Chapter 4

Going Digital: The 1960s

What was Happening in the 1960s

movies people were watching:
The Graduate with Dustin Hoffman
James Bond movies
Midnight Cowboy

TV shows people were watching:
"Twilight Zone"
"Perry Mason"
"The Beverly Hillbillies"
"Mission Impossible"

music people were listening to:
the Beatles and the Rolling Stones
Motown
Simon and Garfunkel's "Mrs. Robinson"

books people were reading:
Rachel Carson's *Silent Spring*
Joseph Heller's *Catch-22*
Harper Lee's *To Kill a Mockingbird*

Quite in contrast to the 1950s, the 1960s in the United States were turbulent: the civil rights movement, which had begun a decade earlier, had grown in strength; the anti-war movement escalated even more rapidly than U.S. military involvement in Vietnam; another movement too, the one for women's liberation, aroused great energies and passions; and there were the assassinations of John Kennedy, Robert Kennedy, and Martin Luther King. In Europe as well as the United States, youth were questioning the established order and demanding change or, in the case of the Hippies, "dropping out". Despite a decade of economic growth based in part on technical advances and of individual technological triumphs, such as the IBM 360 computer, the first heart transplants, and the Moon landing, there arose a widespread anti-technology sentiment.

Telephones were much in the news in the 1960s: the use of satellites to relay telephone signals (and a song entitled "Telstar" headed the popular music charts for three weeks in 1962)[1] (see Figure 1); the introduction of, and the storm of opposition to, All Number Calling (ending the use of letters, as in PE(nnsylvania) 6-5000);[2] the "hot line", with a red telephone at each end, between Washington and Moscow, which was put into service in

[1]*Communications* (a volume of the series Understanding Computers) (Alexandria, VA: Time-Life Books, 1986), p. 89.

[2]John Brooks, *Telephone: The First Hundred Years* (New York: Harper & Row, 1975), pp. 270–273.

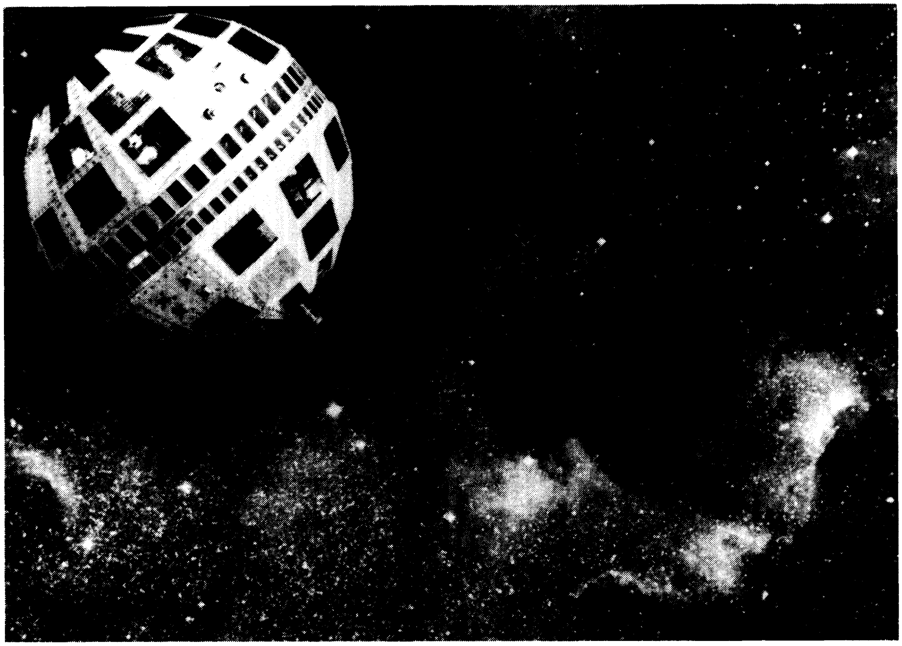

FIGURE 1. The Telstar communications satellite was launched in 1962. (Bell Labs photo reproduced by permission.)

1963; and the introduction the same year of Touch-Tone phones (which used a pair of tones rather than a sequence of pulses to signal a digit of a telephone number). A small cult, the "phone phreaks", became fascinated with telephone technology, and in the next decade many of them moved on to computing as a hobby.[3] And in 1970 both overseas direct-dialing and Picture-Phone service began.[4]

As we have seen, Bell Labs engineers from the 1920s on had been exploring ways to increase the capacity of the telephone system, and in late 1955 management decided to develop a pulse code modulation (PCM) sys-

[3]In 1971 Steven Jobs teamed up with Steve Wozniak to design and sell "blue boxes" that produced the tones to control long-distance telephone switching (so that one could place a call without being billed). Later Jobs and Wozniak founded Apple Computer and launched the era of the personal computer. [James W. Cortada, *Historical Dictionary of Data Processing: Biographies* (New York: Greenwood Press, 1987), pp. 144–146, 287–288.]

[4]*Britannica Yearbook of Science and the Future 1971* (Chicago: Encyclopaedia Britannica, 1970), pp. 166–167. In 1968 Stanley Kubrick's movie "2001: A Space Odyssey" made audiences aware both of picture phones and computer-synthesized speech.

James Cooley: An important part of my education was that people always thought I was in digital signal processing and knew a lot about it, but I wasn't. So in the beginning, when programs weren't available, people would call asking for programs, and I got to talk to them. I learned from them while they thought that they were learning from me. It was a good way to get started in digital signal processing.[1]

James Flanagan: Once we had computers [in the 1960s] that could do reasonably fast simulations and get signals in and out, we could design things like adaptive differential PCM, which is now deployed in the telephone system. This technique conserves bandwidth and doubles the capacity of PCM channels. We also started automatic speech recognition, which had been an interest earlier at Bell Labs ....

Interviewer: It sounds like your research was very much influenced by what computers you had.

Flanagan: Yes, indeed. It was dependent on what the computers were able to do. You didn't go into a huge project that had no chance of computing a result in your lifetime. I did get close to that, however, with a colleague named Kenzo Ishizaka, who is a professor in Japan now. He worked with us for a number of years. We tried to make a voice mimic system—it's still a current research problem—in which the computer models the articulatory system, and synthesizes a signal that tries to match arbitrary natural speech input. It looks at the spectral difference between the natural input and the synthetic version. Then it tries to drive the error to zero by adjusting the parameters of the articulatory model. It's a gradient descent algorithm. We ran our first version of this on a Cray I, and the synthesizer ran something like several hundred times real time. That was on a Cray I. So now we have it down to five or six times real time on a C-90.[2]

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

[2]James Flanagan oral-history interview 8 April 1997, pp. 21, 23–24.

tem. There resulted the so-called T-1 carrier system, which was put into service in 1962 as the world's first common-carrier digital communications system. The speech waveform was sampled 8000 times per second to represent the usual 4000-Hz channel. The amplitude of each sample was described by seven binary digits, using nonlinear quantizing to cover better the wide dynamic range of speech, and an eighth bit was used for signaling.

Voice was thus encoded at 64,000 bits per second. The system was designed to allow several conversions from analog to digital to analog, since digital lines would have to work with analog lines.[5]

The T-1 system spread rapidly, helped by new integrated-circuit designs that offered smaller size, lower power-consumption, better performance, and lower costs, and T-1 set a standard for voice coding that dominated telecommunications for decades.[6] By the end of the 1960s other developed countries had implemented their own PCM systems,[7] and in 1983 more than half of the Bell System exchange trunks were digital.[8]

Digital transmission speeded the adoption of digital electronic switching, which was introduced in France in 1970, with Platon, a time-division switching system, and in the Bell System in 1976.[9] Indeed, there emerged a synergy of digital communication, digital switching, and computer-aided information processing in the telephone system. Here the unification of technologies reversed a historical trend of divergence between transmission and switching.[10]

A 1968 ruling of the Federal Communications Commission had far-reaching effects. The Bell System had always forbidden direct attachment of non-Bell equipment to the telephone network, arguing that it might damage the network, but in 1968 the FCC ruled that people could buy and use the Carterphone, a device to link citizen-band radios and telephones. An important precedent for the Carterphone ruling was the case brought before the FCC in about 1960 by the Hushaphone Company. The Hushaphone was a mechanical device, invented by Leo Beranek, that attached to a telephone handset so that speech could not be overheard by other people in the room. After a lengthy hearing, the FCC ordered the Bell System to

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[5]Eugene F. O'Neill, ed., *A History of Engineering and Science in the Bell System: Transmission Technology (1925–1975)* (New York: AT&T Bell Telephone Laboratories, 1985), pp. 538–542.

[6]Robert W. Lucky, *Silicon Dreams: Information, Man, and Machine* (New York: St. Martin's Press, 1989), p. 235.

[7]Robert J. Chapuis and Amos E. Joel, Jr., *Electronics, Computers and Telephone Switching* (Amsterdam: North Holland, 1990), p. 298.

[8]O'Neill *op. cit.*, pp. 562–563.

[9]Chapuis and Joel *op. cit.*, p. 223.

[10]Patrice Flichy, *Dynamics of Modern Communications: The Shaping and Impact of New Communication Technologies* (London: Sage Publications, 1995), p. 128.

Ben Gold: After a while we were able to test our own vocoder with our program pitch detector. It was slow: to analyze two seconds of speech took the computer about two minutes, 60 to 1 real time. So here is what we had to do. We had to take the speech down to the computer, run the speech in through the computer, run the pitch detector program, record, make our 2-track recording, and bring it upstairs to where the vocoder was. It was pretty slow. So we kept saying, "Wouldn't it be nice if we could program the vocoder on the computer?" So we went back to Bell Labs, and visited Jim Kaiser. There may have been other people there, but he's the one I remember. He said he knew how to build certain digital filters. That was just what we needed. We said, "My God, this is fantastic." We could actually build digital vocoders.[1]

Ben Gold: The FFT was really kind of a bombshell. That was what created digital signal processing as a real field, because it has so many possible applications. I still feel that the period starting with some of Kaiser's stuff, some of our stuff, and working its way up to the divulgence of the FFT, from '63 to '67, was when the whole thing exploded and became a real field.[2]

[1]Ben Gold oral-history interview 15 March 1997, p. 7.

[2]Ben Gold oral-history interview 15 March 1997, p. 3.

permit the use of Hushaphones.[11] These rulings opened the gate to the design, manufacture, and marketing of a whole range of telephone-related equipment, such as modems, answering machines, and office communication systems.[12]

The speed at which modems could operate was limited by the distortion of digital signals during transmission (since the shorter the duration of each bit, the more easily a 0 could be mistaken for a 1 or vice versa). To reduce distortion and thus increase the speed of modems, Robert Lucky, a

[11]Beranek writes [personal communication 2 January 1998]: "The Hushaphone Co. had three of us in the hearing room in Washington: its President, whose name was Tuttle, a young lawyer, and me as expert witness. The Bell System had an actual grandstand brought into the courtroom to hold all their lawyers. They hired the best trial lawyer in New York to cross-examine Tuttle and me. Three U.S. circuit court judges were assigned to hear the case and render a decision. The case went on for several weeks. Hushaphone won the case, and [the Bell System] was told it could not prevent this particular foreign attachment from being used."

[12]*Communications op. cit.*, p. 21.

young engineer at Bell Labs, designed an adaptive equalizer, which adjusted itself automatically to suit the particular call-path established. As a result, modems in the early 1970s could operate at 4800 bits per second (and, on dedicated lines, at 9600 bits per second).[13] A similar advance, also made in the 1960s by Bell Labs engineers, was an adaptive echo canceller.[14]

In the T-1 system described above, speech was transmitted at 64 kbps (kilobits per second). An invention made in 1967 by Bishnu Atal and Manfred Schroeder, called adaptive predictive coding or APC, permitted speech transmission of fair quality at just 4.8 kbps.[15] Unlike the T-1 system, which (in digital form) transmitted the actual waveform, APC is, like the vocoder, a parameter-transmitting system: the voice signal is analyzed at the transmitter, parameters derived from this analysis are sent to the receiver, and there the voice signal is synthesized. The trade-off is a great deal of calculation at both transmitter and receiver for a low bit rate.[16]

Carrying further the idea of APC, Atal invented linear predictive coding or LPC.[17] While APC is a waveform synthesis technique used as a speech coder, LPC is a general method of speech analysis, which can be used for speech compression, speech recognition, speech synthesis, and other purposes. LPC, in a variety of forms, came to be widely used.[18] In Japan at about the same time and independently, Fumitada Itakura and Shuzo Saito invented maximum likelihood analysis, and Itakura developed

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[13]*Communications op. cit.*, pp. 18–20; and Robert W. Lucky, "Techniques for adaptive equalization of digital communication systems" (*Bell System Technical Journal*, vol. 45 (1966), pp. 255–286).

[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. 114–115. In 1980 the adaptive echo canceler became available on a single chip and became widely used in the telephone system.

[15]Bishnu S. Atal and Manfred Schroeder, "Adaptive predictive coding of speech signals" (*Bell System Technical Journal*, vol. 49 (1970), pp. 1973–1986). Atal and Schroeder demonstrated APC at an IEEE meeting in Boston in 1967; the auditors heard the original speech, the transmitted signal, which was noise-like, and the reconstituted speech.

[16]Lucky *op. cit.*, pp. 249–255.

[17]Bishnu S. Atal and Suzanne L. Hanauer, "Speech analysis and synthesis by linear prediction of the speech wave" (*Journal of the Acoustical Society of America*, vol. 50 (1971), pp. 637–655).

[18]Millman *op. cit.*, p. 114.

Ben Gold: The other thing that happened was Oppenheim got very interested in what Charlie [Rader] and I were doing. And just around that time the FFT hit. And it was actually an interesting story about how it hit. I was teaching this course, and it was mainly a course on digital filtering the Z transform, different kinds of filters. There was a lot of stuff along the lines of applications to vocoders. I had a TA, a teaching assistant, named Tom Crystal, who was still a graduate student. Well, a lot of MIT students spend time at Bell Labs. One day, during the 1966–67 academic year, when the course was nearly finished, he brought a little document to me after class. It was by Cooley and Tukey. At that time it hadn't been published as a paper, as a journal article, but simply as an internal memo.

I can tell you my reaction. After the first few paragraphs the hair stood up on my head. I said, "This is unbelievable, and I know that this is very, very important." The rest of the paper, I couldn't understand at all. It was all mathematics, and I was just not good at that. It was quadruple sums, really complicated stuff. It was just algebra, but it was very hairy. So I asked Charlie, who is better at that than me, and Tom Stockham, to "translate", because I knew it was important. They came up with some wonderful, wonderful ways of looking at it, which I think actually sparked the whole field.

At that point, given Oppenheim's work, given the FFT, and given the stuff on digital filters, we said, "There's a book here," and Charlie and I sat down. I had written a fair amount of stuff already for my class, and we just sat down, we said "we're going to write a book, we're going to get Oppenheim to write a chapter, going to get Stockham to write another chapter." Charlie wrote the chapter on the FFT, and that was our book.[1]

Ben Gold: So if something came along that the managers [at Lincoln Lab] felt was good but not sponsorable, they'd use in-house money. The FDP, the Fast Digital Processor, was built with in-house money. It cost a lot of money, and the directors got very antsy about it towards the end. So they said, "We've got to use this for something useful. Let's use it for radar." So we all became radar people. For a few years we worked on radar, and Ed Muehe and Bob Purdy were people that I worked with, but it was really the same people, like Charlie and me, who were pushing radar DSP.[2]

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[1]Ben Gold oral-history interview 15 March 1997, p. 12.

[2]Ben Gold oral-history interview 15 March 1997, p. 20.

the PARCOR (partial correlation) method, which is essentially identical with LPC.[19]

It is remarkable that, at the same time that speech researchers invented APC and LPC, John P. Burg, who was analyzing geophysical signals, invented a procedure, called maximum entropy spectral analysis, that, it turns out, is mathematically equivalent.[20] Furthermore, these results were closely related to methods in statistics for fitting autoregressive models to data.[21]

A great deal of calculation was required not only for the implementation of LPC and other coding schemes, but also for the research leading to such schemes. It was not just a coincidence that this happened in the 1960s, the decade in which computer technology became widely available. It became usual for universities and large businesses to have computers, and much more powerful ones, such as the IBM 360 introduced in 1964, became available. Perhaps even more important for advances in engineering were the so-called minicomputers, such as Digital Equipment Corporation's PDP-8, which first appeared in the 1960s.[22] These machines had the speed of mainframes (though a shorter word-length), but were smaller and much less expensive. This permitted an epoch-making change: for the first time ever, many researchers could have their own computers.[23]

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[19]Fumitada Itakura and Shuzo Saito, "A statistical method for estimation of speech spectral density and formant frequencies" (*Electronics and Communications in Japan*, vol. 53-A (1970), pp. 36–43). The first public presentation of the PARCOR invention was on 21 July 1969 at a meeting of the Technical Committee on Speech of the Acoustical Society of Japan [Itakura personal communication 17 February 1998]. In 1986 the IEEE Morris N. Liebmann Award was given to Atal and Itakura for the invention of LPC.

[20]Manfred R. Schroeder, "Linear prediction, entropy and signal analysis" (*IEEE ASSP Magazine*, vol. 1 (1984), no. 4, pp. 3–13). Maximum entropy spectral analysis came to be applied in a great many areas, such as to determine the periodicity of the polarity reversals of the earth's magnetic field (1972), to produce radio-brightness maps (1977), and to improve the angular resolution of microwave antenna arrays (1980) [Thomas Kailath, ed., *Modern Signal Processing* (Washington, DC: Hemisphere Publishing, 1985), pp. 153–154].

[21]Thomas Kailath personal communication 20 January 1998. Kailath pointed out that in fitting the autoregressive models to data certain equations arose that were solved using the Levinson algorithms for prediction theory mentioned by Enders Robinson in the oral-history excerpt in Chapter 2.

[22]The term 'mainframe' dates from 1964, 'minicomputer' from 1968. The DEC PDP-1, introduced in 1960, is usually considered, retrospectively, the first minicomputer. The extremely popular PDP-8 was introduced in 1965.

[23]Martin Campbell-Kelly and William Aspray, *Computer: A History of the Information Machine* (New York: Basic Books, 1996), pp. 222–226.

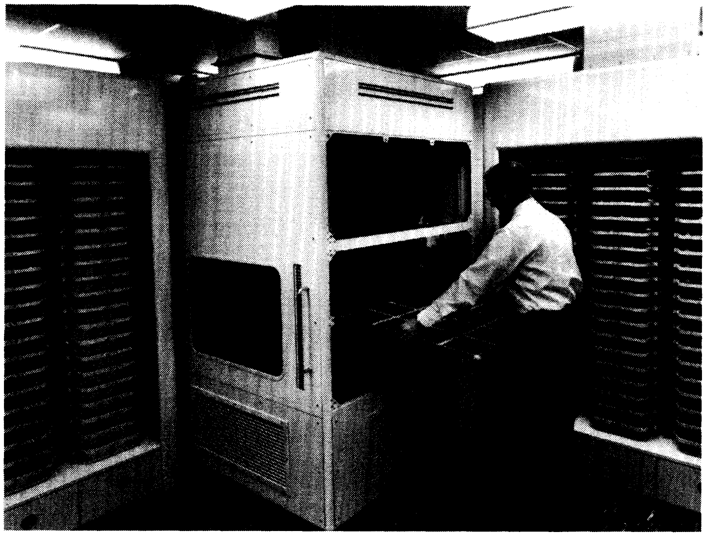

FIGURE 2. This is not a pizza oven but the Lincoln Lab TX-2 computer, which Ben Gold and Charles Rader used to perform numerical simulations of the performance of analog filters. The photograph shows Don Ellis removing one bit-plane from the 64K-word ferrite-core memory. (Lincoln Lab photo reproduced by permission.)

At the forefront in the use of computers in research was MIT's Lincoln Laboratory, itself the site of much computer development.[24] In 1962 or 1963 Ben Gold and Charles Rader, using Lincoln Lab's TX-2 computer, began to simulate the performance of wave (band-pass) filters numerically.[25] (See Figure 2.) Before then, a wave filter, such as might be used in a vocoder, was built from wires and components in order to be tested. At this stage, Gold and Rader thought of the computer only as a tool to speed up the design process for a device that would be implemented in analog hardware.[26]

[24]Two landmark computers of the 1950s, the Whirlwind and the AN/FSQ-7, were developed in part at Lincoln Lab. Two Lincoln Lab researchers, Kenneth Olsen and Harlan Anderson, left in 1957 to found Digital Equipment Corporation, which changed the computer industry by introducing minicomputers. [Eva C. Freeman, ed., *MIT Lincoln Laboratory: Technology in the National Interest* (Lexington, MA: MIT Lincoln Laboratory, 1995), pp. 221–222.]

[25]Charles Rader personal communication 2 January 1998.

[26]Charles Rader oral-history interview 27 February 1997, pp. 3–4.

Ben Gold: One of the reasons that we were doing radar work was that the funding for speech work had dried up, and that was one of the reasons directors ordered us to do radar work, "because we can give you money for that." All of a sudden LPC came along, just another bombshell.

Interviewer: Tell me when.

Gold: I'd say very late sixties, early seventies, probably going into up to the mid-seventies. In any case, we jumped into that pretty quickly. We had a fellow named Ed Hofstetter who was actually an old microwave person. He wasn't that old, but he came from microwave. He got interested in speech, and got interested in programming, something he had never done, and he got very good at it. He was also a good mathematician. When LPC came along, he was one of the first to really pick up on it and understand it. He said, "I could write a real time program using the FDP." At that time nobody could do real-time computation on a general purpose computer to do LPC.

Interviewer: The FDP?

Gold: The Fast Digital Processor. It was a fast computer. He actually programmed the first real-time LPC on the FDP. So that got us back into the speech area, and we actually did quite a bit of work on LPC.

Interviewer: Why did that get you back into the speech area?

Gold: Well, because LPC caused the funding switch to open again, and we got money, and were able to work on stuff.[1]

Thomas Huang: [In about 1960] image processing was so primitive then that we [the Image Processing Group at MIT, headed by Bill Schreiber] had to be concerned with equipment as well. We had to build our own scanner for digitizing images and reproducing them. We built one of the first image scanners in our lab using a CRT. I was using a Lincoln Lab prototype computer called the TX-0, probably the first transistorized computer. I had to program in assembly language. After digitizing each image we had to store the data on paper tape with punched holes. We fed that into the computer and then had to write the result on paper tape again. I remember an image of just 240 x 240 pixels took three rolls of tape.[2]

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[1]Ben Gold oral-history interview 15 March 1997, p. 27.

[2]Thomas S. Huang oral-history interview 20 March 1997, p. 3.

The 1960s was a time when numerical simulation was coming into use in many areas of science and engineering.[27] Not only Gold and Rader, but others in the speech research community made early use of computers. For example, in 1964 A. Michael Noll used computer simulation to show that John Tukey's concept of the cepstrum (the Fourier transform of the logarithm of the amplitude spectrum) was well suited to solving the problem of pitch-extraction in a vocoder,[28] and James L. Flanagan's classic *Speech Analysis: Synthesis and Perception*, published in 1965, describes many computer simulations.[29] This was part of a general trend toward calculational approaches to science and engineering problems, especially numerical experimentation, which is the study of phenomena or systems by numerical simulation rather than by observation or manipulation of physical systems.

A circuit or device may be described by its transfer function, the mathematical relation of input to output. With analog circuits, composed of resistors, capacitors, and so on, only certain transfer functions are practical, while a digital device incorporating a computer could realize almost any conceivable transfer function.[30] Yet before the 1960s few engineers even considered digital circuits for signal processing because computers were hardly available, they were extremely expensive, and they were not fast enough to do real-time signal processing.[31]

As computers became widely used for simulation in the 1960s, there arose the idea, in a number of places, that the signal processing itself might be done by computer. As early as 1961 a computer was built specifically for digital signal processing. This was the TI187 transistorized computer built by Texas Instruments for analyzing seismic data, though not in real time.[32] At Lincoln Lab, Gold and Rader used a computer as a vocoder, but the computer needed 10 seconds to process 1 second of speech.[33] Also in the

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[27]Frederik Nebeker, *Calculating the Weather: Meteorology in the 20th Century* (New York: Academic Press, 1995), pp. 177–183.

[28]Millman *op. cit.*, p. 113.

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[29]James F. Flanagan, *Speech Analysis: Synthesis and Perception* (Berlin: Springer-Verlag, 1965).

[30]Thomas Kailath recalls that Robert Price at Lincoln Lab used to make this point in the early 1960s [personal communication 20 January 1998]. As mentioned below, Price did digital filtering (in the modern sense) around 1960 in his analysis of radar signals from Venus.

[31]Kailath *op. cit.*, pp. xi, 370.

[32]Harvey G. Cragon, "The early days of the TMS320 family" (*Texas Instruments Technical Journal*, vol. 13 (1996), no. 2, pp. 17–26). Even earlier, in 1955, J. Fred Bucy and others at Texas Instruments designed a hybrid analog-digital computer for seismic data reduction [Cragon *op. cit.*].

[33]Freeman *op. cit.*, p. 227.

FUMITADA ITAKURA: Suppose that we sample that linear predictive parameter every twenty milliseconds, fifty times a second, and suppose that we have ten parameters, and each parameter is quantized with ten bits. So for each frame we need 100 bits. If we multiply by fifty, that is five kilobits per second just for the linear predictive parameters. That is too much for vocoding. So we have to find a better method of quantizing linear predictive parameters. I tried to reduce the number of quantizing parameters of LPC parameters, but it was not perfect. I had to think a little more in detail. I went to the new concept of partial autocorrelation, PARCOR. By using those parameters we could quantize parameters to any number of bits. Of course, there might be some degradation of speech resulting in instability.

INTERVIEWER: Was this your own conception, this partial autocorrelation?

ITAKURA: It is a well-known statistical concept.

INTERVIEWER: That was the first time it was applied to the speech problem?

ITAKURA: Right.

INTERVIEWER: This was what you applied for a patent for in May 1969, and then in July you presented the PARCOR vocoder at the special group meeting of the Acoustical Society of Japan.

ITAKURA: Yes, that was a very special day.

INTERVIEWER: The same day as the moon landing.

ITAKURA: Yes. The interesting thing about that is that in Japan we have regular meetings, so every eight months speech scientists and engineers get together to talk and encourage the technology and science of speech. It is informal, with one person designated to talk. Usually twenty or even forty people get together, but that was the day of the more interesting moon landing. Very few attended: the chairman of the committee, myself, Saito, and a few other researchers very close to the work, but they kept coming in and going out. So it was essentially only three people.[1]

[1]Fumitada Itakura oral-history interview 22 April 1997, pp. 16–17.

early 1960s, James Kaiser and Roger M. Golden at Bell Labs began work to transfer "the extensive continuous filter art of the electrical engineer" from the analog to the digital domain.[34]

[34]James F. Kaiser, "Digital filters" (Chapter 7 of *System Analysis by Digital Computer*, Franklin F. Kuo and James F. Kaiser, eds. (New York: John Wiley, 1966), pp. 218–285). What may have been the first hardware implementation of digital filters was at Bell Labs in 1966 and 1967; this work is described in 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).

Though 'filter' originally meant a device to select certain frequencies or ranges of frequencies from a signal, 'digital filter' soon acquired a very general definition: "a discrete time system which operates on an input sequence to produce an output sequence according to some computational algorithm".[35] There emerged a field of digital-filter theory, which drew on the extensive earlier work done in classic circuit theory, numerical analysis, and more recently sampled-data systems.[36] In the United States, James Kaiser, Enders Robinson, Sven Treitel, Ken Steiglitz, Ben Gold, Charles Rader, Alan Oppenheim, Lawrence Rabiner, and Thomas Stockham did important work in the 1960s.[37] In Europe, Hans Wilhelm Schuessler, Anthony Constantinides, Vito Cappellini, and Fausto Pellandini were among the early contributors to digital-filter theory.

The rapid advances in computing technology in the 1960s promised to make this theory of practical importance. Mass production of integrated circuits began in 1962, when Fairchild and Texas Instruments sold a few thousand logic chips, and the capabilities of ICs grew rapidly.[38] The availability of software to make it easier to program a computer was very important. At Bell Labs, Kaiser and Golden used BLODI (from 'block diagram') that could translate a block-diagram description of a system into a program that simulated the system; at Lincoln Lab, Rader wrote a block-diagram compiler of his own.[39]

Also in the mid 1960s, the discovery of a particular algorithm, the fast Fourier transform or FFT, incited a great deal more activity. The Fourier

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[35]Bede Liu, ed., *Digital Filters and the Fast Fourier Transform* (Stroudsburg, PA: Dowden, Hutchinson & Ross, 1975), p. 1.

[36]Some important works on sampled-data systems were two textbooks of Eliahu I. Jury, *Sampled-Data Control Systems* (New York: John Wiley and Sons, 1958) and *Theory and Application of the Z-Transform Method* (New York: John Wiley and Sons, 1964), and the article of Rudolf E. Kalman and John E. Bertram "A unified approach to the theory of sampling systems" (*Journal of the Franklin Institute*, vol. 267 (1959), pp. 405–436).

[37]Kailath *op. cit.*, p. xi. A particularly influential publication was Kaiser's chapter on digital filters in the book *System Analysis by Digital Computer* (New York: John Wiley, 1966), edited by Franklin F. Kuo and James F. Kaiser [Hans Wilhelm Schuessler oral-history interview 21 April 1997, p. 34].

[38]Stan Augarten, *State of the Art: A Photographic History of the Integrated Circuit* (New Haven, CT: Ticknor & Fields, 1983), p. 10.

[39]Rader's compiler, called PATSI (Programming Aid To System Investigation), was used by a few other people at Lincoln Lab; Bert Sutherland built a graphical interface for PATSI, so that a user could draw his block diagram and the computer would simulate the system [Charles Rader personal communication 2 January 1998].

transform, named after the French mathematician Jean Baptiste Joseph, Baron de Fourier (1768–1830), is a mathematical procedure to convert a time-domain signal into a frequency-domain signal. (The human ear automatically performs a Fourier-like transform on impinging sound waves, sending signals to the brain that give the intensity of the sound at different frequencies.)[40] It has been extensively used by communication engineers in studying noise, by physicists in solving partial differential equations, and by statisticians in studying statistical distributions. It can simplify the mathematics, and it also often makes the phenomenon of interest easier to understand.[41]

The straightforward calculation of a discrete Fourier transform of N points requires $4N^2$ multiplications. In 1965 James Cooley and John Tukey showed how to do the calculation with just $2N \log_2 N$ multiplications.[42] (See Figures 3 and 4.) Thus, even for a thousand-point transformation, the FFT reduces the calculation required by a factor of 200, and for larger sample sizes the reduction factor is much greater. Rabiner writes "... the algorithm remained a mathematical curiosity to most electrical engineers until an engineering interpretation was given to the procedure by Charlie Rader and Tom Stockham.... [They] constructed a Mason flow graph interpretation of the FFT from which a wide variety of the properties of the FFT, including bit reversal, in-place computation, twiddle factors, $N \log_2 N$ operation count, decomposition ideas etc., became clear."[43]

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[40]This statement is only roughly correct: the standard Fourier transform corresponds fairly closely to a bank of equal bandwidth filters, while the human ear corresponds to a filter bank where filter bandwidths increase with increased center frequencies [Ben Gold personal communication 30 January 1998]. James Kaiser [personal communication 11 February 1998] argues that the ear is more a transient detector than a frequency analyzer.

[41]W. Morven Gentleman and Gordon Sande, "Fast Fourier transforms—for fun and profit" (*AFIPS Proceedings of the 1966 Fall Joint Computer Conference*, vol. 29 (1966), pp. 563–578).

[42]James W. Cooley and John W. Tukey, "An algorithm for the machine calculation of complex Fourier series" (*Mathematics of Computation*, vol. 19 (1965), pp. 297–301). Cooley has authored or co-authored several articles on the re-discovery and acceptance of the FFT: James W. Cooley, Peter A.W. Lewis, and Peter D. Welch, "Historical notes on the fast Fourier transform" (*IEEE Transactions on Audio and Electroacoustics*, vol. 15 (1967), no. 2, pp. 76–84); James W. Cooley, "The re-discovery of the fast Fourier transform algorithm" (*Mikrochimica Acta*, vol. 3 (1987), pp. 33–45); and James W. Cooley, "How the FFT gained acceptance" (*IEEE Signal Processing Magazine*, vol. 9 (1992), no. 1, pp. 10–13). The deep history of the FFT is given in Michael T. Heideman, Don H. Johnson, and C. Sidney Burrus, "Gauss and the history of the fast Fourier transform" (*IEEE ASSP Magazine*, vol. 1 (1984), no. 4, pp. 14–21).

[43]Lawrence R. Rabiner, "The Acoustics, Speech, and Signal Processing Society—A Historical Perspective" (*IEEE ASSP Magazine*, January 1984, pp. 4–10), quotation from pp. 4–5.

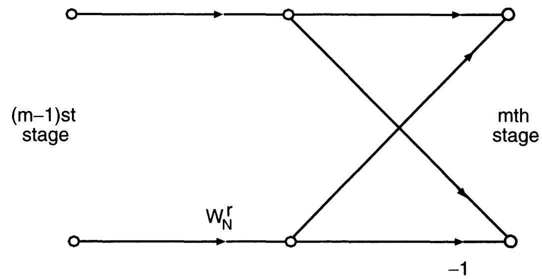

FIGURE 3. In a typical step of the FFT, a pair of values are added or subtracted after being multiplied by predetermined sine or cosine values. This operation may be depicted in the so-called "butterfly structure" shown. (Redrawn after Figure 8 of Gene Frantz and Panos Papamichalis, "Introduction to DSP solutions" (*Texas Instruments Technical Journal*, vol. 13 (1996), no. 2, pp. 5–16).)

The IEEE Group on Audio and Electroacoustics (G-AE), formerly the IEEE Group on Audio, had a major role in the widespread adoption of the FFT. It co-sponsored a special workshop on FFT and spectral analysis in the spring of 1966, and in 1967 papers from this workshop were part of a special issue of the G-AE *Transactions*, which included a classic tutorial paper on the FFT written by members of the G-AE Subcommittee on Measurement Concepts. In 1968 it organized the first of the so-called Arden House Workshops; a hundred researchers exchanged ideas about the FFT, and many of the papers were published in a special issue of the G-AE *Transactions* in 1969.[44]

Researchers developed other algorithms for digital signal processing, such as the Viterbi algorithm in 1967 (used especially in speech recognition),[45] the chirp z-transform algorithm in 1968 (which widened the range of application of the FFT),[46] use of the maximum likelihood principle also in 1968 (for sensor-array signal processing),[47] and adaptive delta modulation in 1970 (for speech encoding).[48]

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[44]Rabiner *op. cit.* The two special issues were the following: Special Issue on Fast Fourier Transforms and Applications to Digital Filtering and Spectral Analysis, *IEEE Transactions on Audio and Electroacoustics*, vol. 15, no. 2 (June 1967), and Special Issue on Fast Fourier Transforms, *IEEE Transactions Audio and Electroacoustics*, vol. 17, no. 2 (June 1969).

[45]Harvey F. Silverman and David P. Morgan, "The application of dynamic programming to connected speech recognition" (*IEEE ASSP Magazine*, vol. 7 (1990), no. 3, pp. 6–25).

[46]Lawrence R. Rabiner, Ronald W. Schafer, and Charles Rader, "The chirp z-transform and its applications" (*Bell System Technical Journal*, vol. 48 (1969), pp. 1249–1292).

[47]Hamid Krim and Mats Viberg, "Two decades of array signal processing research" (*IEEE Signal Processing Magazine*, vol. 13 (1996), no. 4, pp. 67–94).

[48]Millman *op. cit.*, p. 115, and Jerry D. Gibson, "Adaptive prediction for speech encoding" (*IEEE ASSP Magazine*, vol. 1 (1984), no. 3, pp. 12–26).

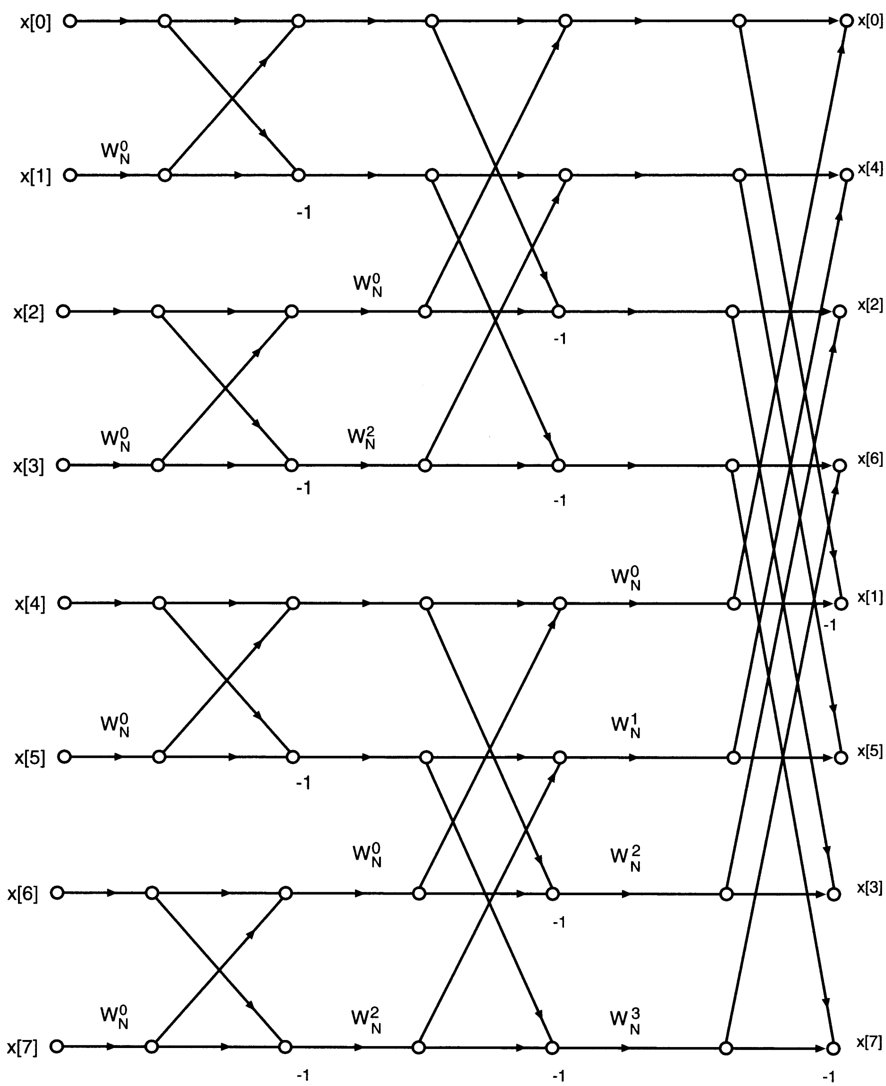

FIGURE 4. This Art Deco design is actually the structure of a 2-point, radix-2, decimation-in-frequency fast Fourier transform. (Redrawn after Figure 7 of Gene Frantz and Panos Papamichalis, "Introduction to DSP solutions" (*Texas Instruments Technical Journal*, vol. 13 (1996), no. 2, pp. 5–16).)

James Kaiser: That got me very interested in signal processing. Now, at the time I arrived at Bell Laboratories [ca. 1960], a change in the means of doing research in the speech area, in the coding area, was under way. Instead of the old way, which was to test an idea about a new way to do things by designing the electronics that embodied that idea and then running tests on that physical embodiment, we were starting to simulate the system—a compression system, an encoding system, whatever—on the general purpose digital computer. Then we would just run a test of the new idea, taking speech and running that through the simulated system and listening to the result. It was much faster and more versatile.

So I got much more interested in how you took continuous systems and got the discrete models necessary for the simulation. With my control background, I knew continuous systems and filter design quite well, and I tried to carry over some of the same ideas to the discrete world. A lot of it carries over as far as the recursive filters are concerned. These design techniques carry over directly via the different transform techniques, the Z transform, the bilinear Z transform, the matched Z transform, and so forth. But one feature of the digital systems is that it's very easy to build finite impulse response digital filters, whereas these are very difficult to build as continuous filters.[10]

James Kaiser: The thing was, there were old timers—especially in industry—that had been designing filters, continuous filters, for years, and all of the sudden these fellows were told, "Now look, I want you to build digital filters." Their reaction was, "I don't know what a digital filter is." They were completely lost and some of these fellows didn't want to learn the new stuff. They said "There's too much new to learn! Don't bother me with the new stuff." So one of my goals was to preserve all the knowledge that those fellows already had and say to them, "Look, all you've got to do is run this little program with that knowledge you already have and it will design a digital filter for you. You know how to do it!" That's basically what the bilinear transformation does. It carries the continuous filter designs over to the discrete design.[11]

Bede Liu: Yes, radar processing, and you also found sonar people interested. That was the group for serious applications [in the 1960s]. It was a serious application because people actually built it to put in systems for that purpose. Speech, I think at that time was mostly in the research stage, but they did have an application goal in that, too.

... I think signal processing is becoming so popular for two reasons. It is application-driven, particularly as seen now, and it is very much tied with technology. So you can identify applications; you have needs to actually build systems to do certain tasks.[12]

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[1]James Kaiser oral-history interview 11 February 1997, pp. 6–7.

[2]James Kaiser oral-history interview 11 February 1997, p. 24.

[3]Bede Liu oral-history interview 10 April 1997, pp. 20–21.

Besides the new digital communications, digital control systems and digital instruments began to be more common. The first commercial digital instruments, introduced in the early 1950s, measured frequency, pulse rate, and time interval.[49] There soon followed digital voltmeters, digital pulse-delay generators, and digital frequency synthesizers.[50] The advantages of digital systems were many: accurate realization, no adjustments, no aging, temperature-independence, reliability, programmability, flexibility, monolithic integration (often the entire circuit on a single chip), and little or no signal degradation in transmission and copying.[51]

But engineers needed to learn about the new possibilities, and here the workshops, conferences, and publications of the IEEE Audio and Electroacoustics Group played an important role. A new field of digital signal processing was emerging, and it received its first textbook formulation in 1969 with Gold and Rader's *Digital Processing of Signals* (with chapters contributed by Stockham and Oppenheim) (see Figure 5).[52] According to Don Johnson, "Their book, more so than the algorithmic developments of the time, pushed digital signal processing into the limelight."[53] The second textbook in the new field, and the first in Europe, appeared in 1973: Hans Wilhelm Schuessler's *Digitale Systeme zur Signalverarbeitung*.[54]

Though speech and other 1-dimensional signals were the concern of most of the pioneers of the new field, image processing came to the fore in a number of ways in the 1960s. It was then that Marshall McLuhan first called attention to the cultural importance of the mass media ("the medium is the message") and argued that society was moving from a "print culture" to a "visual culture". He introduced the concept of the "global village", saying, "Today, after more than a century of electric technology, we have ex-

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[49]Perhaps the first commercial digital instrument was an events-per-unit-time meter offered by Berkeley instruments in 1950; Hewlett-Packard offered a combined frequency- and period-measuring instrument in 1951 [Bernard M. Oliver, "Digital display of measurements in instrumentation" (*Proceedings of the IRE*, vol. 50 (1962), pp. 1170–1172)].

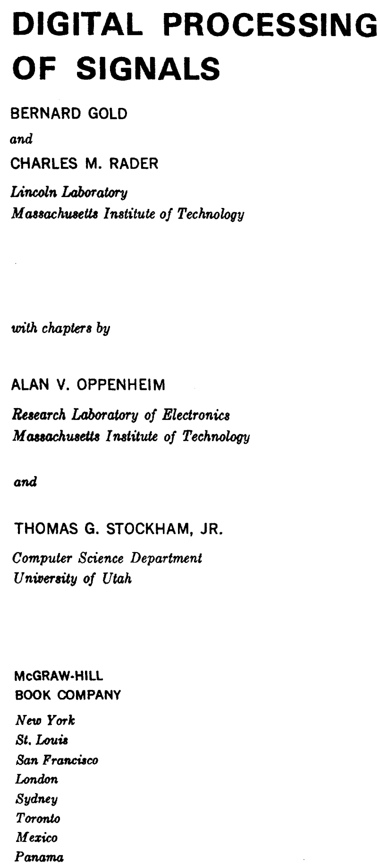
[50]Oliver *op. cit.*

[51]Piet J. Berkhout and Ludwig D.J. Eggermont, "Digital audio systems" (*IEEE ASSP Magazine*, vol. 2 (1985), no. 4, pp. 45–67). There were often, of course, disadvantages such as increased bandwidth for transmission and increased power consumption.

[52]Bernard Gold and Charles M. Rader, with chapters by Alan V. Oppenheim and Thomas G. Stockham, Jr., *Digital Processing of Signals* (New York: McGraw-Hill, 1969).

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

[54]Hans Wilhelm Schuessler, *Digitale Systeme zur Signalverarbeitung* (Berlin: Springer-Verlag, 1973).



DIGITAL PROCESSING OF SIGNALS

BERNARD GOLD

and

CHARLES M. RADER

Lincoln Laboratory
Massachusetts Institute of Technology

with chapters by

ALAN V. OPPENHEIM

Research Laboratory of Electronics
Massachusetts Institute of Technology

and

THOMAS G. STOCKHAM, JR.

Computer Science Department
University of Utah

McGRAW-HILL
BOOK COMPANY

New York
St. Louis
San Francisco
London
Sydney
Toronto
Mexico
Panama

FIGURE 5. This is the title page of Gold and Rader's *Digital Processing of Signals* (1969), which was the first textbook on digital signal processing. (Reproduced by permission of McGraw-Hill.)

tended our central nervous system itself in a global embrace, abolishing both space and time as far as our planet is concerned."[55] For example, in 1963 both Europe and Japan received, via satellite, live television coverage of John F. Kennedy's funeral.[56]

In the 1960s the public was eager for photographs from space: as early as 1959 the Soviet Luna 1 had transmitted back a photograph of the previously unseen far side of the moon; in 1966 the U.S. Surveyor 1 soft-landed

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[55]Marshall McLuhan, *Understanding Media: The Extensions of Man* (Cambridge, MA: MIT Press, 1994 (first published 1964)), quotation from p. 3.

[56]Walter A. McDougall, *… the Heavens and the Earth: A Political History of the Space Age* (New York: Basic Books, 1985), p. 396.

Bede Liu: With floating point arithmetics—without getting into detail—the errors are introduced through a slightly different mechanism. Well, the same mechanism, but in a different way, which makes the analysis very hard. Irwin Sandberg presented a very good paper at the Allerton Conference which pointed out the problem and ways to handle it. The approach he took was a deterministic approach. I guess you can call it classical error-analysis, or the numerical analysis approach.

I found it very interesting. I came back and talked to Toyohisa Kaneko [a graduate student] and said the proper way to look at the problem is through statistical probability. I said, "This is a very interesting paper, but the underlying problem is one of many, many small errors being introduced, and we should be able to analyze the problem using this probabilistic model." We talked about this for a little while, and a few days later Kaneko said, "Yes, everything carries through fine." So we wrote it up and sent the abstract to Allerton again for the '68 conference. One week before the conference he came in and he said, "No, it didn't work." Sure enough, it did not work; we overlooked something. We quickly tried to do a lot of things, and finally I was able to find a way to analyze the problem, which actually made the problem much more interesting. It got the result out. And that's my first work on digital filters.[1]

Hans Georg Musmann: During that time I worked on my thesis [in the early 1960s], I also built up a small computer with transistors—I had to connect the transistors of course by hand! I wanted to learn how computers work and operate. So, together with a colleague, I built up a small computer which could be operated by voice.

Interviewer: How was that done? Did you have any voice recognition techniques?

Musmann: Well, to a certain extent. I developed a very simple technique in which the voice is split up into frequency bands. Then I sampled the output of the frequency bands and used these patterns to distinguish between ten digits. Some commands, like "add in," or "subtract," or "multiply," and so on, were for operating the computer . . . .

Interviewer: Why did you choose to use voice input for that device?

Musmann: I thought it would be very nice to have a computer you can operate with voice. That is what you need still today![2]

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[1]Bede Liu oral-history interview 10 April 1997, pp. 12–13.

[12]Hans Georg Musmann oral-history interview 30 August 1994, pp. 4–5.

on the moon and sent back high-quality photographs of the Ocean of Storms; and later in 1966 the U.S. Lunar Orbiter 1 attracted even greater attention with photographs of the far side of the moon and of the earth from beyond the moon. In 1969 the public marveled at live television of the first steps onto the moon and at detailed pictures telemetered from Mars.[57] These and other ventures into space posed great challenges for image coding, transmission, and reconstruction. Among the early advances in image coding was the block transform method introduced in 1969 by Grant Anderson and Thomas S. Huang,[58] and not long afterwards came the use of the discrete cosine transform for image coding.[59]

In the United States, color television finally caught on, and beginning in 1964 television sets had to have all-channel tuners, that is, for UHF as well as VHF, which gave a great boost to UHF broadcasters.[60] The 1960s saw the beginning, in Japan, of research on high-definition, wide-screen television.[61] Other developments related to image processing were the adoption in 1968 by the International Telecommunications Union of a standard for facsimile apparatus and transmission.[62] At Bell Labs, Leon Harmon studied how much visual information is required for recognition of faces; his quantized portrait of Abraham Lincoln (with only 756 bits of information) became a popular image (Figure 6).[63]

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[57]Better encoding and a higher bit rate (16,000 bits per second rather than 8 bits per second) made the pictures sent by Mariner 6 and 7 in 1969 much more detailed than the pictures sent by Mariner 4 in 1965 [*Britannica Yearbook of Science 1971 op. cit.*, p. 165].

[58]Grant Anderson and Thomas S. Huang, "Picture bandwidth compression by piecewise Fourier transformation" (*IEEE Transactions on Communications*, vol. 19 (1971), pp. 133–140); an earlier version of this paper appeared in the proceedings of the Purdue University Centennial Symposium on Systems and Information Sciences, held 28–30 April 1969. An important early paper on block quantization in general is J.J.Y. Huang and Peter M. Schultheiss, "Block quantization of correlated Gaussian random variables" (*IEEE Transactions on Communications Systems*, vol. 11 (1963), pp. 289–296).

[59]Nasir Ahmed, T. Natarajan, and K.R. Rao, "Discrete cosine transform" (*IEEE Transactions on Computers*, vol. 23 (1974), pp. 90–93).

[60]*The U.S. Consumer Electronics Industry in Review: 94 Edition* (Washington, DC: Electronic Industries Association, 1994), p. 19.

[61]Simon Haykin, *An Introduction to Analog and Digital Communications* (New York: John Wiley, 1989), p. 320.

[62]Henry Petroski, *Invention by Design: How Engineers Get from Thought to Thing* (Cambridge, MA: Harvard University Press, 1996), p. 113.

[63]Lucky *op. cit.*, pp. 345–346. Another Bell Labs researcher, Manfred Schroeder, won first prize at the 1969 Las Vegas Computer Art Competition with a computer-generated image of a human eye composed of a 65,000-letter text (illustrating, one might say, the proverb that one picture is worth a thousand words).

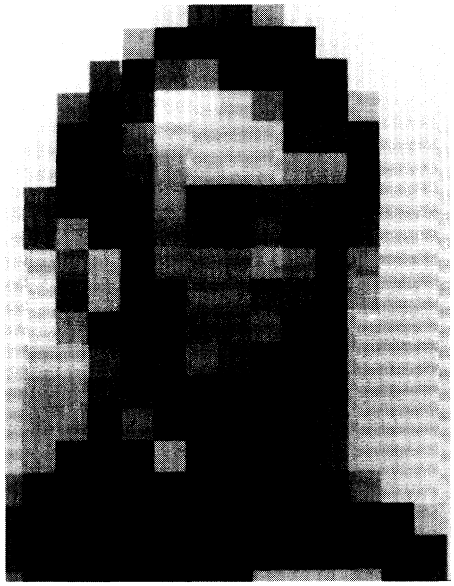

FIGURE 6. Leon Harmon's quantized portrait of Lincoln became a popular image. (Bell Labs photo reproduced by permission.)

In science, imaging technology was advancing rapidly. Computerized tomography or CT scanning (reconstructing a 2- or 3-dimensional image of an object from the data from projections through the object) moved from conception to a practical device in 1971.[64] One of the pioneers of CT scanning, Ronald Bracewell, helped develop another new imaging technology in the late 1960s: very-long-baseline interferometry for high-resolution astronomy and geodesy.[65]

Radar imaging was improved by the application of signal-processing concepts; a milestone in this field was Fred E. Nathanson's 1969 book *Radar*

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[64]Bettyann Holtzmann Kevles, *Naked to the Bone: Medical Imaging in the Twentieth Century* (New Brunswick, NJ: Rutgers University Press, 1997), pp. 145–172. Kevles identifies three pioneers of the technique: Ronald Bracewell, who wished to construct images of the sun and the moon from radio telescope data, and William Ohlendorf and Alan Cormack, both of whom wanted to use x-ray data to form images of parts of the body.

[65]Kenneth L. Kellerman and A. Richard Thompson, "The very-long baseline array" (*Scientific American*, vol. 258 (1988), no. 1, pp. 54–61). The two imaging technologies that Bracewell pioneered—CT-scanning and very-long-baseline interferometry—are entirely different, though he came to both of them from the practice of radio astronomy; very-long-baseline interferometry has had even wider application than has CT-scanning.

Hans Georg Musmann: That was about 1966, 1967 [that I became interested in digital image signals]. Since I was looking especially for representation of visual information, I started with facsimile.

Interviewer: Why were you particularly interested in visual information?

Musmann: I thought, "We have a present type of communication, the telephone; visual information should be the next step in communications." But it took a long time![1]

Alan Oppenheim: When I finished the thesis I decided I was sick of it, and so I thought what I would do is change fields. I would do something different. So what I decided to do was get involved with Ken Stevens and his group on speech stuff. He was very well known in the speech area, and I thought I was going to be taking off in a totally different direction. But within a couple of weeks I realized that speech was the perfect application for deconvolution using homomorphic filtering, because in speech processing deconvolution is a really key aspect. You think of speech as essentially, in an engineering sense, modeled as the convolution of the air flow through the vocal chords and the impulse response in the vocal cavity.

... And then all speech compression systems and lots of speech recognition systems are oriented toward doing this deconvolution and then processing things separately and then going on from there. So speech really became a significant focus for me for applying this homomorphic signal processing and it's become one of the important tools for that.[2]

Alan Oppenheim: ... I also remember that around the time I graduated I was talking to Tom Stockham, and I said, "It's kind of depressing that nobody is picking up on this stuff" and Tom said, "It's not depressing. Actually what's great is that we have this all to ourselves for a while until other people really discover it."[3]

Sidney Millman: [The FFT] revolutionized much of engineering practice, making available many techniques that formerly would have needed prohibitive amounts of computation ....[4]

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[1]Hans Georg Musmann oral-history interview 30 August 1994, p. 8.

[2]Alan Oppenheim oral-history interview 28 February 1997, pp. 9–10.

[3]Alan Oppenheim oral-history interview 28 February 1997, p. 16.

[4]Sidney Millman, ed., *A History of Engineering and Science in the Bell System: Communication Sciences* (1925–1980) (AT&T Bell Telephone Laboratories, 1984), p. 76.

Alan Oppenheim: ... if I was going to sort of identify a viewpoint [of the Oppenheim and Schafer textbook], I would say the following: A traditional way that a lot of people viewed digital signal processing was as an approximation to analog signal processing. In analog signal processing; the math involves derivatives and integrals; you can't really do that on the computer, so you have to approximate it. How do you approximate the integral? How do you approximate the derivative? The viewpoint that I took in the course was you start from the beginning, recognizing that you're talking about processing discrete time signals, and that where they come from is a separate issue but the mathematics for that is not an approximation to anything. Once you have things that are discrete time, then there are things that you do with them. There are examples that I used in the course right from the beginning that clearly showed that if you took the other view it cost you a lot, that you would do silly things by taking analog systems and trying to approximate them digitally and using that as your digital signal processing. I would say that was a strong component of it.[1]

Lawrence Rabiner: They asked, "How do I do it faster, more accurately, more correctly? How do I actually learn something so I do it better?" They were all driven by specific applications.

... The applications drove us into looking at spectrum analysis, they drove us into looking at filtering techniques, they drove us into looking at interpolation. But once you get in there, it's a whole new world. You ask yourself, "Should I stop at the piece I need to solve that problem and then go back?" because then it's a diversion, "Or is it more important to help establish that field before going back?" I always took the second tack.[2]

[1]Alan Oppenheim oral-history interview 28 February 1997, pp. 25–26.

[2]Lawrence Rabiner oral-history interview 13 November 1996, pp. 10, 30.

Design Principles: Signal Processing and the Environment.[66] Digital filters played a part in analyzing the reflected radar signals from Venus in the early 1960s.[67] To process data from a pulse Doppler radar, Herbert Groginsky and George Works designed and built a hardwired FFT signal processor in 1969.[68]

[66]Simon Haykin, "Radar signal processing" (*IEEE ASSP Magazine*, vol. 2 (1985), no. 2, pp. 2–18).

[67]Robert Price at Lincoln Lab used digital filtering in analyzing the radar signals [Thomas Kailath personal communication 20 January 1998].

[68]Herbert L. Groginsky and George A. Works, "A pipeline fast Fourier transform" (*IEEE Transactions on Computers*, vol. 19 (1970), pp. 1015–1019).

INTERVIEWER: Let me be sure I understand. You're saying now that this digital experimentation was considered as signal processing. I was asking whether people ever spoke of signal processing in the analog domain?

LAWRENCE RABINER: Signal processing became a digital concept.

INTERVIEWER: So people don't talk about it until it's realized digitally?

RABINER: Yes. In fact, the term signal processing is always synonymous with DSP—"digital signal processing." You never heard the term ASP—"analog signal processing." It grew out of the term simulation. Simulation itself is a digital embodiment of an analog process, that's the whole concept. After a while you realized you could have a digital simulation of a digital process. What are you really doing then? You're just taking a signal and you're just processing it.

INTERVIEWER: So prior to that, what we would now call signal processing techniques were imbedded in the specific technologies, the specific applications in which they were realized, and never pulled together as a general or universal field of study.

RABINER: That's my perception of it, for the most part. I took a course at MIT on filter design. They called it Filter Design. They didn't call it Analog Filter Design. It was RLCs and passive networks. No one thought of calling that analog signal processing.[1]

CHARLES RADER: Ben [Gold] had been working with the TX-2 computer, which was a very interesting machine in the history of computing. If you look at the history of computing and computers as a kind of a map or a tree, where every computer was the predecessor to one or several others, and so on, right up the main trunk of that tree you'll find the TX-2 computer, and then it branches from there. But, it was for its day, which was 1961—and of course it was several years old then—quite an impressive machine. It had a significantly large memory, about 64,000 words. They were 36 bit words. Actually 38, but some of those bits were parity bits. It had built-in hardware index registers, and a thin film memory, and a whole lot of other nice features. One of its most unusual characteristics was that it was accessible to the user. You could attach equipment to it, and you accessed it directly rather than submitting your punched card deck to a cabal of operators.[2]

[1]Lawrence Rabiner oral-history interview 13 November 1996, pp. 8–9.

[2]Charles Rader oral-history interview 27 February 1997, p. 3.

Charles Rader: But the major thing is that if you wanted to try something out that involved changing a filter, changing the bandwidths, changing center frequency, etc., you had to build another one. That took a few weeks at best. So speech research, in effect, was being hampered by the need to actually build the hardware to try it out. And one of the thoughts that we had was if we could simulate the vocoder, instead of building it, it would speed the pace of research, because we thought, "Gee, programming things doesn't take any time." Nowadays people understand that that isn't true.[1]

Charles Rader: The butterfly diagram was a huge breakthrough. It enabled us to understand fast Fourier transform algorithms. It enabled us to explain it to other people. And, I suppose, with a bit of a messianic streak, we did. We organized a talk in which we both explained what we understood about the algorithm.[2]

Charles Rader: We began to have the idea that we should publish. We wrote a paper called "Digital Filter Design Techniques in Frequency Domain," which was published in *IEEE Proceedings*. We also organized, along with Al Oppenheim and Tom Stockham, a two-week summer course at MIT on this new field of digital signal processing. That was perhaps a turning point in my career, because we prepared extensive notes for the course. We had a thick stack of notes, and we realized this could be the basis of a book. So we got permission from the Laboratory, and we wrote the book. We thought it was just going to be a simple matter of transferring the notes. It actually took about two years to go from the notes to a book. The course was '67. The book came out in '69.

It was almost the first book to cover any of the material, with one very significant exception. There was a book by a group of people from Bell Labs called *Systems Analysis by Digital Computer*. The major authors were Jim Kaiser and Frank Kuo. But it had a chapter on digital filters that Jim Kaiser had authored. That chapter, in my opinion, deserves to be called the first book that introduced any of this material.[3]

Charles Rader: In 1969, I thought digital signal processing theory had gone as far as it was going to go for a while, and I changed fields completely. Talk about a bad decision. I changed into a group that was doing communications satellite work, and I became an assistant group leader of what was then called Group 69. And I worked on a couple of communication satellites that were launched in 1975, and are still working.[4]

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[1]Charles Rader oral-history interview 27 February 1997, p. 4.

[2]Charles Rader oral-history interview 27 February 1997, p. 10.

[3]Charles Rader oral-history interview 27 February 1997, pp. 11–12.

[4]Charles Rader oral-history interview 27 February 1997, p. 47.

Interviewer: How did the digital revolution in signal processing effect the oil exploration business?

Enders Robinson: The thing that converted electrical engineering to digital was the Fast Fourier Transform. Tukey had been looking for that for many years—it wasn't just something he did instantly, he had been working on that problem. Somehow he had this vision that we needed that Fast Fourier Transform, and we did because convolution takes a lot of computer time in the time domain, but if you use the Fast Fourier Transform, it's much faster. A lot of operations were possible with the Fast Fourier Transform. Tukey knew that, worked on it, and published it with Cooley in 1965. I think as soon as he did that the electrical engineers went digital. That was a big transition in electrical engineering. The oil company people had switched a few years before because they could see reflections in marine records by deconvolution.[1]

Manfred Schroeder: In 1964 or '65 we were beginning to buy smaller computers for online research. Before that we used mainframe computers doing batch processing, machines from IBM, General Electric, and so forth. Then we began using small machines from DEC and Honeywell for online speech and hearing research. Much of our batch processing was shifted to the smaller machines that we owned.

Incidentally, we had a hard time getting these minicomputers. Bill Baker said, "Look Manny, you know we spent all that money on the big machines, and now you want all these smaller machines." I said, "But this is a completely different thing." We demonstrated for the management the online use of these small computers for speech research or whatever. I still remember Bill Baker saying "Manny, we are grateful for your having insisted." That is, he saw that you needed big computers and small computers. This was in 1964. We were probably pioneers in the use of small computers.[2]

[1]Enders Robinson oral-history interview 6 March 1997, pp. 41–42.

[2]Manfred Schroeder oral-history interview 2 August 1994, p. 55.

The field of signal processing was, even in the 1960s, wider still. There were studies of sound propagation in the ocean and of hydrophone arrays.[69] Digital filters were used with missile data.[70] A wide range of applications, often involving statistical, nonstationary, and multichannel data, served to broaden the range of theory and techniques brought into the field

[69]Millman *op. cit.*, pp. 120–121.

[70]J.F.A. Ormsby, "Design of numerical filters with applications to missile data processing" (*Journal of the* ACM, vol. 8 (1961), pp. 440–466).

MANFRED SCHROEDER: That day in 1967 I was pacing up and down my office [in the presence of Bishnu Atal], saying "We have to do something about the vocoder speech quality." We had solved the pitch problem with the cepstrum method, but it still didn't sound right like a real human voice. The trouble in a vocoder was that we went on the basis of a fixed and inflexible paradigm. We needed to code speech so as to leave room for error. From this conversation with Bishnu evolved the idea of predictive coding.

The idea is the following. As speech is being encoded, continuously predict, on the basis of preceding samples, the subsequent speech; compare the prediction with the actual speech; and transmit the prediction error, which is called prediction residual. In decoding, use of the same prediction algorithm and knowledge of the prediction residual allow accurate reconstruction of the actual speech.

Well, that idea, which we called adaptive predictive coding, turned out to be absolutely marvelous. The idea was already in existence for television picture coding, but those coders were fixed. With speech we needed predictors that changed with every speech sound, and to make that clear we called this adaptive predictive coding, APC. We wrote a paper for the *Bell System Technical Journal* and presented the technique at Wescon, a major IEEE conference, in 1967. Some Japanese also discovered it, but happily somewhat later.[1]

HANS SCHUESSLER: The essential point was that the first people who used digital signal processing had been those who had to transmit speech.

INTERVIEWER: The speech-processing people?

SCHUESSLER: Yes. I think that was the reason that the people at Bell Labs—this group of Flanagan's—became interested in signal processing as such, since they saw a chance to improve the speech-processing business, to transmit speech with higher efficiency. Really a very important point was that this group—these people under Flanagan, Larry Rabiner especially—had the time and support to push the whole area. They did a lot of ground-breaking work, no doubt about that.[2]

[1]Manfred Schroeder oral-history interview 2 August 1994, p. 61.

[2]Hans Wilhelm Schuessler oral-history interview 21 April 1997, p. 17.

of signal processing. For example, Wiener filtering theory was extended to solve problems in missile tracking and guidance and in signal detection with radar, sonar, and multipath communications.[71] Another example is

[71]See, for example, Robert Price, "Optimum detection of random signals in noise, with application to scatter-multipath communication I" (*IRE Transactions on Information Theory*, vol. 6 (1956), pp. 125–135), and Robert Price and Paul E. Green, Jr., "A communication technique for multipath channels" (*Proceedings of the IRE*, vol. 46 (1958), pp. 555–570).

Hans Schuessler: But back in Germany in '64 we published a paper on the calculation of the impulse response of continuous networks by using the z-transformation. And that's just the type of work which has been done more or less at the same time at Bell Labs by Jim Kaiser. At that time they had a by far larger program to simulate continuous systems on a digital computer, calculating the impulse response and all that. But we were following at least the same idea. And what we did was just—you see it here—starting with a cascade connection of differences, we transformed this continuous system of second order or so into a digital one. And then we are in what we call now the IIR domain. And all these things have been done using the z-transformation and we ended up with transfer functions in the z-domain.[1]

Interviewer: Can we talk about how you got involved with adaptive filtering?

Bernard Widrow: ... I was thinking of trying to build a machine that would be brain-like, but it was just going to take too long. I'd had to do something that would have some practical value, but I still wanted to work in that field. So I got an idea, going back to sample data systems, of what today we would call a digital filter. A digital filter has parameters, it has coefficients. I got the idea to develop a mechanism that would vary the coefficients from the filter so that the filter could improve itself and become a better filter. In a sense a filter can learn to improve itself by acting as a filter, and I figured out a way to do this. I didn't have a brain, but I had a filter. I did not have a brain learning concepts and learning sophisticated things; I had a filter learning very simple things. But it was learning. It was something we could do even in those days, and it was something practical and something useful and something good for engineering, and that was my business.

... The biggest application that exists today is the modem in your computer. Almost all modems in the world use several adaptive filters, and most of them are built as FIR filters, and they filter the signal coming from the telephone line.

... We were using adaptive filters for another purpose. We were making adaptive antennas. We published the first paper on adaptive antennas in 1967 in the *Proceedings of the IEEE*. I think it was December 1967. That paper turned out to be a citation classic, yet it came very close to getting rejected. Three reviewers reviewed it. One reviewer said it's okay to publish; another reviewer had a lot of trouble with it, but he said with some fixing it might be publishable—he was sort of on the fence; and the third reviewer really slammed it. He said, "This is just a bad idea. This is not going to work."[2]

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[1] Hans Wilhelm Schuessler oral-history interview 21 April 1997, p. 10.

[2] Bernard Widrow oral-history interview 14 March 1997, pp. 30, 47, 51.

Bernard Widrow: I was just busy with adaptive filters [in the late 1950s]. So I continued at Lincoln Labs doing work on adaptive filtering, and with the computers that we had you could simulate an adaptive filter. I knew the computers were going to be better, that they were going to get faster, but I wasn't thinking about cheaper. There was just no way at that time one could anticipate what would happen, that you were going to have a personal computer, or that you can have a signal processing chip—you can have a chip that costs a few bucks and you can do adaptive digital programming with it. Who would ever imagine a thing like that? But that wasn't really what was driving us. It was an intellectual interest. And in my mind, it was the beginning.[1]

James Cooley: This [the activity of the speech and signal processing people at MIT] was the first really impressive evidence to me of the importance of the FFT.[2]

Gene Frantz and Panos Papamichalis: The FFT has probably been the most widely used tool in signal processing and its development in the mid-60's really started the DSP revolution.[3]

Jack Deller: Try to imagine the signal processing profession without the FFT. Not easy is it? ... In a little more than $N \log N$ time, the FFT has helped to create a group of specialists who, without it, might be, for example, electrical engineers who actually knew something about electricity. To help spawn an entire field of engineering: Now that's hyper![4]

[1]Bernard Widrow oral-history interview 14 March 1997, pp. 39–40.

[2]James W. Cooley, "How the FFT gained acceptance" (*IEEE Signal Processing Magazine,* vol. 9 (1992), no. 1, pp. 10–13), p. 11.

[3]Gene Frantz and Panos Papamichalis, "Introduction to DSP solutions" (*Texas Instruments Technical Journal*, vol. 13 (1996), no. 2, pp. 5–16), p. 9.

[4]*IEEE Signal Processing Magazine*, vol. 9 (1992), no. 1, p. 6.

the extension of Kalman filter theory for handling large or growing amounts of data in control systems and signal-detection systems; in these and other areas time-variant filters were needed.[72] Closed-form solutions often had to

[72]See, for example, Rudolf E. Kalman, "A new approach to linear prediction and filtering problems" (*Journal of Basic Engineering*, vol. 82 (1960), pp. 34–45).

give way to recursive numerical (computer-based) algorithms and time-variant implementations.[73]

Though the hi-fi movement reached its peak in the 1950s, the 1960s were also exciting years for audiophiles. There was the beginning of stereo FM broadcasting in the United States in 1961;[74] the standards for stereo FM, established by the National Stereophonic Radio Communications Commission, were later adopted by other countries.[75] In 1963 Philips introduced the audio cassette. In 1964 an MIT researcher, Amar G. Bose, founded the Bose Corporation to exploit some of his patents in acoustics and electronics, and the next year the young engineer, Ray Dolby, started another company whose first product was a "S/N (Signal-to-Noise) Stretcher".[76] The year before Robert Moog began marketing a music synthesizer, which was an analog device, and in 1968 Walter Carlos's "Switched On Bach", which used a synthesizer, sold more copies than any classical recording before it.[77] Electret microphones were commercialized by Sony Corporation in 1968.[78] And in 1970 quadraphonic records began to be sold in the United States.

The 1960s saw major changes in the professional organization for signal processing. First of all, on 1 January 1963 the AIEE and the IRE merged to form the Institute of Electrical and Electronics Engineers, and so came into being the IEEE Professional Group on Audio. Second, the Group on Audio expanded its interests, notably in the direction of electroacoustics, speech processing, and the newly recognized field of digital signal processing. There was, thus, a name change in 1965 to Professional Group on Audio and Electroacoustics, and in the late 1960s the creation of technical committees, within the Group, on speech communications and digital signal processing. There were also technical committees on electroacoustics

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[73]See, for example, Thomas Kailath, "The innovations approach to detection and estimation theory" (*Proceedings of the IEEE*, vol. 58 (1970), pp. 680–695).

[74]Mark Kahrs, "Professional and consumer gear: hardware & software" (*IEEE Signal Processing Magazine*, vol. 14 (1997), no. 5, pp. 51–57).

[75]Millman *op. cit.*, p. 125.

[76]*RLE Currents*, vol. 8 (1996), no. 1, pp. 8–9, and *IEEE Spectrum*, vol. 25 (1988), no. 11, p. 57.

[77]Steven Lubar, *Infoculture: The Smithsonian Book of Information Age Inventions* (Boston: Houghton Mifflin, 1993), p. 191.

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

and underwater sound.[79] In this decade engineers who dealt exclusively with audio tended to associate with the Audio Engineering Society rather than the IEEE Group. As before, and since, some aspects of signal processing were concerns of other organizations. For example, the IRE Professional Group on Instrumentation, which evolved into the IEEE Instrumentation and Measurement Society, dealt with "the electrical generation, processing, recording, reproduction and display of data in digital form".[80] Other IRE Professional Groups concerned with signal processing were those for Automatic Control, Circuit Theory, and Information Theory.

For the future of the Group on Audio and Electroacoustics, and for the field of signal processing generally, a most significant event was the embrace by that professional group of the new possibilities opened up, in communications, control, instrumentation, and other areas, by the fast Fourier transform. The person within the Group who did most to bring this about was IBM researcher William Lang, who helped to organize workshops and conferences, to arrange for publications, to attract the speech processing community, and to change the name of the Group.[81] As we will see in later chapters, the Group, which became the IEEE Signal Processing Society, has continued its work in expanding the realm of signal processing.

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[79]Rabiner *op. cit.*

[80]Oliver *op. cit.*, p. 1170.

[81]Rabiner *op. cit.*