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The Research Laboratory of Electronics at the Massachusetts Institute of Technology

DIGITAL SIGNAL PROCESSING AT RLE:
Beyond the Realm of the Compact Disc

For audiophiles, perhaps the most commercially visible application of modern digital signal processing is the highly acclaimed compact disc. Digital information is stored in microscopic bumps and smooth areas on the surface of an aluminized plastic disc. The disc itself measures only 120 millimeters wide and 1.2 millimeters thick, and its 500-megabyte capacity can store up to 74 minutes a side. The laser beam mechanism contained in the compact disc player is equivalent to the conventional record player stylus. Behind the technology used to produce these superior recordings are the digital signal processing techniques developed by scientists to dramatically enhance sound. But these techniques are not just limited to acoustic applications in the home entertainment market. What is going on behind the advances that have become so familiar to us as today's high-tech consumers?

Digital signal processing is the manipulation of digitally represented signals by means of combining sophisticated algorithms implemented in modern integrated-circuit digital computer technology. By sampling data (from speech or image sources, for example) and applying signal processing procedures (such as spectral analysis or filtering), researchers can estimate and transform a signal's features. Some techniques are used to measure characteristics which suggest further analysis of a specific system. Other techniques, such as filtering, remove noise or recover meaningful data from degraded signals. Once processed into a more workable form, researchers can then study the features, components, or source of a signal.

The Basic Science of Signals

Signals can be represented in either analog or digital form. Analog signals

(continued on pg. 2)

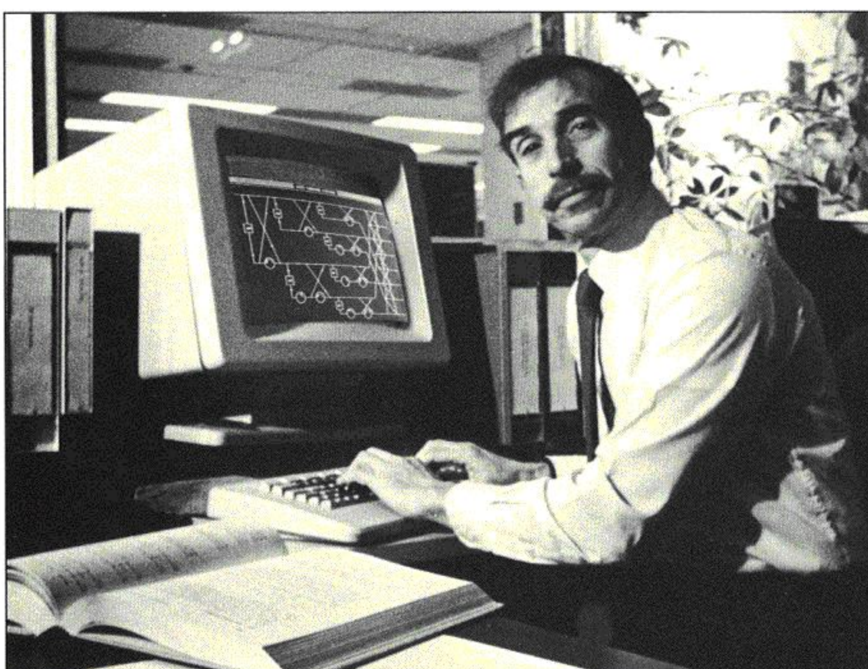


Photo by John F. Cook

One goal of the research activities in RLE's Digital Signal Processing Group is the development of an algorithm design environment based on the symbolic rearrangement of signal processing expressions. The current version of a system, demonstrated above by Professor Alan V. Oppenheim, has been designed and implemented on a LISP machine by graduate student Michele M. Covell as part of her doctoral thesis.

Director's Message

Building on long-standing expertise in signal processing, an increasing experience with contemporary digital computers, and strong ties to industrial research labs and MIT Lincoln Laboratory, it was natural for RLE investigators to assume a central role in the early and continuing evolution of digital signal processing. Although the digital computers of the '60s were not very powerful by today's standards, RLE researchers were quick to exploit their capabilities, and to benefit from the relative ease of experimenting with a variety of signal processing systems in many application areas. Without the need to build analog signal processing systems from physical components, systems could be readily simulated. Problems of component tolerance and aging, as well as parametric variations with temperature and voltage, disappeared and new skills for programming signal processing algorithms were needed. Spurred by the burgeoning digital electronic technology, new algorithms in efficient forms found many uses, and digital processing became an essential part of an electrical engineer's training.

Since the beginning of digital signal processing research, algorithms and implementation architectures have grown in a synergistic way. Fruitful connections with computer science and modern control theory have expanded the rich set of techniques and capabilities, and there has been steady growth in a wide variety of applications. Spectral estimation, vocoding, image processing, seismic and underwater signal processing, knowledge-based signal processing, signal recovery from degraded forms, fault-tolerant signal processing architectures, and innovative, highly parallel computing systems are some of the areas where there has been intense research activity and correspondingly significant results. The ambient computing technology has provided not only the high-performance framework for new and innovative algorithms, but has also led to the consideration of new design techniques that efficiently explore the large space of design alternatives afforded by space-time trade-offs.

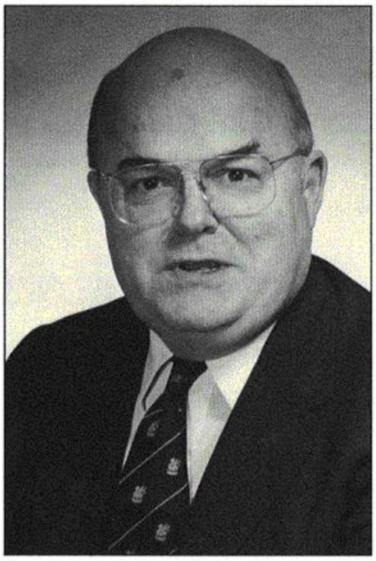


Photo by John F. Cook

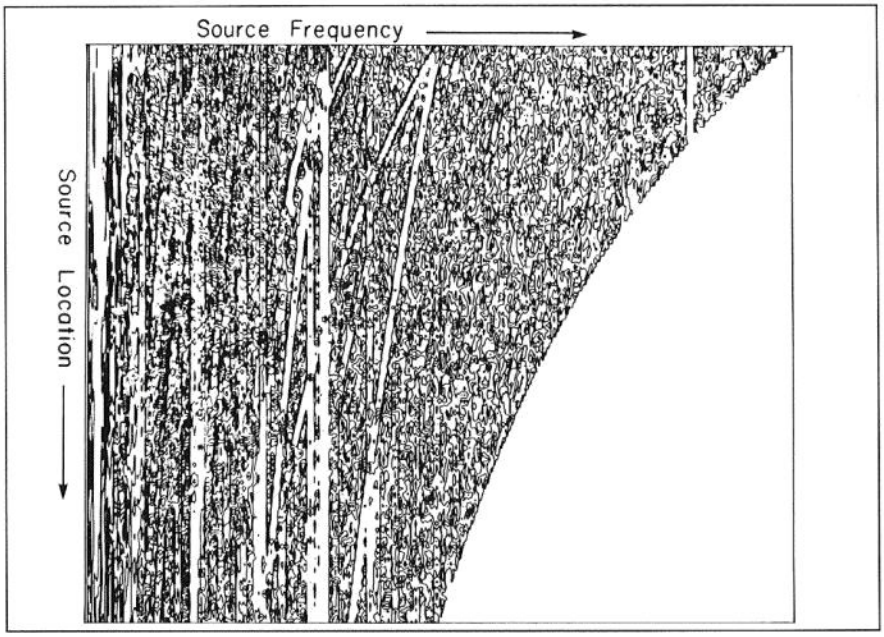
Professor Jonathan Allen, Director Research Laboratory of Electronics

RLE is proud of its outstanding expertise in the digital signal processing area, and the increasing impact that our research is having on many ubiquitously found systems. We look forward to new and powerful results in the future, and an increasing presence of these techniques throughout the Laboratory's activities.

are continuous and can be illustrated as a curve which indicates the amplitude of a signal as a function of a continuous, independent variable such as time. Some analog signals repeat with a certain frequency, measured in hertz, and each repeated cycle has a duration, known as its period. Analog signals arise from a wide variety of natural and man-made sources, such as earthquakes (seismic) or radio broadcast transmitters (electromagnetic). Many analog signals are essentially bandlimited, that is, all their energy is contained below a certain cut-off frequency. In these cases, all of the signal's information can be represented by a sequence of numbers obtained by sampling it at a frequency that is at or above twice its highest frequency. This is a consequence of the fundamental sampling theorem that allows scientists to easily make transformations between the analog and digital domains.

When engineers and scientists sample analog signals such as speech, they must first transform the analog signals into a series of corresponding numbers which accurately represent the shape of the analog waveform. The sampling process quantizes the continuous analog signal values into a limited set of discrete binary numbers, each of which comprises a string of *bi*nary dig*its*, or *bits*. Once an analog signal is converted into the digital domain, computers can analyze the frequency spectra and other characteristics of a signal by using a wide variety of digital signal processing algorithms.

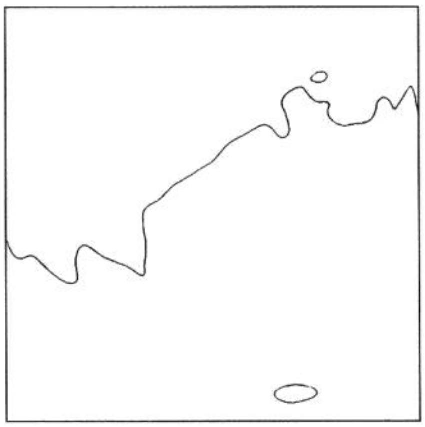
Algorithms are effective procedures, usually represented as computer programs, which are used to solve signal processing problems such as digital filtering, correlation, and signal parameter estimation. Many applications require the input sampled sequence to be transformed to the frequency domain, and this is accomplished through the use of the discrete Fourier transform (DFT) algorithm. In the early history of digital signal processing, the computational expense of the DFT was enormous, but in 1965, James Cooley and John Tukey invented the fast Fourier transform (FFT) algorithm, which computed the same output values as the DFT but at a much faster rate. Interestingly, the FFT can be divided into many smaller processes that can be



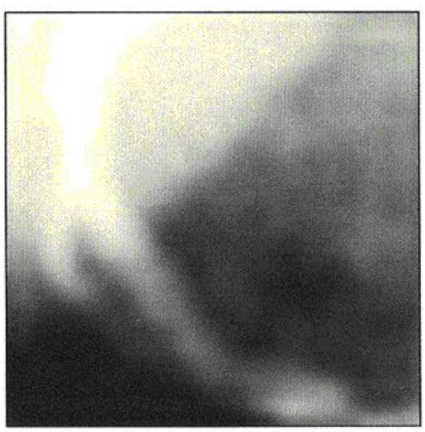
Signal processing algorithms are being developed at RLE to process periodic gravitational wave data for which the source location and frequency are unknown. Gravitational waves are generated when matter in the universe moves, causing a change in its gravitational field. The algorithms, developed by Professor Alan V. Oppenheim and graduate student Tae H. Joo, employ two-dimensional data processing in location-frequency space, and combine measures of coherence in multichannel data. This work has application to more general problems of narrowband signal detection in wideband noise, particularly when there is relative motion between transmitter and receiver. This project is carried out in collaboration with Professor Rainer Weiss of MIT's Physics Department, who built the 1.5–meter prototype interferometric gravitational wave detector.

RECONSTRUCTION FROM THRESHOLD CROSSINGS

ORIGINAL



THRESHOLD CROSSINGS



RECOVERED

Under appropriate constraints, an image can be exactly represented by its crossings of a single threshold. Shown above are the original image of an angiogram (upper left) and the threshold crossing representation (upper right). The recovered image (below) is obtained by applying a recently developed reconstruction algorithm to the threshold crossing. In general, signal processing techniques have played an important role in the development of medical technology. Research at RLE has included a variety of image representation and processing techniques applied to medical images such as the estimation of coronary artery boundaries in angiograms. Other medical applications for signal processing developed at RLE include the measurement of blood flow characteristics by analyzing the spectrum of heart sounds using an active ultrasonic measurement system.

computed simultaneously. This has led to the design of special-purpose computers for digital signal processing that are substantially faster than common single-sequence computers. Although some signal processing algorithms are very flexible computer programs that can be used for a variety of applications, new algorithms must constantly be developed since there are many different signal processing techniques, and a single algorithm cannot be used in all cases.

In the mid-'60s, the field of digital signal processing evolved rapidly when computers and digital circuitry became fast enough to process large amounts of data. The advent of digital computers for signal processing provided the flexibility to simulate signal processing systems before their implementation on analog hardware, thus introducing greater accuracy and cost effectiveness. Widespread signal processing applications resulted from the increased speed of computers, the introduction of fast array-processor peripherals, and the development of the microprocessor.

Many signal processing applications that require high sampling rates, high production volumes, or low cost can only be satisfied by special-purpose digital hardware. The use of these systems did not become widespread until custom VLSI circuits made them an attractive alternative to existing technologies. Digital signal processing systems are well-suited to VLSI implementation since they have highly parallel algorithmic structure, local connectivity, and circuit modularity.

Practical Applications of Signal Processing

In the 1950s, signal processing methods for seismic imaging were developed by the oil industry to determine the presence of oil and gas deposits deep in the earth. One imaging method exploded underground dynamite charges and recorded the vibrations as seismic traces on a seismograph. The seismic traces were then examined for arrival times and strength of the vibrations to determine the structure of sedimentary rock layers and whether oil was present. Seismic data analysis is also used in earthquake measurements and nuclear test monitoring.

Today, signal processing has many practical applications:

• Speech signals can be synthesized from discrete-time models of the human vocal tract using digital signal processing. The speech synthesizer is a time-varying digital filter which models the response of the articulators (mouth, throat, etc.) to excitation from the vocal cords and air turbulence. This results in quality voice synthesis for computers, and data reduction for the digital voice communications used in talking consumer products. Analysis of speech production can also be performed by examining the speech signal's spectrum and its related properties. These techniques can be used to determine the spectral properties of speech (such as the voiced sounds in vowels or the unvoiced sounds in fricatives) and the distance between pitch pulses in voiced sounds. Sounds can also be analyzed by examining their spectra. Once obtained, this information can be useful in the development of an automatic speech recognition or compression system.

• Signal processing is central to many defense-related systems, and plays a major role in automatic control, navigation, and target tracking. Radar and sonar systems rely heavily on the application of signal processing technology.

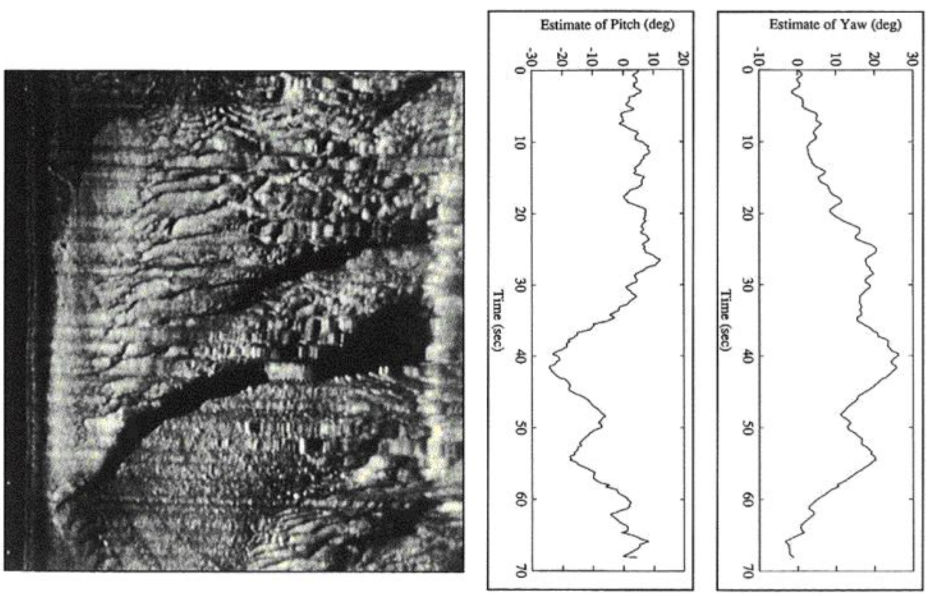
• In neurophysiology, signal processing techniques are used to mathematically characterize signals obtained by electroencephalography from the scalp of a subject. Frequency spectrum analysis is used to reveal the possible presence of energy prominences at certain frequencies, which are important to physicians for diagnostic purposes.

• Two-dimensional or image processing is used to analyze visual data such as x-rays, aerial photographs, and satellite transmissions from space. With the advent of high-definition television, new image processing algorithms are being developed to enhance picture quality, while new techniques are being investigated to improve the quality of existing transmissions. The potential of high-definition television coupled with image processing technology has opened the way for the all-digital television sets of the future.

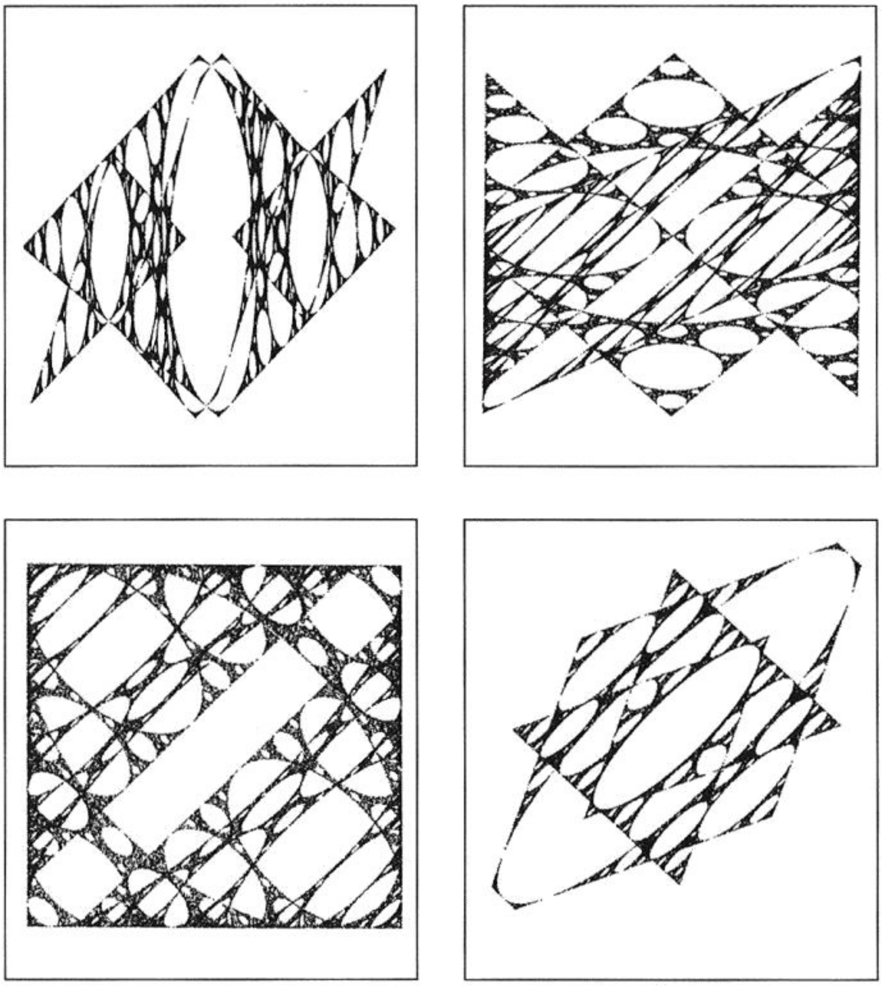
RLE's Digital Signal Processing Group

During the '60s, many areas of research capitalized on the development of computers by redefining traditional disciplines within a new computational environment. An example of this evolution was the broadening of studies in continuous linear and nonlinear systems to discrete systems. At RLE, the rapid growth of digital signal processing led to new techniques in speech such as spectral estimation.

Today, RLE's Digital Signal Processing Group pursues basic research in signal processing and algorithm design with applications to speech processing, image and video processing, and underwater acoustics. One of the group's primary focuses is to develop new signal processing algorithms and to explore their characteristics when applied to real and synthetic signals. These include algorithms for noise suppression and bandwidth reduction, robust model estimation, time and frequency scale modifications, homomorphic analysis, and the recovery of signals from limited information. Although the development of new algorithms is a major focus, the group also strives to maintain close ties to practi-



In side-scan sonar imaging, typically the transmitter and receiver undergo pitch, roll, and yaw as they are towed through the water, which results in distortion of the desired image. Professor Alan V. Oppenheim and graduate student Daniel T. Cobra, in collaboration with scientists at the Woods Hole Oceanographic Institution, develop and apply algorithms to estimate and then correct for this unwanted motion. Shown is an original side-scan image taken in Vineyard Sound (photograph) with estimates of pitch and yaw (graphs) that were obtained directly from the image.



Fractal signatures of state evolution in a chaotic digital filter are illustrated above. In certain finite-precision environments, the implementation of digital filters can exhibit this type of chaotic behavior. Professor Alan V. Oppenheim and graduate student Gregory W. Wornell examine the various applications of both chaotic dynamical system theory and fractal process theory to signal processing. This research could potentially lead to new models for quantization, speech generation, and signal coding.

cal applications and implementation issues, because algorithm efficiency not only depends on the number of required operations in a signal processing procedure, but also the algorithm's suitability to the computer architecture it will eventually run on.

Professor Alan V. Oppenheim is concerned with fundamental issues in signal processing. His primary interest is in the development of signal processing algorithms for speech, image, and seismic data. One major focus of his research is geophysical signal processing applications, including the development of new algorithms for wave propagation, analysis of data collected by seismic arrays, and the solution of inverse problems to infer underlying geological or oceanographic structures. In earlier work, Professor Oppenheim, in collaboration with the Woods Hole Oceanographic Institution, developed an algorithm that estimates the acoustic reflection coefficient at the ocean bottom. His studies in seismic digital signal processing have led to the development of several significant new techniques for seismic data analysis, such as cepstral analysis and homomorphic deconvolution.

Currently, he is working on the transformation of side-scan sonar data in order to extract topographic information from sonographs. Side-scan sonar, invented by Professor Harold Edgerton of MIT, has been used since the early '60s as an important tool in underwater exploration. It has enabled marine geologists to survey the ocean floor, to locate objects at the bottom of the sea, and to prospect for mineral deposits. The side-scan sonar collects reflected underwater sound waves and produces a graphic record (sonograph) of topographic features and the reflectivity of materials on the sea bed. But, since various distortions prevent the sonographs from presenting an accurate depiction of the topology, digital signal processing techniques are being applied to the sonographs to enhance the data.

Although signal processing has historically emphasized numerical techniques, knowledge-based signal processing is a new direction which combines the heuristic knowledge of speech and other signals with more traditional signal processing tools and concepts. Recent work at RLE has focused on knowledge-based pitch detection, signal enhancement, and distributed sensor networks. One area addressed by Professor Oppenheim is the symbolic representation and manipulation of knowledge and expressions in signal processing. The issues associated with this work include the uniform representation of knowledge, derivation of new knowledge from knowledge that has already been provided, and strategies to control the use of this knowledge. Also being investigated is the problem of signal matching using multiple levels of description in the context of vector coding of speech signals. This work seeks to reduce matching complexities by using multilevel signal representations that are simplified by abstraction of detail. Potential applications for knowledge-based signal processing are "intelligent signal processing" oriented toward signal understanding and characterization, "expert" systems capable of superior performance, image storage, pattern recognition, and signal processing system design environments.

In addition, Professor Oppenheim has applied new digital signal processing techniques to speech processing. This work is aimed at low-cost, high-quality compression and enhancement of degraded speech, and algorithm development for robust speech compression in the presence of additive noise. He has also investigated adaptive

Professor Jae S. Lim (standing) and graduate student Matthew M. Bace demonstrate a new method that was recently developed to reduce channel degradation in the National Television Standards Committee television system. The receiver-compatible method to reduce channel degradation is illustrated using a conventional transmitter and receiver (figure 1), a new transmitter and conventional receiver (figure 2), and a new transmitter and new receiver (figure 3). This method has potential application to high-definition television systems.

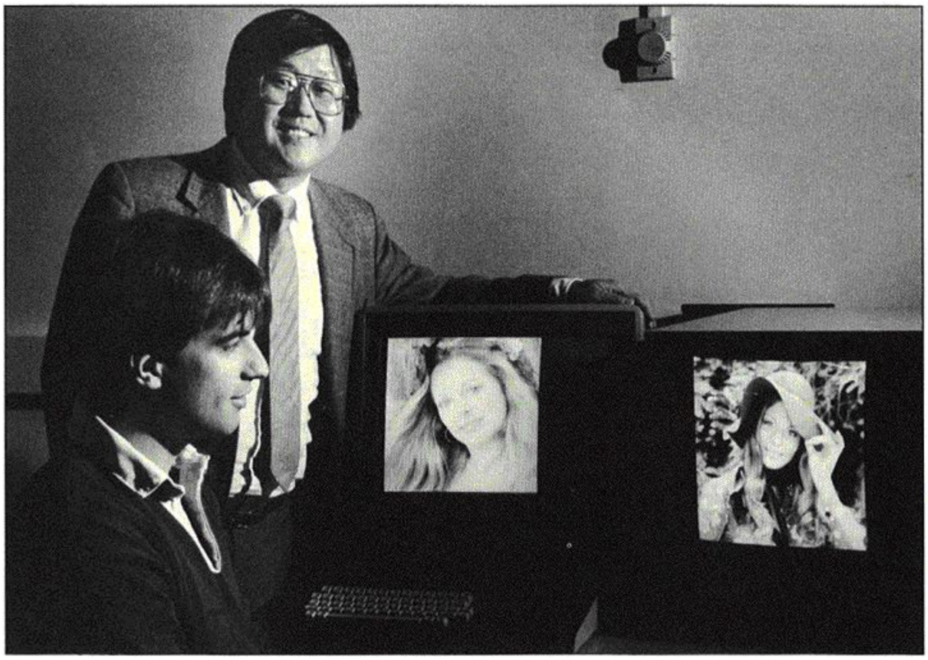


Photo by John F. Cook

Figure 1

Figure 2

Figure 3

noise cancellation techniques in multiple microphone environments.

At RLE, an intense effort is devoted to multidimensional signal processing which focuses on signal estimation and reconstruction from incomplete or noisy data, power spectrum estimation, and speech enhancement. Image or two-dimensional signal processing is also examined, including image reconstruction from a single threshold crossing. A new image restoration system has been developed to improve degraded images, and a new set of conditions has been introduced under which an image can be restored from its Fourier transform phase, or magnitude alone.

Professor Jae S. Lim's research interests include multidimensional signal processing, signal reconstruction and enhancement, and speech and image processing. He has developed a technique for very low bit rate video conferencing by transforming an image into a bi-level image, and then coding the difference between the consecutive image frames. Professor Lim has also investigated topics associated with image coding, where an image is represented in as few bits as possible while preserving its quality and minimizing transmission costs. Image coding has practical applications in video telephones, facsimile machines, digital television, and medical image storage.

Currently, Professor Lim is exploring various image processing issues. He is investigating new techniques for motion estimation which can be used for video interpolation, enhancement, and coding. He is also developing a motion compensation algorithm that will stabilize video images from undersea cameras used in exploration, reconnaissance, and salvage operations. Since undersea cameras are hampered by the effects of wave motion and ocean currents, they cannot monitor images at a constant depth with a constant angle. Although it may be impossible to mechanically stabilize these cameras, recent image processing research at RLE may make it possible to process the video images, and thus, make the camera appear stable. Professor Lim is also studying signal processing issues related to the improvement of current television systems and the design of high-definition television systems. In a recent effort, he developed a receiver-compatible system that reduces transmission channel degradation such as multipath echoes, noise, and interference. The new system requires a simple transmitter modification, and even though the transmission channel degradation is not reduced with an existing receiver, it is reduced significantly with a new receiver.

In addition to image processing research, Professor Lim has developed a new model-based speech analysis system for high-quality speech production. This research explores methods to apply a new speech model, known as the multi-band excitation speech model, in areas such as speech coding, enhancement, and time scale modification. A robust, good-quality speech coding system has been developed from this model with potential applications to mobile telephones.

Professor Bruce R. Musicus explores fast, efficient, highly parallel algorithms and special-purpose computer architectures for digital signal processing. The goal of his work is to develop more robust and sensitive algorithms that will extract information from digitally sampled data, and to devise unconventional architectures that are matched to specific algorithms with improvements in processing speeds. He has developed an inexpensive, high-performance cellular architecture design for these applications. Professor Musicus has also studied architectural transformations in signal processing systems, and has developed methods to manipulate signal flow graphs into several task-invariant architectural forms. His other research interests include stochastic estimation and signal reconstruction and enhancement.

Professor Arthur B. Baggeroer pursues studies in underwater acoustics, multidimensional signal estimation, and array processing. A major focus of his work involves geophysical and acoustic field experiments in the Arctic marginal ice zone to collect sonar and oceanographic data for signal processing. The results of this work characterize the acoustic reflections from the ocean bottom. Professor Baggeroer has also conducted research at the Woods Hole Oceanographic Institution on an underwater communication system used for the telemetry of data from untethered oceanographic sensors on the sea floor.

Dorothy A. Fleischer

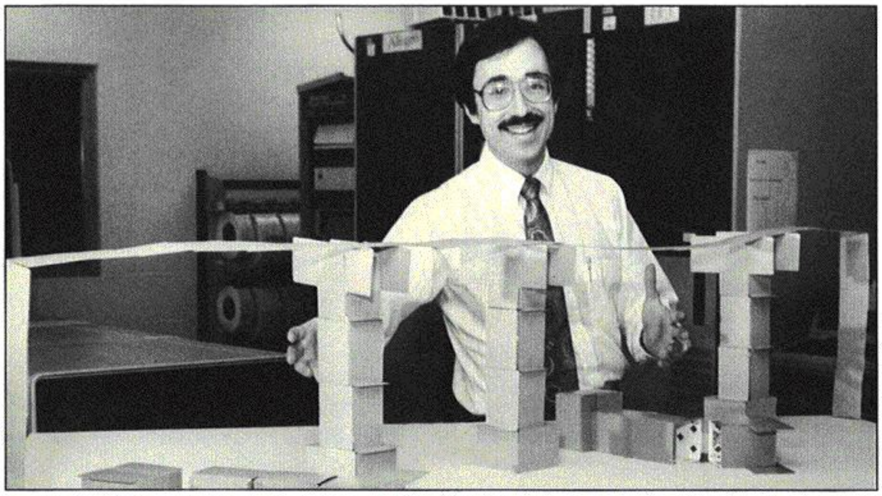


Photo by John F. Cook

Professor Bruce R. Musicus uses playing cards to demonstrate a new concept in fault-tolerant signal processor architectures. A small amount of redundant computation is introduced into the system, and the workload is spread across many processors. A simple test can detect which processors have failed, and can quickly correct the failed processors' outputs without repeating their calculations.

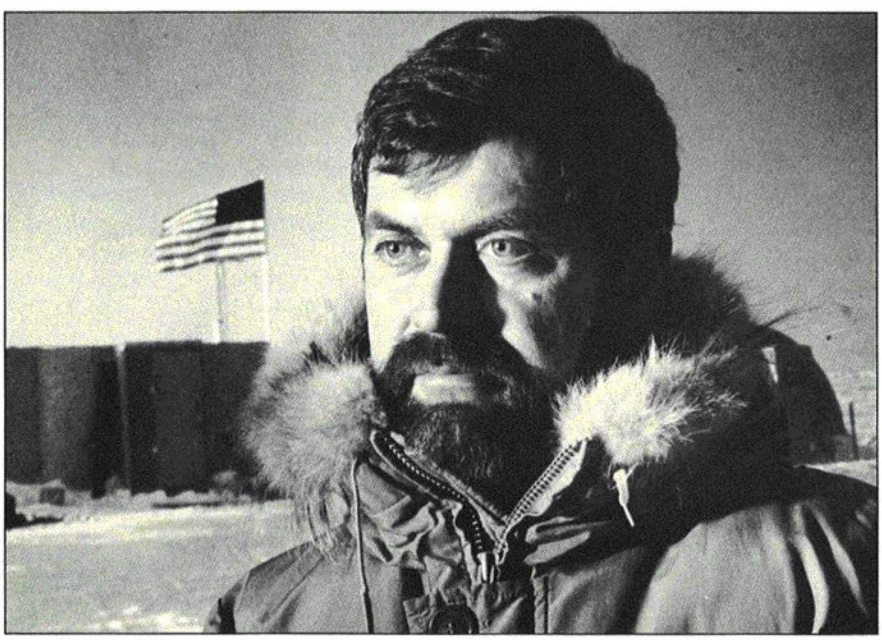


Photo by John F. Cook

In a worldwide cooperative effort with several academic and scientific institutions, Professor Arthur B. Baggeroer conducts field experiments at Fram Straits, Svalbard, Norway (82°N, 7°E) on techniques for modelling long-range sound propagation in the Arctic Marginal Ice Zone. These experiments seek to measure acoustic transmission and scattering, and seismic refraction in the Arctic Ocean. Campsites have been set up at various times over the last ten years using a horizontal receiving array, a multichannel data handling and storage system, and a digitally operated data analysis system.

FACULTY PROFILE:

Alan V. Oppenheim

After receiving his doctorate from MIT in 1964, Alan V. Oppenheim joined the Institute faculty, where he is currently Professor of Electrical Engineering and Computer Science. His research interests are in the general area of signal processing and its applications to speech, image, and geophysical signal processing. He is co-author of the widely used textbooks Discrete-Time Signal Processing, Digital Signal Processing, *and* Signals and Systems. *He is also the editor of several advanced books on signal processing.*

Professor Oppenheim is a member of the National Academy of Engineering and a Fellow of the IEEE. He has been a Guggenheim fellow and a Sackler fellow, and has received many awards for outstanding research and teaching, including the 1988 IEEE Education Medal, the IEEE Centennial Award, and the Society and Technical Achievement awards of the IEEE Society on Acoustics, Speech and Signal Processing.

• *Who influenced you most in the early stages of your career?*

As a graduate student, I worked with Manuel Cerrillo in RLE during the early phases of my doctoral research. Cerrillo was a captivating and inspiring person who had many beautiful and sometimes strange ideas. Among other things, he interested me in modern algebra, and that eventually led to the ideas in my doctoral thesis. I also worked closely with Amar Bose as a graduate teaching assistant, and he was an inspiration to me in terms of standards, ideals, creativity, and teaching methodology. Another influential person, during my time as a graduate student and afterwards, was Tom Stockham. He was tremendously excited and encouraging about the nonlinear theory that I was developing, and later played a major role in its application to audio and image processing.

Perhaps the person who had the greatest impact on me after I joined the faculty in 1964 was Ben Gold from Lincoln Laboratory. I first met Ben when he was a visitor in Ken Stevens' speech group. Ben has a friendly, informal, and supportive style, in addition to deep insights and broad experience. Because of these qualities, I and many others sought him out for discussion and consultation. Ben and I worked together for many years, and are still very close personal friends.

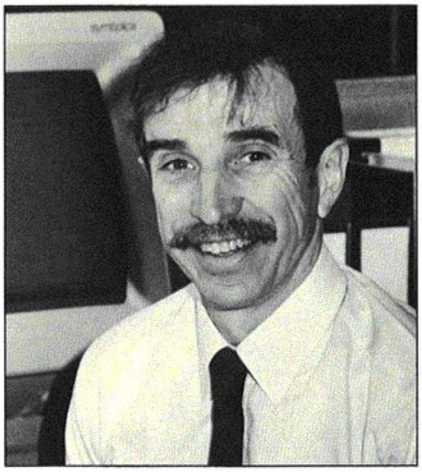


Photo by John F. Cook

Professor Alan V. Oppenheim

• *What was the nature of your work at Lincoln Lab from 1967-69?*

I initially became involved at Lincoln Laboratory as a consultant, primarily through my interactions with Ben Gold and Tom Stockham. In 1967, I decided to take a two-year leave of absence from the faculty and spend full time there as a staff member. During that period, I was fortunate to work closely with Ben and Tom, as well as with Charlie Rader on a variety of speech and image processing problems, and on several theoretical issues related to the emerging field of digital signal processing. The landmark paper on the fast Fourier transform had just been published by Jim Cooley and John Tukey, and there was a rapidly growing interest in exploiting the newly discovered flexibility, efficiency, and accuracy of processing signals on a computer or with digital hardware. At Lincoln, there were many signal processing applications that took advantage of this new technology, enthusiastic and creative colleagues, and a long list of interesting problems. It was a unique and special opportunity to work there during that time. My involvement with Lincoln has since continued at various levels, including two extremely enjoyable and educational years as half-time Associate Head of the Data Systems Division under Al McLaughlin.

• *How did RLE's research in signal processing evolve?*

Signal processing has a long, rich history dating back to the 17th century, and perhaps before. Wherever signals arise, signal processing plays a role in some form. At RLE in the '50s and '60s, there was considerable activity in analog signal processing in the context of communication theory (Norbert Weiner, Claude Shannon, Yuk Wing Lee), speech processing (Ken Stevens), video (Bill Schreiber), biomedical processing (Murray Eden), and audio (Amar Bose and Manuel Cerrillo). In the early '60s, Amar Bose, Tom Stockham, and Jim Bruce used computer processing to simulate room acoustics, and Ken Stevens was using the TX-0 computer for speech analysis. However, at that time, a typical viewpoint at MIT and elsewhere was that processing signals on a computer was inherently non-real-time, and was principally intended to approximate or simulate an analog system. It wasn't until after the disclosure of the fast Fourier transform, and the associated developments that it spawned, that an appreciation developed for the advantages of implementing real-time signal processing systems digitally. Many of these developments happened at Lincoln Laboratory and Bell Labs. As a consequence of having been part of the Lincoln activity, when I returned to MIT in 1969, I started a research group in RLE and offered what I believe was the first digital signal processing course in the country.

• *People have said your research group is the strongest of its kind in the world. What are your feelings on that?*

To the extent that it's true, I'm very pleased. Over the years, there have been several factors that have contributed to our group's good fortune. Our relationship with Lincoln has been important, not only in terms of the people we interact with, but also in terms of the technical problems that they deal with. I feel strongly about this because in order to do good research, you need coupling to the real world, and this was one of the most important aspects of my Lincoln experience.

Another factor has been our relationship with industry. When the chemistry is right, the relationship can be tremendously productive and exhilarating. It's a great opportunity for technology transfer. We are introduced to important real problems, which motivates us to think in certain directions; the companies see the results of our work, and that motivates them to apply our research. The two companies that we've had perhaps the most successful relationships with are Sanders Associates and Schlumberger. Recently, we've also been working with a small company called Atlantic Aerospace. These companies all have real problems that they're trying to solve, and they're looking for innovative solutions. At RLE, we have exciting solutions that are looking for problems. If we can match these two situations in the right way, everybody wins.

Also, the MIT students are extraordinary. Fortunately, our group has had the resources to develop an environment that attracts some of the best students. We've also been extremely fortunate in terms of the support and encouragement that we've received from our research sponsors.

• *The faculty members in your group have diverse talents and research interests. What's the "glue" that brings you together?*

Our common culture is signal processing and a deep interest in the development of innovative algorithms. Also, our individual interests in applications strongly overlap. For example, I'm concerned with speech processing, image processing, geophysical signal processing, and underwater acoustics. Art Baggeroer, who is partially affiliated with our group, has a strong background in stochastic signal processing and is intimately involved with communication systems, geophysical signal processing, and underwater acoustics. Jae Lim has focused heavily on speech, image, and video processing. Bruce Musicus is the individual in our group who is most significantly involved with hardware. He is concerned with signal processing architecture issues and the theoretical aspects of signal processing algorithms. All of us come from a common intellectual background, and both Jae and Bruce did their doctoral research in this group.

• *Is there a current area of research that you're excited about?*

There are many areas that we're excited about, and as usual, more ideas to pursue than the time to pursue them. Some of these areas are applications-oriented, and others are highly speculative ideas which, if they lead somewhere, are likely to be in the category of "solutions in search of problems." An example on the applications side is our work on removing distortions in side-scan sonar images. Ideally, when the sonar equipment is towed through the water, it should move in a straight line. However, the ocean's environment and ship's movement introduce distortions such as the pitch, roll, and yaw of the sonar. We are exploring the processing of side-scan sonar imagery to estimate the pitch, roll, and yaw from the data, and then use the estimates to remove the distortions.

Another applications-oriented area is noise or signal cancellation, whereby we estimate and then subtract out the unwanted components from a composite signal. These techniques often involve making measurements in real-time, and based on those measurements, adaptively subtracting out what's not wanted; essentially relying on destructive interference. Techniques of this type have been used for multiple microphone speech enhancement, in noisy industrial environments, and in other controlled environments. But, it's in the less controlled environments where the problems become significantly more exciting and difficult.

We're also excited about the broad area of multidimensional signal processing, on which Jae and I have worked for many years, and on which he has recently published a new textbook. One problem area involves the reconstruction of multidimensional signals from only partial information and some known constraints. For example, we have shown that, under certain general constraints, an image can be recreated from its crossings of only a single threshold. The procedure becomes increasingly robust as more thresholds are used, and while this theory can be thought of as one of those solutions looking for the right problem, potentially it may be the basis for some innovative image coding algorithms. Jae is also heavily involved in two-dimensional image processing and enhancement, and in video or three-dimensional processing, including motion estimation and high-definition television.

Somewhat more speculative, but very exciting, is our work on symbolic signal processing. The notion of manipulating signals and signal processing systems symbolically rather than numerically requires a very different kind of signal processing. Several years ago, we became seriously involved with combining numeric and symbolic signal processing. Our interest in this was originally motivated by Bob Kahn when he was at DARPA, who suggested that we incorporate some of the principles of artificial intelligence into our research. We found the symbolic processing technology used in artificial intelligence and other disciplines to be of particular interest. We pursued this in several directions. One has been in combining classical numerical signal processing with rule-based systems to use more qualitative or subjective information and constraints. We've worked closely with Sanders Associates on combining numerical processing with rule-based systems, and through Sanders, several of our ideas have been successfully incorporated into practical signal processing systems.

We've also found the symbolic processing technology used in artificial intelligence and other disciplines to be of particular interest and importance in developing a signal processing algorithm design environment. In a somewhat similar manner to the way in which the MacSyma system manipulates mathematical expressions symbolically, our system manipulates signals and signal processing expressions symbolically to achieve semi-automated or user-interactive algorithm design. Typically, in signal processing system design, an algorithm is first specified at a relatively high level. Then, the system designer manipulates and rearranges the algorithm using a well-formulated set of rules, properties, or identities that are blended with formal or informal cost measures, experience, or intuition. Parts of this process can be automated, and often a computer can sort through many more options than a designer. Also, a design environment can potentially incorporate more expertise into this process than a relatively inexperienced system designer. Bruce Musicus' work in mapping signal processing algorithms onto VLSI architectures also couples nicely into this work. In the somewhat distant future, one could imagine a sophisticated signal processing design environment that can symbolically explore algorithm rearrangements and implementations based on cost measures associated with VLSI implementation, and can suggest details of the implementation. Probably not this year, though.

In a very different direction, we've recently become interested in chaos theory and fractals, and how they relate to image modelling, spread-spectrum communications, and signal coding. We're exploring a variety of exciting ideas, but there's nothing specific that I can report on right now. Mentioning chaos and fractals brings to mind an issue that I tend to be sensitive to and somewhat nervous about in my research. There's always a certain set of topics or areas that receive a lot of attention in the technical or popular press, and tend to develop into fads. Inevitably, expectations quickly outrun reality. A lead article somewhere will describe how a particular science or technique is going to solve some major problem, then, what they'd really like to show in next week's issue is how it *did* solve the problem. If it didn't, often disappointment and disinterest follow. For this reason, I prefer to maintain a low profile on highly speculative ideas or directions.

• *What is the relationship between digital signal processing research and computer hardware technology?*

It's really a symbiotic relationship. In fact, in my opinion, much of the vitality of the digital signal processing field stems from the close coupling between applications, algorithm development, and the technologies available for signal processing. Ten years ago, nanosecond signal processing microprocessors with floating-point arithmetic (which are commercially available now) would have seemed totally far-fetched. Similarly, a decade ago, the fast Fourier transform algorithm in real-time, at any reasonable data rates, was confined to supercomputers or racks of special-purpose digital hardware. There are now special-purpose chip sets, and even a single-chip fast Fourier transform processor is available. The pace of technology development is extraordinary, and there's certainly no indication that it's slowing down. Of course, that gives those of us who are primarily involved in algorithm development considerable confidence, that if we were to come up with a robust, intellectually sound innovative algorithm, it will eventually be implementable at the size, speed, or cost needed to make it practical. Undoubtedly, this is a naive assumption, but so far, things have worked out that way. In a similar way, those who develop the technology make the assumption that algorithms and applications will require and make use of the new technologies.

It is worth mentioning that many of the developments referred to under the heading of digital signal processing are not really tied to computer or even digital hardware. In the revision of my book, *Digital Signal Processing*, coauthored with Ron Schafer, we changed the title to *Discrete-Time Signal Processing* in recognition of the fact that it is the discrete-time nature of the techniques and technology that we are focusing on. Digital techniques will continue to be extremely important, but there are many other discrete-time technologies such as surface-acoustic-wave and charge transport devices. And, of course, there will always be new continuous-time signal processing technologies.

• *In your opinion, who has made the most substantial contributions to digital signal processing?*

One of the important early people involved in digital signal processing and digital filtering was Jim Kaiser. Jim had worked in the MIT Electronics Systems Lab on sampled data control systems as a graduate student. After MIT, he went to Bell Labs and became involved in digital filter design for vocoder simulation. Jim is clearly one of the pioneers in digital filtering, and he recognized very early the potential of filtering signals with computers and special-purpose digital hardware.

At Lincoln Laboratory, Ben Gold and Charlie Rader were also using digital filters for vocoder simulation and made many substantial contributions through their work on speech processing and processor architectures; as did Tom Stockham with his work on digital audio and image processing. Larry Rabiner and Ron Schafer (both former graduate students in RLE) contributed significantly with their work on speech processing at Bell Labs. Bill Lang of IBM played an important role in promoting the IEEE group on Audio and Electroacoustics as the "home" for digital signal processing. Eventually, the group's name was changed to Acoustics, Speech, and Signal Processing.

Clearly, the contributions of Jim Cooley and John Tukey in identifying the fast Fourier transform algorithm were seminal. It was probably the most significant event in launching digital signal processing in its current form.

• *What do you see for the future of signal processing?*

Fortunately, there will always be signals that need to be processed; advances in technology will continue to offer new opportunities for sophisticated systems; and this, in turn, energizes creativity in algorithm development and refinement for practical systems. Frankly, I don't ever see an end to this cycle. There are many areas of theoretical research that will likely see significantly more activity in the future than they have so far. One that stands out in my mind is multidimensional signal processing as it arises, for example, in image and video processing, and in a variety of array processing problems. It's a difficult area theoretically, in part because multidimensional polynomials don't have the nice properties of one-dimensional polynomials. Also multidimensional data sets of any reasonable size in each dimension result in very large amounts of data to be processed. Consequently, current practical algorithms tend to be relatively simple and conservative. As the technology offers expanded opportunities for computational speed, data storage, and accessing, I believe that some dramatic new algorithms will emerge.

Another area in which there is considerable need for innovation and new ideas is nonlinear theory and nonlinear signal processing. This has always been a difficult area because very simple nonlinear systems can generate apparently complex behavior. Of course, this might be part of the promise if that complex behavior can be understood.

Complex behavior of simple nonlinear systems is intimately connected to chaos theory, where there is now considerable interest and activity. Perhaps developments in this area will eventually lead to novel nonlinear signal processing systems, and to a better understanding of the behavior in current systems.

With regard to applications, sophisticated signal processing systems in the military context will continue to play an important role, and in fact, there is considerable evidence that the importance of signal processing will expand significantly in areas such as "smart weapons," surveillance, intelligence, communications, and battlefield management. In the consumer arena, signal processing will certainly continue to play a significant role. Signal processing is so much a part of our everyday lives that we almost don't notice it in our home entertainment and communication systems. In the not-too-distant future, a television receiver will undoubtedly be a special-purpose digital computer because of today's major developments and interest in signal processing technology for digital and high-definition television. Other types of very sophisticated consumer-oriented digital signal processing are on the horizon. The telephone system is clearly going digital, and voice mail is increasingly common. The difficulties of voice mail present interesting opportunities for digital signal processing research and its applications. Currently, for example, no answering machine or voice mail system can scan messages for key words. The notion of adding word spotting and very limited speech recognition to voice mail systems requires sophisticated signal processing that incorporates processing algorithms and the application-specific work done by people like Victor Zue. Other applications, such as adaptive audio systems in which the equalizer frequency response adjusts automatically, are do-able, but the current technology doesn't yet make it cost-effective or fast enough for it to be practical.

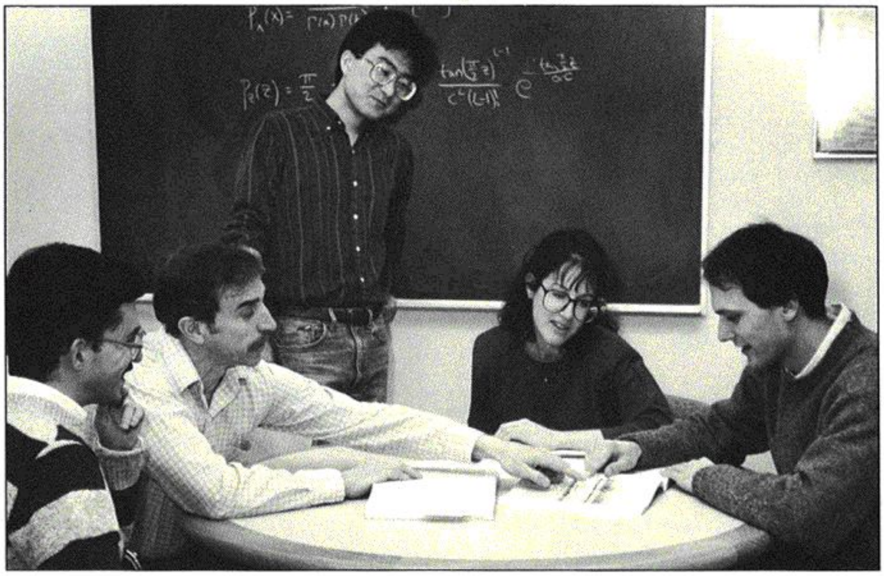


Photo by John F. Cook

Discussing the results from their signal processing research are (from left) graduate student Daniel T. Cobra, Professor Alan V. Oppenheim, graduate students Tae H. Joo, Michele M. Covell, and Gregory W. Wornell.

• *What is the nature of your research at the Woods Hole Oceanographic Institution?*

As you know, I spend summers working there as part of the Joint MIT/Woods Hole Program. One project that I have worked on, in collaboration with George Frisk, a physicist at Woods Hole, involves the study of ocean bottom acoustics. George ran a series of experiments to collect acoustic measurements, and I've worked with him to interpret the data and develop theories on how to extract model parameters of the ocean bottom from it. One of the interesting things that came out of this work was the so-called fast Hankel transform algorithm. More recently, I've worked with Jules Jaffee on problems related to correcting distortions in side-scan sonar data.

Both the intellectual and recreational aspects of Woods Hole make it an extremely enjoyable place to spend time. The town of Woods Hole, which is really part of Falmouth, has a fairly academic flavor because the Oceanographic Institution, the Marine Biological Laboratory, and the U.S. Geological Survey are located there. A magazine article once commented that, in the summertime, there's a beach at Woods Hole with the highest density of Nobel laureates of any beach in the world. The town itself has a unique style that's perhaps best summarized on a tee-shirt that says on one side, "Isn't Falmouth nice," and on the other side, "Isn't Woods Hole weird." I find the Institution exciting because many large-scale experiments there involve data collection in a natural environment. I also have a special love of the ocean, and my extracurricular activities include windsurfing and sailing. It's nice to integrate those activities into an environment where I can also carry out my research.

• *Why did you choose a teaching career?*

That question triggers many memories for me. I became a graduate teaching assistant because I liked tutoring as an undergraduate, and I needed the financial support. After my first few weeks as a teaching assistant, I felt that a new world had opened up for me. Teaching forces you to understand things at a very fundamental level, in part, because in preparing for class, there's just a natural motivation to want to be fully prepared. Even so, no matter how carefully you think you understand something, it's likely a student will ask a question that you hadn't anticipated. A strong side benefit to this is understanding something better and at a deeper level than you thought possible. Another attraction of teaching, for me, is the thrill of a "successful" performance. Of course, not every class goes that way. And the ones you thought did, often didn't—and vice versa.

As I said earlier, Amar Bose had a tremendous impact on my view of teaching. I recall Amar commenting on the enormous leverage and responsibility that teaching provides in terms of propagating your standards and ideals.

If those standards and ideals are transmitted to just a small number of students who go into teaching careers, they will likely do the same for their students, and this expands through a succession of generations. There are many events that continually remind me of this far-reaching and often unseen impact, and one in particular happened at a recent technical conference. I had just gotten into an elevator with one other person, and when the doors closed, he looked at me and said, "I don't know if you remember me, but I took your videotape course several years ago." Seriously.

• *What's the most challenging part of your career—scientist, teacher, or author?*

I enjoy the interrelationship of the three, so I don't think they should be separated. Of course, the role of a teacher isn't just limited to a classroom, you are also a thesis advisor and mentor. I strongly believe that in order to be an effective teacher, either in or outside the classroom, you must be involved in research because it keeps ideas fresh, exciting, and on the cutting edge. By the same token, teaching benefits research by providing a mechanism to convey and refine ideas. Every semester that I teach the digital signal processing course, new insights emerge about the material that end up affecting my research. As a researcher, it's important to present your ideas in writing, in a classroom, in conferences, or in whatever way forces you to communicate your ideas and present them for scrutiny. Many people hate to write because the idea of putting one's ideas down on paper for peer review can be terrifying. Although I don't enjoy writing, it's always nice to have written. Once it's done, it's a thrill to see it. But, for most of us, it's an agonizing process.

• *What's the most rewarding aspect of your work?*

The individual interaction with my students. My hope is that by the time they graduate, each of my students will have an international reputation, and that we are personal friends and professional colleagues. I can't achieve this with every student, but watching the relationship evolve from one of student-teacher to one of close friend and peer is extraordinarily gratifying. That is my reward.

circuit breakers

Photo by John F. Cook

Dr. A. Nihat Berker, Professor of Physics, was elected to Fellow of the American Physical Society in February 1989. The Society confers the title of Fellow to those members who have contributed to the advancement of physics by independent, original research, or who have rendered some other special service to the cause of the sciences. Professor Berker was cited for the development of the position space renormalization-group technique and its application to studies of phase transitions in physisorbed systems and liquid crystals.

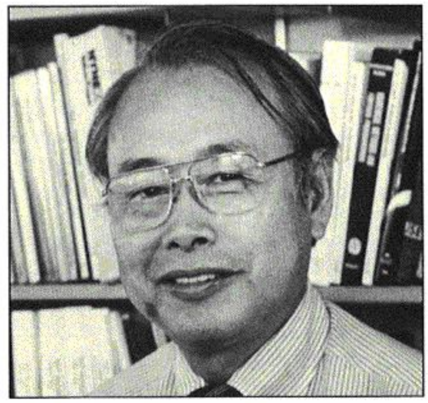


Photo by John F. Cook

Dr. Francis F. Lee (S.B. '50, S.M. '51, Ph.D. '66), Professor Emeritus of Electrical Engineering and Computer Science, has retired from the Institute. Since beginning his association with RLE as a research assistant in 1952, Professor Lee focused his research on digital systems applications and sensory aids for the handicapped. In the mid-'60s, he worked on character recognition and speech production. Following his studies on the relationship between written language and phonetic transcription, Professor Lee developed a computer-driven system to synthesize speech by rule, which contributed to a reading machine for the blind.

Photo by Phokion Karas

Recently appointed White House Chief of Staff, ***Dr. John H. Sununu*** (S.B. '61, S.M. '63, Ph.D. '66) continues the RLE tradition of serving as the President's "right-hand man." Coincidentally, Dr. Sununu was a graduate thesis student at RLE when then-RLE Director Jerome B. Wiesner was summoned to Washington to serve as President John F. Kennedy's science advisor. Dr. Sununu investigated measurements of cesium vapor thermal conductivity in RLE's Plasma Magnetohydrodynamics Research Group in 1961, when the above photo was taken.

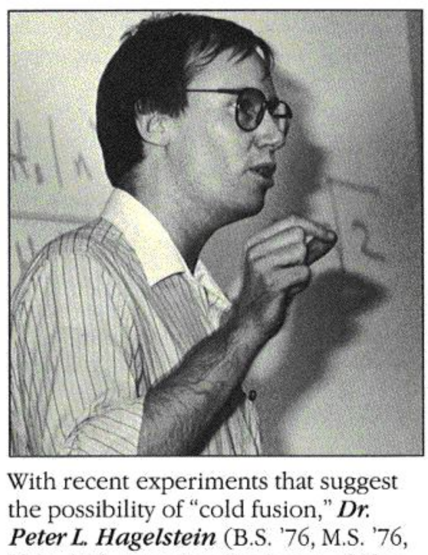
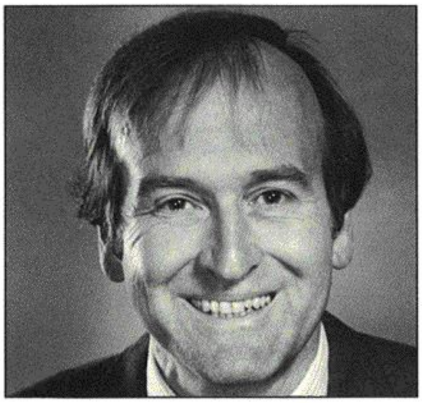


Photo by John F. Cook

With recent experiments that suggest the possibility of "cold fusion," ***Dr. Peter L. Hagelstein*** (B.S. '76, M.S. '76, Ph.D. '81), Associate Professor of Electrical Engineering and Computer Science, has advanced a speculative theory which seeks to explain the nature of these processes. Professor Hagelstein presented an overview of these developments to interested colleagues at MIT on April 14, 1989.



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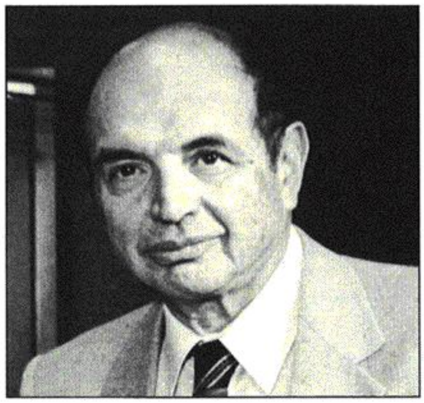


Photo by John F. Cook

Dr. Henry I. Smith, Professor of Electrical Engineering and Computer Science, and Senior Research Scientist *Dr. Robert H. Rediker* were among the ninety outstanding engineers elected as members of the National Academy of Engineering in February 1989. Election to the Academy is among the highest professional honors awarded to engineers. Academy membership distinguishes those individuals who have made important contributions to engineering theory and practice, and have demonstrated unusual accomplishment in new and developing fields of technology. Professor Smith was cited for his contributions to submicron structure technology and research, and his leadership in teaching and promoting submicron structures. Dr. Rediker was honored for his pioneering contributions and leadership in constructing semiconductor compound light emitters and lasers.

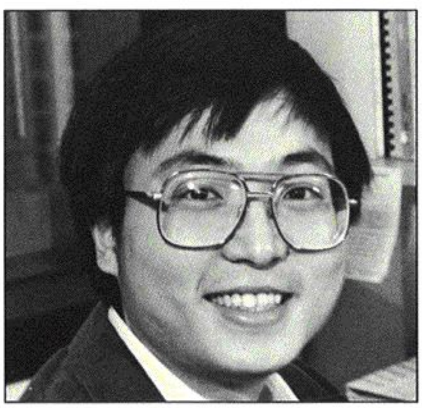


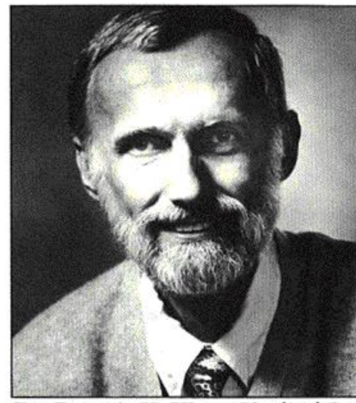
Photo by John F. Cook

Dr. Xiao Dong Pang was appointed Research Scientist in RLE's Sensory Communication Group in March 1989. Previously, Dr. Pang was a Fairchild Fellow at MIT. In his new position, Dr. Pang will conduct research on aids for the hearing impaired, on the abilities of the human hand to sense and operate upon the environment, and on multimodel human interfaces for teleoperator and virtual environment systems. Dr. Pang received his B.S. ('81) in Electrical Engineering from Zhejiang University, an M.S. ('83) in Biomedical Engineering from Vanderbilt University, and Ph.D. ('88) in Electrical Engineering and Computer Science from MIT.

Photo by John F. Cook

Dr. Mandayam A. Srinivasan was appointed Research Scientist in RLE's Sensory Communication Group in February 1989. Dr. Srinivasan will conduct basic research on hand biomechanics and the sense of touch, as well as research directed towards the use of manual sensing and control in teleoperator systems. Dr. Srinivasan came to MIT in 1988 as a Fairchild postdoctoral fellow. He holds a B.E. in Civil Engineering ('75) from Bangalore University, an M.E. in Aeronautical Engineering ('77) from the Indian Institute of Science, and an M.S. in Applied Mechanics ('77) and Ph.D. in Mechanical Engineering ('84), both from Yale University.

In Memoriam



Dr. Dennis H. Klatt, 50, died December 30, 1988, at the MIT Medical Department, following a long battle with cancer. Dr. Klatt was a Senior Research Scientist in RLE's Speech Communication Group. After obtaining his undergraduate degree in 1961 from Purdue University, Dr. Klatt received his doctorate in communication sciences from the University of Michigan in 1964. He joined MIT in 1965 as an Assistant Professor, and became a Senior Research Scientist in 1978.

Dr. Klatt made significant contributions to the areas of speech production and synthesis. In collaboration with Professor Kenneth N. Stevens, he conducted research in vocal tract modelling and furthered the understanding of how individual speech sounds are produced. Dr. Klatt also developed a rule system for constructing speech waveforms from abstract linguistic specifications. He sought to incorporate naturalness into his speech synthesis systems, and his work was applied commercially by the Digital Equipment Corporation in DECtalk. Dr. Klatt also pursued research in speech perception and the specification of acoustic properties for various phonetic distinctions. As the author of more than 60 scientific papers and a co-author of *From Text to Speech: The MITalk System*, he was working on a book at the time of his death.

Dr. Klatt received the Silver Medal in Speech Communication from the Acoustical Society of America, and the John Price Wetherhill Medal from the Franklin Institute.

Photo by John F. Cook

History of Digital Signal Processing at RLE

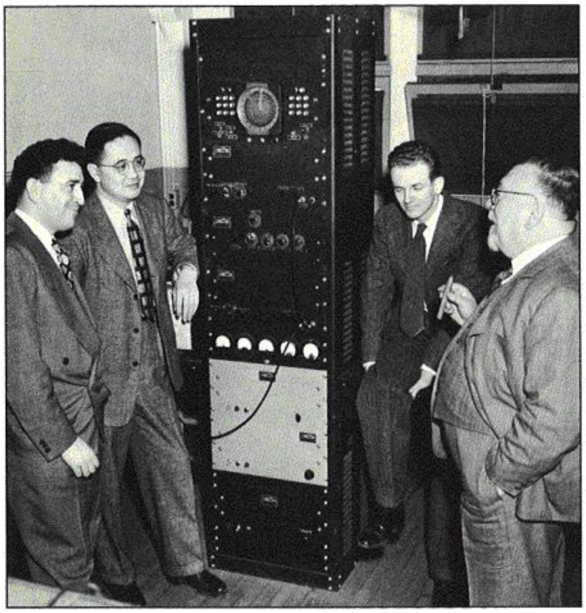


Photo by Ben Diver

"The statistical methods of linear filtering and prediction evolved largely by Professor Norbert Wiener were used to some extent during World War II, but the field of statistical communication theory actually blossomed further after the War. Having worked with Professor Wiener for many years, Professor Y. W. Lee brought to the Laboratory the mathematical basis for the theory. Within a very short time, the potential significance of correlation techniques had fired the imagination of all communication engineers. To a very important extent the general enthusiasm was due to experimental evidence (provided by Lee, Wiesner, and Cheatham) that weak signals could be recovered in the presence of noise by using correlation techniques. From that point the field evolved very rapidly."—Professor Henry J. Zimmermann, RLE Director, in Twelve Years of Basic Research: A Brief History of RLE, *1958*

ca. 1948

From left: Professors Jerome B. Wiesner and Yuk Wing Lee with his first doctoral student Thomas P. Cheatham, Jr., and Professor Norbert Wiener. Cheatham built the first electronic (analog) correlator, shown in the photograph. The final version of the machine represented the waveform as a sequence of sample values, and multiplied parts of samples by generating a pulse with a height proportional to one sample and a width proportional to the other. Cheatham's analog correlator paved the way for Henry E. Singleton's digital correlator in 1949, which rapidly performed storage, multiplication and integration by a binary digital process.

Photo by Ben Diver

1953

Professor Ernst A. Guillemin (left) and Dr. Manuel V. Cerrillo discuss filter theory. Professor Guillemin was well-known for his commitment to teaching and research in circuit theory, and Dr. Cerrillo was devoted to signal processing research in man and machine using both auditory and visual signals as media. Together, Professor Guillemin and Dr. Cerrillo collaborated on problems in time-domain network synthesis.

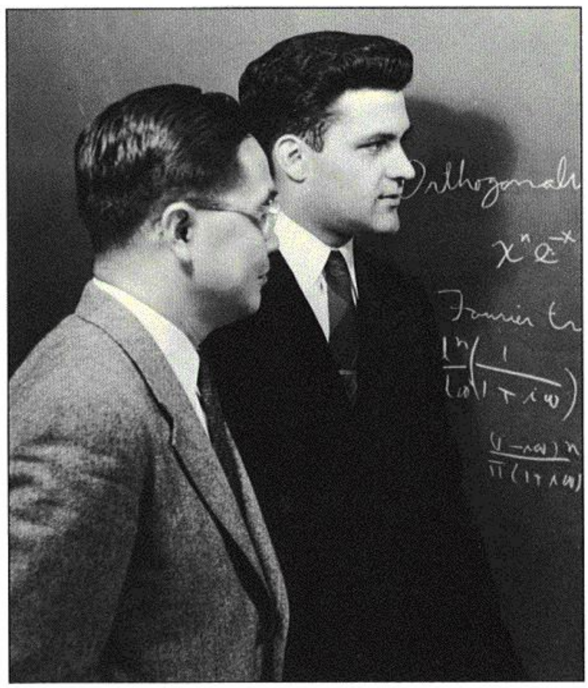


Photo by Ben Diver

mid-1950s

Professors Yuk Wing Lee (left) and Amar G. Bose convinced Norbert Wiener to present a series of lectures on nonlinear problems in random theory. Professor Lee tape-recorded the lectures, and photographed hundreds of complex equations diagramed on the blackboard by Wiener. In collaboration with Professor Bose, Lee compiled this material into the first book on the subject in English, Nonlinear Problems in Random Theory, *published in 1958.*

"With the growth of computing in RLE in the early 1960s, it was natural that many areas of research would exploit this new capability, and in many cases new research groups evolved through coupling traditional disciplines with a new computational emphasis. One example is the extension of the earlier study of continuous systems, both linear and nonlinear, to a study of discrete systems, thus forming a new area now called digital signal processing. Professor Alan Oppenheim built up a very strong group within RLE, initially addressing problems in speech such as spectral estimation and coding techniques, but in more recent years becoming involved in two-dimensional image-processing techniques."—Karl L. Wildes and Nilo A. Lindgren, A Century of Electrical Engineering and Computer Science at MIT, 1882-1982

1960

RLE's Speech Communication Group, under the direction of Professor Kenneth N. Stevens (far right), developed signal processing programs on the TX-0 computer for speech applications. Gathered around the console of the TX-0 (from left) are Hiroya E. Fujisaki, Paul T. "Pete" Brady, Jr. (seated), Gordon M. Bell, and Kenneth N. Stevens.

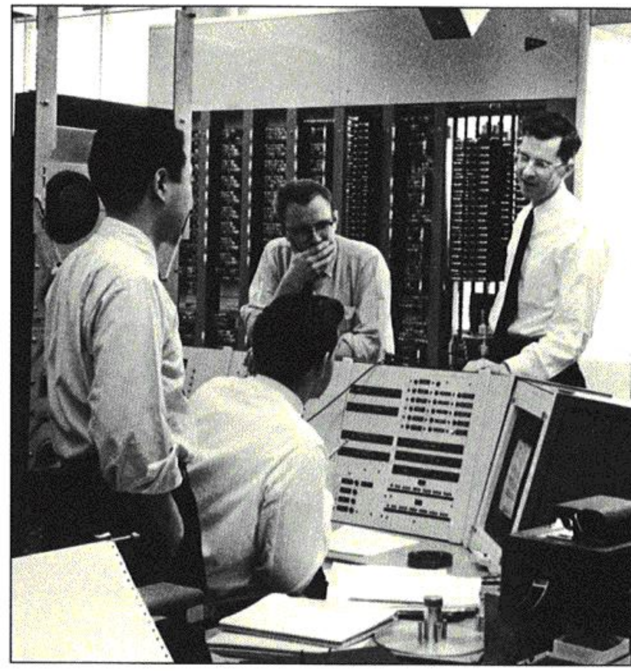


Photo courtesy of MIT Historical Collections

1963

Variation on a theme (from left): graduate student Thomas G. Kincaid, violinist George Humphrey of the Boston Symphony Orchestra, and graduate students John F. McDonald and Alan V. Oppenheim. This ensemble worked with Dr. Manuel V. Cerrillo on the application of group theory to generate synthetic Strauss waltzes.

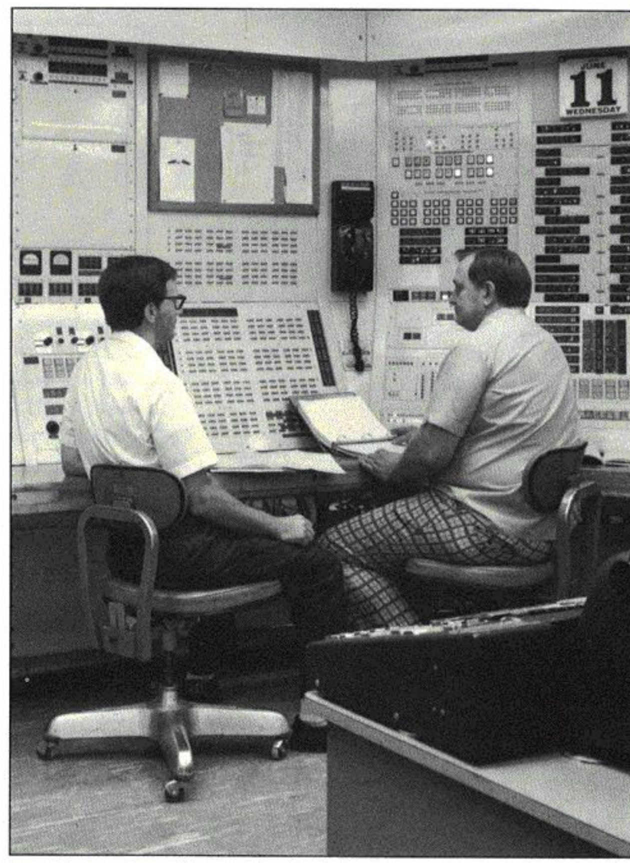


Photo courtesy of Lincoln Laboratory

1975

During the '60s and '70s, researchers from RLE used the TX-2 computer at Lincoln Laboratory for a variety of applications, including simulation, imaging, time-sharing research, and the investigation of neural nets. Seated at the TX-2's console are Omar Wheeler of Lincoln Laboratory (right) and an unidentified scientist.

UPDATE: Communications

Publications

RLE has recently published the following technical reports:

Finding Acoustic Regularities in Speech: Applications to Phonetic Recognition, by James Robert Glass. RLE TR No. 536. December 1988. 152 pp. $10.00.

Formalized Knowledge Used in Spectrogram Reading: Acoustic and Perceptual Evidence from Stops, by Lori Faith Lamel. RLE TR No. 537. December 1988. RLE TR No. 537. 180 pp. $12.00.

Levinson and Fast Choleski Algorithms for Toeplitz and Almost Toeplitz Matrices, by Bruce R. Musicus. RLE TR No. 538. December 1988. 95 pp. $8.00.

The OKI Advanced Array Processor (AAP)–Development Software Manual, by Bruce R. Musicus. RLE TR No. 539. December 1988. 242 pp. $17.00.

A Theoretical and Experimental Study of Noise and Distortion in the Reception of FM Signals, by Amar G. Bose and William R. Short. RLE TR No. 540. January 1989. 56 pp. $7.50.

Adaptive Array Processing for Multiple Microphone Hearing Aids, by Patrick M. Peterson. RLE TR No. 541. February 1989. 125 pp. $9.00.

The annual ***RLE Progress Report No. 131*** will be available in mid-June. ***RLE Progress Report No. 131***, covering the period January through December 1988, contains both a statement of research objectives and a summary of research efforts for each group in RLE. Faculty, staff, and students who have participated in these projects and funding sources are identified within each chapter. The ***RLE Progress Report*** also contains a list of current RLE personnel.

The Research Laboratory of Electronics welcomes inquiries regarding our research and publications. Please contact:
Barbara Passero
Communications Officer
Research Laboratory of Electronics
Room 36-412
Massachusetts Institute of Technology
Cambridge, MA 02139
telephone: (617) 253-2566
telefax: (617) 258-7864

The Future of FM Broadcasting

In a recent RLE Technical Report (TR No. 540), Dr. Amar G. Bose, Professor of Electrical Engineering and Computer Science, and Dr. William R. Short of Bose Corporation Research Staff have published results of a research program on the effects of multipath on FM broadcasting. Specifically, the report focuses on the newly developed FMX technology which claims to double the stereo reception area of FM broadcasting. The FMX system seeks to overcome the multipath effects of standard FM broadcasts by transmitting an extra channel of stereo information which eliminates background noise. In their research findings, Drs. Bose and Short have demonstrated that the multipath effects on current FM stereo equipment were greater with FMX than with current FM stereo transmissions. The report contains mathematical models, analysis, and computer simulations

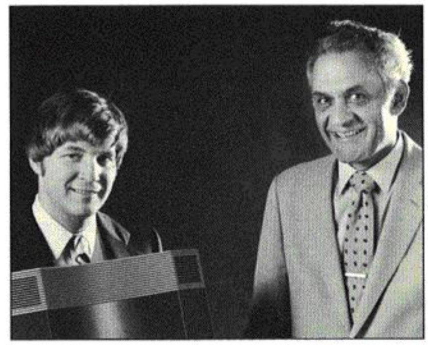


Photo courtesy of Bose Corporation

Dr. William R. Short and Professor Amar G. Bose

based on field tests conducted at MIT's radio station WMBR. The report concludes that FMX actually degrades reception and does not increase the stereo broadcast range. To order a copy of Technical Report No. 540, please contact the RLE Communications Group.

UPDATE: RLE Collegium

The RLE Collegium was established in 1987 to promote innovative relationships between the Laboratory and business organizations. The goal of RLE's Collegium is to increase communication between RLE research and industrial professionals in electronics and related fields.

Collegium members have the opportunity to develop close affiliations with the Laboratory's research staff, and can quickly access emerging results and scientific directions. Collegium benefits include access to a wide range of publications, educational video programs, RLE patent disclosures, seminars, and laboratory visits.

The RLE Collegium membership fee is $20,000 annually. Members of MIT's Industrial Liaison Program can elect to transfer 25% of their ILP membership fee to the RLE Collegium. Membership benefits are supported by the Collegium fee. In addition, these funds will encourage new research initiatives and build new laboratory facilities within RLE.

For more information on the RLE Collegium, please contact RLE or the Industrial Liaison Program at MIT.

RLE *currents*

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The staff of ***currents*** would like to thank Professor Alan V. Oppenheim for his technical guidance during the preparation of this issue.

Inquiries may be addressed to RLE ***currents***, 36-412, Research Laboratory of Electronics, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Cambridge, MA 02139.

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