

Pictured, from left, are Tom Stockham and Charlie Rader.

# Algorithm Kings

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*The birth of digital signal processing.*

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**T**here will always be signals; they will always need to be processed; and there will always be new technologies and algorithms for implementing the processing. That conviction has given me a feeling of job security ever since I first fell in love with signal processing as an undergraduate student at MIT in the mid-1950s. Added to that is the fact that new mathematical insights and formalisms inevitably change the structure of algorithms and the types of processors used to implement them. The technology and mathematics for signal processing continue to evolve, and clearly the signal processing algorithms and platforms of today would have been viewed as

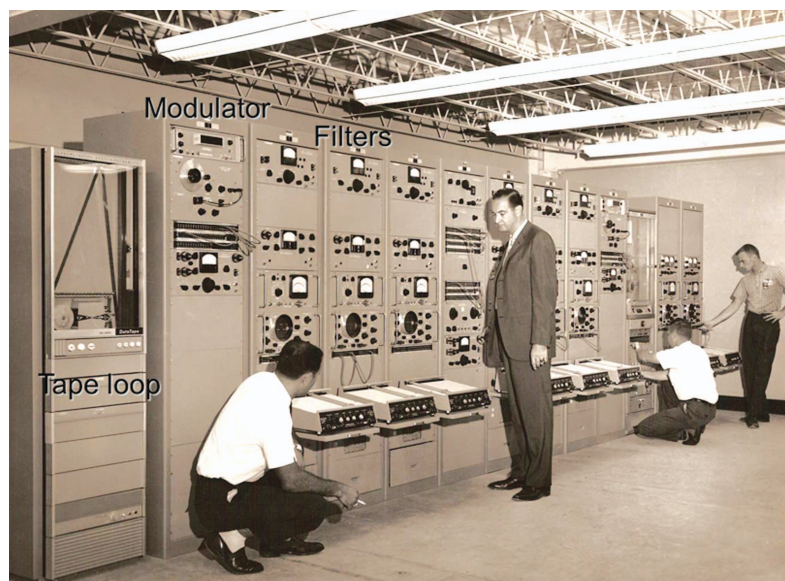
**Before the compact disc, there was another major innovation—the Speak & Spell—that truly launched DSP into the high-volume consumer arena.**

witchcraft during my undergraduate years. Figure 1 is a photograph of a state-of-the-art spectrum analyzer used at Pratt and Whitney in the early 1960s for analyzing jet engine noise. The technology consisted of discrete components, including individually packaged transistors and probably vacuum tubes as well. In contrast, spectrum analyzers today are primarily software-based digital processors that take full advantage of the fast Fourier transform algorithm and other sophisticated signal processing algorithm techniques based on mathematical insights and using implementation technologies that were not available at the time the picture was taken.

Among the pioneers of computer-based signal processing algorithms were Sven Treitel and Enders Robinson, who carried out early research on these methods as part of their work for the Geophysical Analysis Group at MIT. The techniques—implemented digitally off-line on large databases—included filtering, spectral analysis, and parametric signal modeling. At the time, most of these techniques were based on digitally approximating various well-known analog methods. During the same period, analog speech-compression systems (voice coders, or *vocoders*) were being developed and refined at Bell Laboratories by Jim Flanagan and Jim Kaiser and at MIT's Lincoln Laboratory by Ben Gold and Charlie Rader. In that context, the digital computer was typically used to simulate analog filters for channel vocoders (basically filter banks) in order to determine appropriate analog filter parameters. While that work became the basis for how digital recursive filters are now designed for digital implementation, it was only thought of at the time as a simulation tool.

Another context in which digital simulation was being used to design and refine analog signal processing systems was exemplified by the work of Amar Bose and Tom Stockham at MIT in measuring and designing systems to compensate for room acoustics. In this context again, the digital processing was done in non-real time on large mainframe computers and on other digital computer systems such as the TX-0 at MIT and the TX-2 at Lincoln Laboratory, both of which required assembly language programming. The TX-0 was essentially a transistorized—and therefore smaller and faster—version of the Whirlwind computer. It was equipped with a display system, in this case a 12-in oscilloscope hooked to output pins of the processor, allowing it to display 512 by 512 points in a 7 in by 7 in array. An addition on the TX-0 took 10  $\mu$ s, and programming was done in assembly language. The TX-2, which was a spin-off of the TX-0, had a huge amount of ferrite core memory for the time: 64,000 36-bit words. Figure 2 shows the TX-2's "user interface," i.e., the operator console. RAM, drum memory, and logic circuits filled another large area.

Circuit technology continued to evolve. Meanwhile, many of us in the research community were developing algorithms for signal processing that would be difficult or impossible to implement in real time in analog hardware, driven in part by the strong belief that the magic of Moore's law and the development of IC technology would eventually make these algorithms practical. Among the class of algorithms being developed were nonlinear techniques such as homomorphic signal processing and cepstral analysis as well as parametric signal-modeling algorithms of various types.

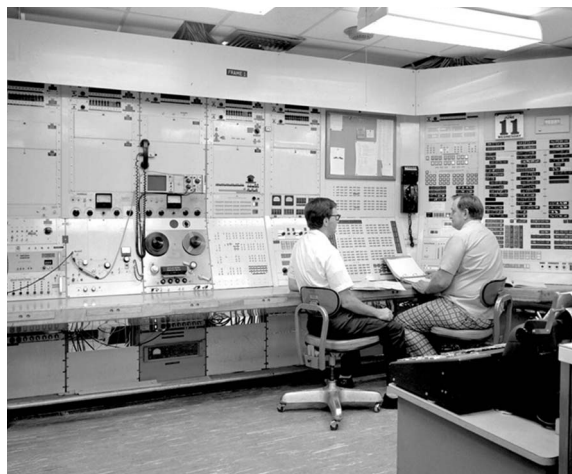


**FIGURE 1:** A state-of-the-art spectrum analyzer used at Pratt and Whitney in the early 1960s for analyzing jet engine noise.



There were many skeptics at the time about the feasibility of doing sophisticated signal processing digitally. Tom Barnwell, one of the pioneers of digital signal processing (DSP), offered the following tongue-in-cheek description of the skeptics' view: "DSP is a discipline that allows us to replace a simple resistor and capacitor with two antialiasing filters, an A-D and D-A converter, and a general-purpose computer or array processor, as long as the signal we are interested in doesn't vary too quickly."

The "big bang" in the digital processing of signals was the publication in 1965 by James Cooley and James Tukey of what is now known as the fast Fourier transform (FFT) algorithm. The excitement generated by this paper led to the first Arden House Workshop in 1968 to discuss and explore its implications. Figure 3 is a picture of Jim Cooley discussing this algorithm at that first Arden House Workshop on DSP. The Cooley-Tukey paper was highly mathematical, and the structure of the algorithm was hard to decipher until it was made widely accessible through a flow-graph interpretation developed by Tom Stockham and Charlie Rader, shown in Figure 4.



**FIGURE 2:** The TX-2 "user interface" at Lincoln Laboratory. (Reprinted with permission of MIT Lincoln Laboratory, Lexington, Massachusetts.)

The FFT was so efficient that it was often faster to do convolution (i.e., linear filtering) by transforming to the frequency domain, multiplying by the desired filter frequency response, and then transforming back to the time domain. This in fact became the procedure eventually used by Stockham in his work with Bose on analyzing and equalizing for room acoustics.

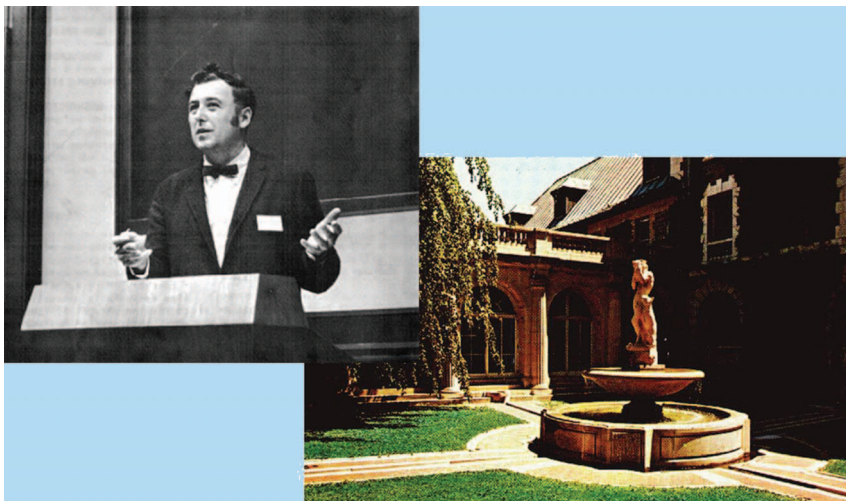
The publication of the Cooley-Tukey paper essentially marked the beginning of the field of DSP in its modern form. At that time, signal-processing courses at universities were focused on continuous-time analog techniques. At MIT in 1965

and again in 1967, Ben Gold taught a course on digital signal processing focusing largely on z transform methods, digital filter design, and the FFT. That course was based primarily on notes by himself and Charlie Rader, which in 1967 were published as a book, with chapters contributed by myself and Tom Stockham. In 1969, returning to my faculty life at MIT from a two-year leave of absence at Lincoln Laboratory, I established what was perhaps the first regular course on DSP, the notes from which resulted in a widely used textbook on

the topic that I wrote along with Ron Schafer. By then, the field was gathering tremendous momentum, and there was a growing conviction that DSP would probably be a practical way of doing real-time signal processing, at least at low bandwidths such as that required for speech analysis. Part of this conviction was tied to a strong belief in the never-ending truth of Moore's law.

As Moore's law continued along its exponential path, innovative research on algorithms for signal processing that could take advantage of the freedom and flexibility afforded by digital technology continued. The importance of the continuing development of innovative algorithms for DSP was underscored in the December 2010 report from the President's Council of Advisors on Science and Technology, which commented that "in many areas, performance gains due to improvements in algorithms have vastly exceeded even the dramatic performance gains due to increased processor speed."

Following the Cooley-Tukey paper, the continuing development of innovative signal processing algorithms led to, among other new applications, the use of DSP for real-time digital audio recording and to digital remastering of analog recordings. In 1975, Tom Stockham, generally acknowledged as the father of digital



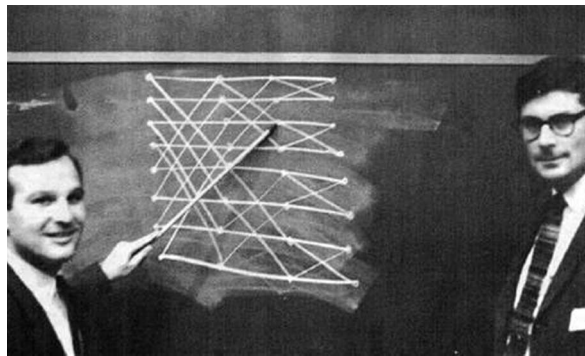
**FIGURE 3:** Jim Cooley discussing the FFT algorithm at Arden House in 1968 and the Arden House courtyard.

audio, founded Soundstream Inc. to develop and commercialize this technology, which was somewhat controversial. In fact, one skeptic was vocal about the “fact” that digital audio was bad for your health (perhaps because of the sharp corners on the ones and zeros?). Unfortunately, Tom was somewhat ahead of his time. It wasn’t until 1985, with the development of the compact disc, that digital audio and the associated signal processing hit the mainstream.

Before the compact disc, there was another major innovation—the Speak & Spell—that truly launched DSP into the high-volume consumer arena. In his article for this issue of *IEEE Solid-State Circuits Magazine*, Gene Frantz describes in some detail the invention and development of the Speak & Spell. In this article, I’d like to describe how it was viewed and the impact that it had from the perspective of those of us teaching DSP and doing research on signal processing algorithms.

In the late 1960s, a class of DSP algorithms for speech analysis, using the technique of linear prediction, was being developed, with a potentially important application being speech coding. This class of techniques, going back to the work of Treitel and Robinson and further developed in the context of speech processing by Bishnu Atal, Manfred Schroeder, and others, became widely referred to as “linear predictive coding,” or LPC. The analysis technique became very well understood, and the associated speech synthesis used digital filters implemented in what is referred to as a “lattice structure.”

The U.S. Department of Defense chose this speech-coding structure as the preferred method of secure speech transmission and started a relatively large project, spanning several years, to develop the standards and technology for real-time implementation of LPC. This was at



**FIGURE 4:** Tom Stockham and Charlie Rader presenting their flow-graph interpretation of the FFT algorithm. (Reprinted with permission of MIT Lincoln Laboratory, Lexington, Massachusetts.)

about the time that Gene Frantz was “toying” with the idea of the Speak & Spell. In the design of the Speak & Spell, Frantz and his team used the well-established speech analysis technique of LPC to code the speech and lattice filters for the synthesizer. In designing and implementing the chips for the Speak & Spell, the technology used was relatively “off the shelf,” i.e., the Speak & Spell was not pushing the state of the art in either IC technology or algorithm development.

What astonished many of us in the DSP community was that apparently these algorithms were commercially viable enough to warrant the considerable investment in designing

in multimedia, home entertainment, cell phones, CD and MP3 players, and many other devices. This pioneering electronic toy also garnered a great deal of public attention for the field of DSP itself.

#### About the Author

**Alan V. Oppenheim** received S.B. and S.M. degrees in 1961 and an Sc.D. degree in 1964, all in electrical engineering, from MIT. In 1995, he was the

recipient of an honorary doctorate from Tel Aviv University. He is currently Ford Professor of Engineering at MIT. In 2009, he also became an honorary professor at Xi’an Jiaotong University, in Shaanxi, China. He is a coauthor of the textbooks *Discrete-Time Signal Processing* and *Signals and Systems* and the editor of several advanced books on signal processing. He is a member of the National Academy of Engineering and a Life Fellow of the IEEE. He has received a number of awards for outstanding research and teaching, including the IEEE Jack S. Kilby Signal Processing Medal, the IEEE Education Medal, the IEEE Signal Processing Society’s Signal Processing Education Award,

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special-purpose chips to implement them so that they could be incorporated into low-cost, high-volume educational toys. In a sense, this was another “big bang” (or maybe an aftershock) that catapulted DSP into its current position as a pervasive technology. In any case, the Speak & Spell served to launch DSP into the world of consumer electronics, where it now plays an essential role

the IEEE Centennial and Millennium awards, and the IEEE’s Acoustics, Speech, and Signal Processing Society Award, Technical Achievement Award, and Senior Award. He has also received several awards at MIT for excellence in teaching, including the Bose Award, the Everett Moore Baker Award, and recognition as a MacVicar Fellow.

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