of high-resolution model-based sensor array processing. And I had six or seven students work in that area. It was timely because those were the days of SDI, and that was one of the funding sources for this, because they were interested in determining the direction of incoming missiles.[1]

[1]Thomas Kailath oral-history interview 13 March 1997, pp. 6–7.

Chips designed specifically for signal processing began to appear at the end of the decade. The Speak & Spell toy contained a speech synthesis chip (TMC0281),[45] and, as mentioned above, in 1979 Bell Labs completed a fully-integrated single-chip adaptive echo-canceler.[46] In February 1979 Intel introduced a single-chip DSP computer, the Intel 2920, but its lack of a hardware multiplier limited its speed, and its 9-bit analog-to-digital and digital-to-analog converters limited its accuracy.[47] As the decade came to a close, a number of companies—AMI, AT&T, Intel, Matsushita, Motorola,

[45]Gene Frantz and Panos Papamichalis, "Introduction to DSP solutions" (*Texas Instruments Technical Journal*, vol. 13 (1996), no. 2, pp. 5–16).

[46]Gritton and Lin *op. cit.*

[47]Harvey G. Cragon, "The early days of the TMS320 family" (*Texas Instruments Technical Journal*, vol. 13 (1996), no. 2, pp. 17–26), and Edward A. Lee, "Programmable DSP architectures: part I" (*IEEE ASSP Magazine*, vol. 5 (1988), no. 4, pp. 4–19).

James Kaiser: Hank McDonald came into my office—I remember this very well—one Friday afternoon and he said, "Jim, look, I've got this idea." Then he described to me in detail his idea on multiplexing. He showed me how he wanted to do it, and he was doing serial arithmetic, not parallel arithmetic. That had the advantage of using only one wire with a number of sequential bits on that single line. That way, when you switch things around to do the multiplexing you are only switching one line, which is a whole lot easier than switching 12 lines all at exactly the same time for a 12-bit number. So Hank, after he outlined his multiplexing scheme asked, "What's wrong with it?" I just sat and looked at his scheme and finally said, "Hank, I can't see anything wrong with the idea." We discussed it more, talked over what we thought the properties of the design were, and I said, "I just cannot see any reason why that shouldn't work. In fact, that structure will let us do both recursive filter implementations and non-recursive filter implementation, to whatever order we want. That is an absolutely beautiful structure."

So at that point, Hank started to do a detailed hardware design. He had a number of projects also going on, however, so he enlisted the help of Leland Jackson, a student we had with us who was beginning to work on a Ph.D. at Stevens Institute of Technology. Jackson's work at Stevens was on analyzing quantization effects in digital filters, both correlated noise (the limit cycle problem), and then uncorrelated noise—i.e., how you sequence the computations you are going to do to minimize noise when you are building a complicated filter. Under Hank's tutelage, Leland did the hard work of implementing Hank's structure. We completed that implementation in '67 and reported on it at the IEEE Convention in New York. That was McDonald and Jackson, and me, but I was by far the least important contributor to the team. I mean, Hank had the basic multiplexing idea, Leland basically built it, and I helped with just a few of the examples and acted as a sounding board. That was my part in it. That implementation paper really got things going.[1]

[1]James Kaiser oral-history interview 11 February 1997, pp. 16–17. The paper was Leland B. Jackson, James F. Kaiser, and Henry S. McDonald, "An approach to the implementation of digital filters" (*IEEE Transactions on Audio and Electroacoustics*, vol. 16 (1968), pp. 413–421).

NEC, and Texas Instruments—were working energetically to design and build single-chip DSPs.[48] As we will see in the next chapter, success came in the early 1980s.

[48]Cragon *op. cit.* The AMI S2811 was announced in 1978, but not delivered until later; at Bell Labs, the single-chip DSP1 was completed in 1979, but was not marketed outside the company [Lee *op. cit.*].

James Kaiser: It was at that point—1973 or so—I wrote another conference paper. This was for the IEEE Symposium on Circuits and Systems, ISCAS '74. It was a 4-page conference paper, and it talks about I^0 sinh and it lays out FIR design by the window function method. It's very complete, all in just one little 4-page paper. That made a little more of an impact, but still, I think when engineers read "Bessel functions," they don't want to know about it. They don't want to know about Bessel functions at all, unless they're in electromagnetic theory. Then it's second nature to them, but otherwise, their attitude about Bessel functions is, "I'll take your word for it, but don't bother me with the details." So people don't use it, simply because they haven't taken that extra little 5% effort that's required to understand what's going on. I call this the "Bessel function syndrome."[1]

[1]James Kaiser oral-history interview 11 February 1997, p. 43.

In the late 1960s and early 1970s, analog-to-digital converters (ADCs) were getting faster and faster, the speed doubling roughly every two years; and in the early 1970s ADCs and DACs (digital-to-analog converters) with 16-bits of dynamic range were available. This further stimulated the development of digital electronics in instruments, such as the digital oscilloscope, one of the first being the 1090A Explorer put on the market in 1972 by Nicolet Instruments.[49] Another new technology of the 1970s, fiber optics (discussed below), also improved instruments, as fiber-optic sensors came to be widely used in measuring devices.[50]

Large-scale-integrated electronics was certainly the most momentous new technology of the 1970s, but there were two other new semiconductor technologies of great value to signal processing. These were charge-coupled devices (CCDs) and surface-acoustic-wave devices (SAW devices), both of which appeared about 1970. CCDs were often used as optical sensors, as in video cameras, but they were also used for time delay, as in echo generation for music or ghost cancellation in television.[51] SAWs turn an electrical sig-

[49]Karen Fitzgerald, "Digital scopes" (*IEEE Spectrum*, vol. 25 (1988), no. 11, pp. 82–86).

[50]Gerhard M. Sessler, "What's new in electroacoustic transducers" (*IEEE ASSP Magazine*, vol. 1 (1984), no. 3, pp. 3–11).

[51]Robert W. Broderson and Richard M. White, "New technologies for signal processing" (*Science*, vol. 195 (1977), pp. 1216–1222).

HANS GEORG MUSMANN: At that time [in the mid 1970s] the problem was the digitization of a video signal, which is required for realizing such a visual communication system, and reducing the bit rate for moving images, which is almost three thousand times that of a speech signal. I always said our goal would be to cut down the bit rate of a video signal to that of a speech signal. Otherwise it's too expensive to use it.

INTERVIEWER: The factor of three thousand is for television resolution?

MUSMANN: Yes. Of course, you can use a smaller picture. That was also proposed later on. But even if you take an image one fourth as large, then you'll still have a factor of eight hundred or something like that. You can reduce the frame frequency from fifty hertz to ten hertz. Then you come down to a hundred times the rate of the speech signal. Nobody thought at that time that this compression factor could be achieved by coding....

This indicated to me that if we wanted to create a visual communication system, we would have to cut down the bit rate for visual information to that of speech—or be close to it. Otherwise, nobody would be able to afford it. So we studied some techniques. And, in 1979, two years after the facsimile work, we demonstrated in the United States a transmission of moving images requiring just 64 kilobits per second....

There was a big interest. The communications industry saw that it might be possible to transmit moving images via speech channels. But they were still hesitating, waiting to see if the problems with motion could be solved in the future. We transmitted mainly those parts of an image which had changed, and we took the other parts from the stored preceding image in the memory of the receiver.

INTERVIEWER: So the receiver had to have a frame memory?

MUSMANN: Yes, and this memory was a problem at that time. The memories of computers were not transistor memories, but magnetic-core memories. Each core was one bit. I needed a frame memory: 400,000 picture elements and 8 bits per picture element. The price was 250,000 D-marks![1]

ALAN OPPENHEIM: The reason why it was so significant was that, prior to the Speak-&-Spell, you could basically think of digital signal processing as high-end, funded largely by the military or by high-end industry like the seismic industry, because it was expensive to do.[2]

[1]Hans Georg Musmann oral-history interview 30 August 1994, pp. 16–18.

[2]Alan Oppenheim oral-history interview 28 February 1997, p. 36.

Lawrence Rabiner: People and companies jumped in: TI, Motorola, and Rockwell, jumped in really quickly. Academia jumped in really quickly. Al Oppenheim and Ron Schafer had a book out there that was focused on the academic audience. Ben Gold and I had another book out there with more of an engineering focus. Ben and Charlie [Rader] had this early book out, but it was, like all early books, a little too early. Even in '75, when the books came out, I think the first line stuff was pretty well-developed, and that was the real key.... There was a tremendous pent-up demand for this stuff. You can go out and do things with a computer, and after a while it became digital hardware and then it became real.[1]

Charles Rader: Now, at the same time that I had found a way to express a Fourier transform as a convolution, a colleague of mine, by the name of Leo Bluestein, who was then at Sylvania, but who had actually shared an office with me a few years earlier when he was at Lincoln Lab, came around one day and said, "I've got this interesting result." He said, "I can do a Fourier transform where the number of points is a perfect square, and I have this algorithm." And his algorithm was not very interesting in itself, but part of the way through explaining the algorithm, he had done some manipulations to change a Fourier transform into a convolution, in still another way. It had nothing to do with number theory. It had to do with multiplying the input wave form by what engineers would call a chirp wave form. It was a sinusoid whose frequency increases continuously as time progresses. And then, if you would agree afterwards to un-multiply the Fourier transform by a chirp, what you found in the middle was you were convolved with a chirp. So, multiplication involved, post-multiplication to undo it, was another way of converting a Fourier transform to a convolution.

I said to him, "Leo, we can use the FFT to do convolutions, so forget about your clever little algorithm for doing the middle part. Let's use the FFT for that." The advantage of that was that the chirp algorithm would work for any length sequence, and so you could do any length sequence using any length FFT. That was the so-called chirp Z transform.[2]

[1]Lawrence Rabiner oral-history interview 13 November 1996, p. 20.

[2]Charles Rader oral-history interview 27 February 1997, pp. 15–16.

nal into an acoustic signal and back again; applications include pulse expansion, pulse compression, and transverse filters.[52]

[52]Broderson and White *op. cit.*, and Thomas Kailath, ed., *Modern Signal Processing* (Washington, DC: Hemisphere Publishing, 1985), pp. 374–375.

Just as in the previous decade, photographs from space caught the public attention in the 1970s. The Soviet Venera 7 sent back the first pictures from the surface of Venus in 1970; the U.S. Mariner 10 reached Mercury in 1973; and U.S. Viking 1 and 2, launched in 1975, sent back the first pictures from the surface of Mars the following year. There was also the radar mapping of Venus by the U.S. Magellan probe in 1980. Signal processing was involved in the encoding, reconstruction, and enhancement of these images, and it played a large part in all communications with satellites and space probes.[53]

In the 1970s digital signal processing was increasingly applied to radar and sonar imaging. Researchers at ERIM and Lincoln Lab showed that DSP was useful in forming radar images of targets, determining speed of targets, and eliminating clutter.[54] In the 1970s underwater acoustic systems reached a high level of performance, and the problem of separating and classifying signals came to the fore, as attention shifted from signal acquisition to signal processing.[55] And there were important advances in geophysical signal processing, such as Richard Lacoss's work on adaptive filters for seismic applications.[56]

Communications entered a new era in the 1970s with the development of fiber-optic technology. Light, because of its high frequency, promised great bandwidth for communications. Required, though, would be a controllable and intense source of light and a low-loss transmission medium. Rather suddenly, in about 1970, both of these became feasible: Corning Glass demonstrated highly transparent fibers, and Bell Laboratories demonstrated semiconductor lasers that could operate at room temperature. (See Figure 6.) In 1977 fiber-optic telephone systems were put into service both by General Telephone & Electronics (in Santa Monica, California) and AT&T (in Chicago).[57] Just over a decade later the first fiber-

---

[53]Much of the communications was handled by the National Aeronautics and Space Administration's Deep Space Network, operated by the Jet Propulsion Laboratory [John R. Pierce and A. Michael Noll, *Signals: The Science of Telecommunications* (New York: Scientific American Library, 1990), p. 198].

[54]Dale A. Ausherman, Adam Kozma, Jack L. Walker, Harrison M. Jones, and Enrico C. Poggio, "Developments in radar imaging" (*IEEE Transactions in Aerospace and Electronic Systems*, vol. 20 (1984), pp. 363–400); and Freeman *op. cit.*, p. 228.

[55]William S. Burdic, *Underwater Acoustic Signal Analysis* (Englewood Cliffs, NJ: Prentice-Hall, 1984), p. 14.

[56]Richard Lacoss, "Data adaptive spectral analysis methods" (*Geophysics*, vol. 36 (1971), pp. 661–675).

[57]Trudy E. Bell, "Fiber optics" (*IEEE Spectrum*, vol. 25 (1988), no. 11, pp. 97–102).

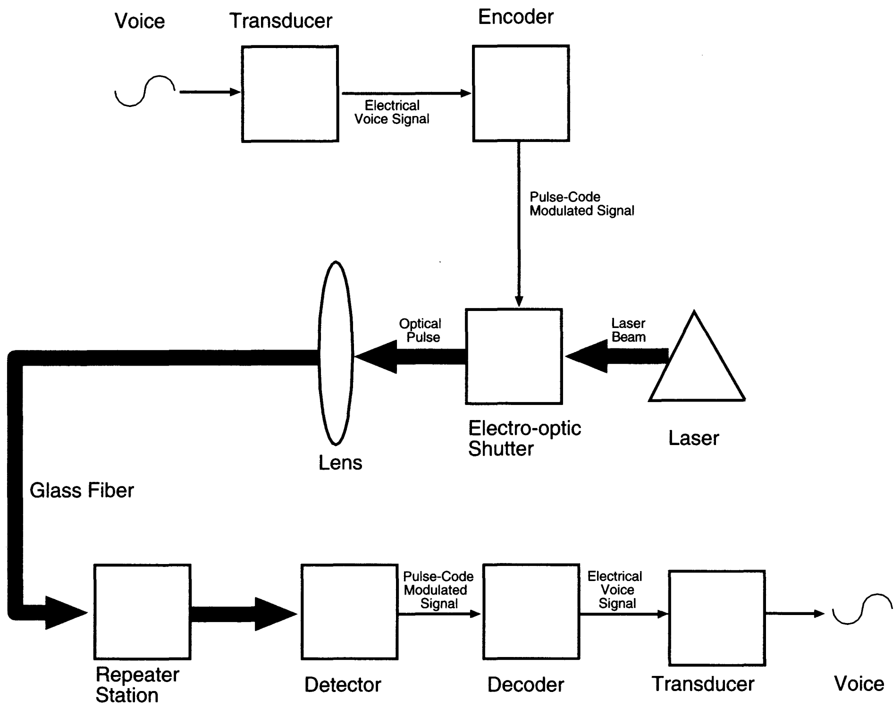

FIGURE 6. Fiber-optic cable is typically employed as shown. The analog signal is converted to electrical pulses, which control a shutter in front of a laser beam directed down the cable. En route the optical signal may be strengthened at repeater stations. At the receiving end, the optical pulses are converted to electrical pulses, from which is recovered the original analog signal. (Redrawn after figure on p. 51 of *Britannica Yearbook of Science and the Future 1975* (Chicago: Encyclopaedia Britannica, 1974).)

optic cable across the Atlantic went into service; its capacity exceeded all previous cables combined.[58]

Signal processing came to public attention also through music. The early music synthesizers, such as Robert Moog's, were analog devices, with oscillators, voltage regulators, filters, and other components to produce and manipulate the waveforms that became music (or at least output). In the 1970s there appeared synthesizers capable of digital sampling, so that any

[58]*Britannica Yearbook of Science and the Future 1991* (Chicago: Encyclopaedia Britannica, 1990), p. 325.

Charles Rader: In the mid-'70s there was a congressional committee on assassinations, the assassination of President Kennedy and Martin Luther King, among others. That committee was given testimony from some acoustics experts who believed that one of the police radio channels in Dallas had recorded the gunshot sounds. And by a further analysis, they thought they could prove that one of what they thought were four shots had originated not from the school depository building, but from the so-called grassy knoll....

So, the Justice Department asked the National Academy of Engineering to put together a team and look at this data and technique and judge its validity. Along with several Nobel Prize winners, like Luis Alvarez, I was on this committee. I played a significant role in proving the negative conclusion that the sounds that were being identified as gunshots were in fact distorted human speech. I knew what the speech was, and could associate it with a time, about a whole minute after the assassination, because it was a Dallas sheriff talking about moving men around to check out the possible site where somebody had seen somebody suspicious. We were able to figure out a lot about this horribly distorted data.[1]

[1]Charles Rader oral-history interview 27 February 1997, pp. 52–53. Another signal-processing pioneer, Bernard Widrow, was involved with this investigation. The Dallas police had recorded the sound from a police motorcycle with a stuck transmit-switch that was believed to have been located near the site of the assassination. A method published shortly before by B. Widrow *et al.* ("Adaptive noise cancelling: principles and applications", *Proceedings of the IEEE*, vol. 63 (1975), pp. 1692–1716) was used to remove the noise of the idling motor of the motorcycle. [Widrow personal communication 2 February 1998.]

instrument, or any sound at all, could be recorded, modified, and replayed.[59] In the 1970s in Paris, IRCAM (Institut de Recherche et Coordination Acoustique/Musique) embarked on its mission of integrating traditional musical media with the new medium provided by the computer. Giuseppe Di Giugno at IRCAM designed a device, the 4X, for electronically modifying sounds of traditional instruments. The 4X has been used in performances, as, for example, of the 1981 composition *Répons* by Pierre Boulez for "six instrumental soloists, chamber orchestra and real-time digital-signal processors".[60]

[59]Lubar *op. cit.*, p. 193.

[60]Pierre Boulez and Andrew Gerzso, "Computers in music" (*Scientific American*, vol. 258 (1988), no. 4, pp. 44–50).

MANFRED SCHROEDER: We were working on speech synthesis because by taking speech apart and synthesizing it again at the other end we could compress its bandwidth by a factor of 5 or more.

... [But] it was not just bandwidth compression. Another reason we were interested in speech synthesis was to read documents, as in reading machines for the blind. Optical scanners were already in existence in the '60s, but it was not easy to get speech from such devices. We wanted to do that, even if that was not immediate telephone business.

There was a beautiful application made by Western Electric, for the guys who wire these complicated circuits. Here's a complicated circuit-chart. In wiring it, they would often solder a wrong connection. So someone... wrote an automatic program that translated the wiring chart into a code that we translated into spoken instructions. The wiring man at Western Electric used earphones. He never had to turn his eyes off what he was doing. The earphones would tell him to connect the green wire to terminal 47, that kind of thing. That was, I think, the first application of synthetic speech within the Bell System.[1]

HANS SCHUESSLER: Very early Alfred Fettweis started this idea to transform an analog selective system into a wave-digital filter and to obtain just a digital filter. The main point is that in that case the starting point is and has to be an analog filter. But the knowledge about the design of analog selective systems is more or less gone. And that's why he did not have so much success in the States, as far as I see it.[2]

ROBERT W. LUCKY: One of the more practical engineers in the audience [at a workshop about 20 years ago] stood to tell people how he had made a custom VLSI chip that did some powerful signal-processing algorithms. At that time, such a feat was very unusual and the people in the audience mostly held the belief that VLSI chips were the exclusive province of a small number of designers chained in the basement at Intel or somewhere like that.

I remember distinctly how the idea that an ordinary engineer could make a custom chip for mathematical algorithms created an unwelcome thought in the audience. People averted their glances and stirred in their seats.... It wasn't that the mathematicians and engineers present at this workshop were actively against VLSI or anything like that. It was just that this new capability introduced a new element into their world—an element that they did not understand and one that they personally would be unlikely to master.[3]

[1]Manfred Schroeder oral-history interview 2 August 1994, p. 62.

[2]Hans Wilhelm Schuessler oral-history interview 21 April 1997, p. 16.

[3]Robert W. Lucky, *Lucky Strikes ... Again* (New York: IEEE Press, 1993), p. 229.

The IEEE Group on Audio and Electroacoustics, which, as already mentioned, had established a technical committee for digital signal processing, did much to advance the new technology. Since DSP was a new field and its practitioners had quite different backgrounds, the terminology was not fixed, with different words being used for the same thing and the same word being used in slightly different ways. The G-AE helped to remedy this situation with the publication in 1972 of detailed recommendations for the use of some 200 terms.[61] The G-AE worked with the newly established IEEE Press to publish in the 1970s several volumes of reprints—many of the important pioneering papers in DSP were otherwise hardly available—and a DSP bibliography in 1972, with an updated version in 1975.[62] The G-AE worked with the IEEE Press to put out in 1979 a set of DSP algorithms, expressed in FORTRAN, which was widely influential.[63] The IEEE Press also published in the 1970s several reprint volumes of papers on speech communications.[64]

Another notable event of the decade was the initiation of the International Conference on Acoustics, Speech, and Signal Processing (ICASSP) in 1976, and ICASSP has been held every year since. Two name changes were significant: in 1974 the Group changed its name from Audio and Electroacoustics to Acoustics, Speech, and Signal Processing, and in 1976 it attained IEEE Society status and thus became the Acoustics, Speech, and Signal Processing Society.[65]

Through these and countless other activities, signal processing was emerging as a recognized discipline. There was a professional society, with

---

[61]Lawrence R. Rabiner, James W. Cooley, Howard D. Helms, Leland B. Jackson, James F. Kaiser, Charles M. Rader, Ronald W. Schafer, Kenneth Steiglitz, and Clifford J. Weinstein, "Terminology in digital signal processing" (*IEEE Transactions on Audio and Electroacoustics*, vol. 20 (1972), pp. 322–337).

[62]The first two reprint volumes, which were very widely used, were Lawrence R. Rabiner and Charles M. Rader, eds., *Digital Signal Processing* (New York: IEEE Press, 1972), and Digital Signal Processing Committee, eds., *Selected Papers in Digital Signal Processing II* (New York: IEEE Press, 1976). The two bibliographies were Howard D. Helms and Lawrence R. Rabiner, *Literature in Digital Signal Processing: Terminology and Permuted Title Index* (New York: IEEE Press, 1972), and Howard D. Helms, James F. Kaiser, and Lawrence R. Rabiner, *Literature in Digital Signal Processing: Author and Permuted Title Index*, revised and expanded edition (New York: IEEE Press, 1975).

[63]Digital Signal Processing Committee, eds., *Programs for Digital Signal Processing* (New York: IEEE Press, 1979).

[64]Rabiner *op. cit.* (footnote 28).

[65]Rabiner *op. cit.*

Richard Hamming: So I turned to a friend, Jim Kaiser (J.F. Kaiser), who was one of the world's experts in digital filters at that time, and suggested he should stop his current research and write a book on digital filters—book writing to summarize his work was a natural stage in the development of a scientist. After some pressure he agreed to write the book, so I was saved, so I thought. But monitoring what he was doing revealed he was writing nothing. To rescue my plan I offered, if he would educate me over lunches in the restaurant (you get more time to think there than in the cafeteria), to help write the book jointly (mainly the first part), and we could call it Kaiser and Hamming. Agreed!

As time went on I was getting a good education from him, and I got my first part of the book going but he was still writing nothing. So one day I said, "If you don't write more we will end up calling it Hamming and Kaiser."—and he agreed. Still later when I had about completed all the writing and he had still written nothing, I said I could thank him in the preface, but it should be called Hamming, and he agreed—and we are still good friends![1]

*Richard W. Hamming, *The Art of Doing Science and Engineering: Learning to Learn* (Amsterdam: Gordon and Breach, 1997), pp. 164–165.

publications, conferences, and workshops. There were classic texts, notably Alan Oppenheim and Ronald Schafer's *Digital Signal Processing* and Lawrence Rabiner and Ben Gold's *Theory and Application of Digital Signal Processing*, which formed the foundation for courses at major universities.[66] And both techniques and applications were being rapidly developed.

[66]Alan V. Oppenheim and Ronald W. Schafer, *Digital Signal Processing* (Englewood Cliffs, NJ: Prentice-Hall, 1975 and 1988), and Lawrence R. Rabiner and Ben Gold, *Theory and Application of Digital Signal Processing* (Englewood Cliffs, NJ: Prentice-Hall, 1975). Large numbers of DSP engineers first learned the subject from these two books; the Oppenheim and Schafer text was the standard introduction, while the Rabiner and Gold text had more of an engineering focus. Perhaps the first DSP course given at a university was one given at MIT in 1965; the material taught in this course formed a large part of Ben Gold and Charlie Rader's *Digital Processing of Signals* (New York: McGraw-Hill, 1969) (as stated on p. vii of this book).