The Acoustics, Speech, and Signal Processing Society — A Historical Perspective

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ON the occasion of the centennial year of IEEE, each of the Groups and Societies of IEEE have been taking stock of where they stand, and trying to understand how they got there. In this spirit I have been asked to set down in print a historical perspective of how the ASSP Society has grown from a relatively small group (whose main concern was Audio) to one which maintains a broad range of interests in the areas of acoustics, speech, and signal processing (ASSP).

The path taken by the Society has been a tortuous one. The group was the first technical group created in 1949 by one of the IEEE predecessor organizations, the IRE (Institute of Radio Engineers), and was originally called the Professional Group on Audio (PGA). The people most responsible for the PGA in its early years were Ben Bauer, Leo Beranek, John Hilliard, and Dan Martin. Its prime purpose was the dissemination of technical information concerning the newly emerging fields of high fidelity recording and transmission of audio. A rapid perusal of the Transactions on Audio from its inception in 1953 until somewhere in the mid-1960's indicates that the main topics of interest included stereophony, loudspeaker design, magnetic recording techniques, amplifier design, audio frequency noise generation, loudness measurement techniques, microphone standardization, tape mastering, and some early work on speech communications.

In 1963 the IEEE was created as a merger of the IRE with the AIEE (American Institute of Electrical Engineers). In 1965 the group formally changed its name to the IEEE Group on Audio and Electroacoustics (G-AE) to reflect the increasing importance of electroacoustic transducers in the field of audio and high fidelity sound reproduction.

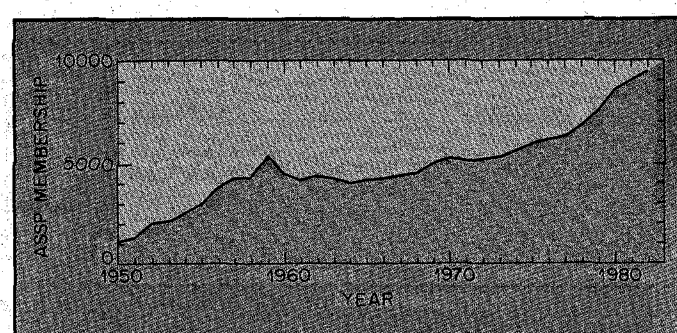

Figure 1. ASSP Membership Statistics from 1950 to 1982.

In the early 1960's the Audio and Electroacoustics Group was going through a difficult period. The field of audio had matured to the point where there were few new ideas of theoretical (or practical) interest, and where most of the earlier theory had been put to practical use by manufacturers who were more interested in selling products than in advancing technology for their competitors. Thus the G-AE transactions became thin and membership in the group began to decline.

By the mid 1960's, the AdCom (Administrative Committee) of the G-AE started to realize that something had to be done to rejuvenate the group or it would die an untimely death. Fortunately several new developments took place at about this same period of time that led to the beginnings of what we have now come to call the ASSP Society. Two technological events occurred that were to radically change the way people thought about two of the most fundamental concepts of linear systems namely spectrum analysis and filtering of signals.

The first technological event was the discovery ("rediscovery") of an algorithm for computing the discrete Fourier transform (DFT) of a finite sequence of N samples that required on the order of $N \log_2 N$ operations (multiplications plus additions) instead of the N^2 operations as required by a conventional direct DFT evaluation. Of course the people most responsible for this new algorithm, subsequently misnamed the FFT algorithm (it should have been called the Fast Discrete Fourier Transform, FDFT), were Jim Cooley of IBM and John Tukey of Bell Laboratories who published a classic description of the algorithm in the *Mathematics of Computation* in 1965 [1]. However, as is the case with many technological breakthroughs, 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 of MIT Lincoln Laboratory. Rader and Stockham 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. This work culminated in the classic engineering presentation of the FFT in the book by Gold and Rader [2].

The second technological event (set of events) which occurred in the early 1960's was the discovery that it was both theoretically and practically feasible to process signals in the digital domain in much the same way as they previously had been processed in the analog domain, i.e., one could design and implement (on a digital computer as hardware was still on the far horizon) a digital system that would provide a desired type of filtering on a given input signal. Interestingly enough the impetus for studying this type of problem arose from research into speech processing systems where people wanted to investigate newly proposed ideas for processing speech signals to reduce the required bandwidth for transmission and storage. With analog processing one had to actually build the system to determine whether a new idea had any merit. Typically, the turnaround time between generation of an idea and testing was 1–3 years. Hence the speech researchers were highly motivated to devise a simulation method for testing new ideas for speech processing systems. The two laboratories most involved in this technological breakthrough were Bell Laboratories, where Jim Kaiser and Roger Golden were devising practical techniques for designing digital filters which modeled classical analog designs as well as novel techniques for designing digitial filters without an analog prototype (i.e., the window design method), and MIT Lincoln Laboratory where Ben Gold and Charlie Rader were simultaneously doing similar pioneering work. This work culminated in a series of classic filter design papers [3–5]. A merging of the ideas of digital filtering and the use of the FFT led to a set of fast convolution algorithms by Stockham [6] and Helms [7].

The fledgling field of digital signal processing became known to several technical groups in the IEEE and by 1965 several of the early researchers were starting to look for an IEEE group which would support this new field of research. The AE group, with the foresight and perspicacity of Bill Lang at IBM, deliberately and aggressively sought out this new technology. To promote the FFT algorithm and its applications to spectral analysis, the group cosponsored a special workshop at the Spring Meeting of the Acoustical Society of America in Boston in 1966. This workshop, entitled "Uses of Power Spectrum Analysis," was one in which several prominent researchers described practical applications of the FFT in a wide variety of areas (including one entitled "Spectral Analysis of the Call of the Male Killer Whale" [8]). The papers presented at this workshop were published in the first special issue of the G-AE transactions in June 1967. Also published in this special issue was the classic tutorial paper "What is the Fast Fourier Transform" [9], written by several members of the G-AE Subcommittee on Measurement Concepts.

The aggressive efforts of the G-AE to attract technical work in the new field of digital signal processing led to a heavily increased commitment to the application area of speech processing. As mentioned earlier, the vast majority of the early signal processing researchers were involved in speech processing work. Hence it was a natural outgrowth to encompass this well-established field into the G-AE. To promote the group's interest in speech processing, the G-AE cosponsored, with the U.S. Air Force, an International Conference on Speech Communication and Processing, held in Cambridge, Massachusetts in November 1967. This conference, attended by about 400 people, was the first large scale conference held by the G-AE, and formally gave notice to the speech processing community that the G-AE was interested in technical achievements in this area. Continuing its aggressive policies, two special issues of the ASSP Transactions were published (March and June 1968) containing a good sampling of the papers presented at the speech conference.

Although the G-AE had several technical subcommittees in the early 1960's, their function was primarily to define standards on how certain measurements (generally re-

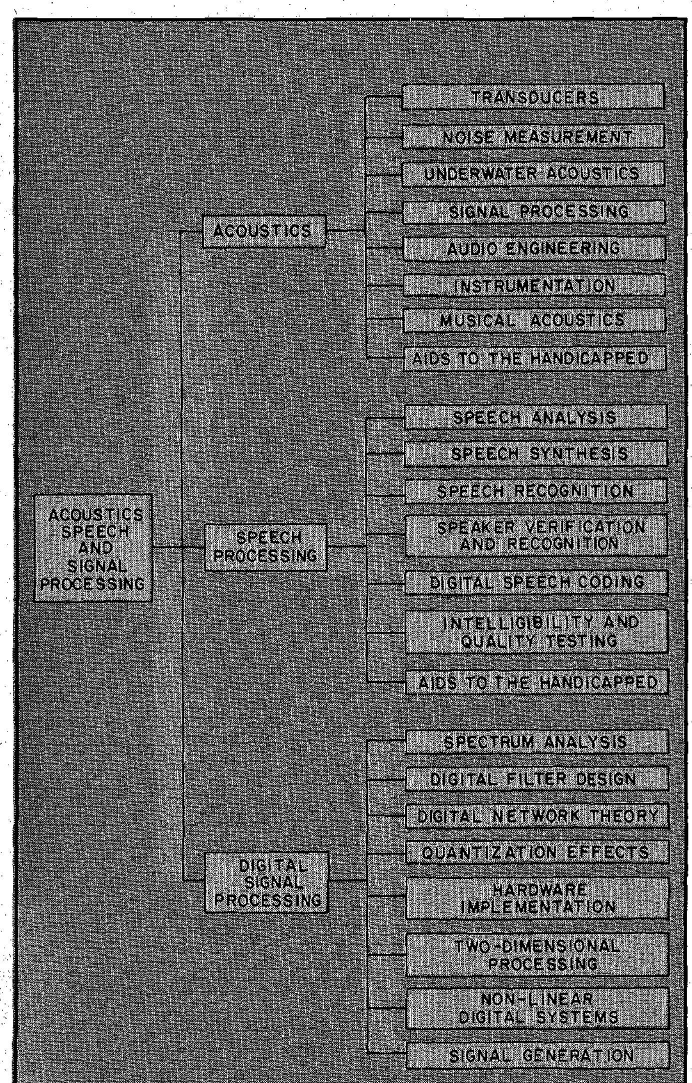

Figure 2. Major Technical Areas of ASSP Society

lated to equipment for hi-fi systems) were to be made. One of these committees, the Subcommittee on Measurement Concepts, decided to expand its role in two ways. First it worked on a tutorial paper on spectral analysis [10]. Second it decided to try to bring together in one meeting hall, a large body of researchers who were interested in the many aspects of the recently discovered FFT algorithm. Again with the genius and guiding inspiration of Bill Lang, the Arden House Workshop on the FFT took place in October 1968 with 100 experts exchanging thoughts, ideas, and insights into the myriad of different aspects of this seemingly straightforward algorithm. From this workshop came another special issue of the ASSP Transactions (June 1969) in which many of the key papers presented at the Arden House meeting were published. In addition a most informative and entertaining exposition on the history of the FFT algorithm was published in the form of a series of recollections by five of the people most heavily involved in the initial development and applications of the FFT [11].

By mid 1968 the G-AE was indeed rejuvenated and flourishing. The Transactions were growing rapidly in size and quality and the group was continually attracting new people with new ideas of how the group should keep growing. It had become quite clear that the fields of digital signal processing and speech communication had grown substantially and that some organized structure was required to be responsible for nurturing future growth and change in these areas. Thus the technical committees on digital signal processing and speech communication were formed. The digital signal processing committee was formed as an outgrowth of the measurements committee with a drastic change in membership. The speech communication technical committee was formed from scratch. Somewhat smaller technical committees in the areas of electroacoustics and underwater sound were also formed but they tended to play less active roles in the growth of the G-AE in the period from 1968 to the present.

The digital signal processing (DSP) committee immediately assumed responsibility for the highly successful Arden House Workshop and subsequently held 3 additional such meetings. The dates of these Arden House Workshops were January 1970, January 1972, and January 1974, and the topic area of all three meetings was the broad area of digital signal processing. Each of these meetings was highly successful in attracting a full contingent of dedicated and energetic researchers (worldwide) who presented new and interesting ideas on how the signal processing field was being advanced. Among the major innovations presented at these three workshops were:

1. An efficient hardware implementation of a digital filter [12].
2. The frequency sampling FIR design method [13].
3. A comprehensive analysis of the effects of finite precision implementation of IIR digital filters [14].
4. The first computer aided design algorithm for IIR filters [15].
5. The introduction of the concept of block floating point arithmetic [16].
6. The introduction of the concept of designing optimal FIR digital filters [17, 18].
7. The introduction of network concepts to digital filter structures [19].
8. The introduction of number theoretic transforms [20].
9. The extension of filter design techniques into 2 dimensions [21, 22].
10. The introduction of the ideas behind decimation and interpolation [23–25].
11. The introduction of lattice filter structures [26].
12. The introduction of advanced linear prediction techniques [27].

The above list represents only some of the highlights of work presented at these three exciting Arden House Workshops. To make the research results more widely available, special issues of the group transactions in June 1970, October 1972, and June 1975 were devoted to the Arden House papers.

The DSP committee, which in the early 1970's was primarily composed of researchers from the Bell Laboratories, Princeton, IBM, MIT community (and who met about 6 times per year) was not content with just the job of running Arden House workshops. Based on the success of the forerunner committee with the tutorial paper on the FFT (as well as a subsequent tutorial paper "On Digital Filtering" [28] in the September 1968 transactions), the DSP committee chose the ambitious task of defining a standardized set of terminology for use in DSP papers. This task was no easy one, since, as is the case with almost all new fields, the field was growing rapidly and in a somewhat uncontrolled manner. The people working in the field had come from several disciplines including mathematics, communications, circuit theory, hardware, computer architectures etc., and each had brought their own set of terminology associated with their previous work. The DSP committee attempted to pull together this vast body of work and to unify the terminology to promote communication among the disciplines. This work culminated in the paper "Terminology in Digital Signal Processing" [29] published in the December 1972 transactions.

The DSP Committee took on one other major responsibility in the early 1970's. About this time the IEEE had formed an internal publishing group called the "IEEE Press" whose charter was to publish collections of reprints of papers which were of historical or continuing technical interest to a wide range of readers and which were not readily available in any other form. Because of the rapid growth in interest and membership in the AE group, some of the earliest collections of papers in signal processing had become virtually extinct, since all back copies of the G-AE transactions had long since been dispersed. The G-AE was one of the first groups asked to organize a suitable collection of reprints for an IEEE Press book. This task was taken on by the DSP Committee and the result was the DSP Green Book [30] collection of papers published

Table I
Classification of ASSP Areas of Interest
(Courtesy of Jont Allen)

1. ACOUSTICS
- 1.1 DIGITAL AUDIO
  - 1.1.1 Equalization
  - 1.1.2 Speaker Design
  - 1.1.3 A/D and D/A Technology
- 1.2 UNDERWATER
  - 1.2.1 Sensor Arrays
  - 1.2.2 Applications to Geophysics
- 1.3 ELECTROACOUSTICS
  - 1.3.1 Transducers

2. SPEECH
- 2.1 SPEECH TRANSMISSION AND CODING
  - 2.1.1 Narrow-band
  - 2.1.2 Medium-band
  - 2.1.3 Wide-band
  - 2.1.4 Channel noise and distortion effects
  - 2.1.5 Encryption
- 2.2 ENHANCEMENT AND NOISE REDUCTION
  - 2.2.1 Noise suppression
  - 2.2.2 Dereverberation and echo-cancelling
- 2.3 ANALYSIS AND RECONSTRUCTION
  - 2.3.1 Short-time spectral analysis (LPC, cepstral, etc.)
  - 2.3.2 Parameter Estimation
    - 2.3.2.1 Formant analysis
    - 2.3.2.2 Pitch and excitation waveform detection
    - 2.3.2.3 End-point and V/UV detection
    - 2.3.2.4 Area function analysis
  - 2.3.3 Spectrum and rate modification
- 2.4 SYNTHESIS
  - 2.4.1 Synthesis-by-rule
  - 2.4.2 Synthesis from stored units
  - 2.4.3 Synthesis from articulatory models
- 2.5 RECOGNITION
  - 2.5.1 Discrete utterance recognition
  - 2.5.2 Connected word recognition
  - 2.5.3 Continuous speech recognition
  - 2.5.4 Speech understanding, syntax, and semantics
  - 2.5.5 Word spotting
  - 2.5.6 Performance evaluation of speech recognizers
  - 2.5.7 Speaker identification and verification systems
  - 2.5.8 Language identification
- 2.6 PRODUCTION/SYNTHESIS
  - 2.6.1 Physical models
    - 2.6.1.1 Vocal tract models
    - 2.6.2.2 Vocal cord models
  - 2.6.2 Articulatory measurements
- 2.7 SUBJECTIVE EVALUATION
  - 2.7.1 Speech perception
  - 2.7.2 Measurement of speech intelligibility
  - 2.7.3 Measurement of speech quality
- 2.8 AIDS-FOR-THE-HANDICAPPED
  - 2.8.1 Hearing aids
  - 2.8.2 Speech displays and translators
  - 2.8.3 Measurement and diagnosis of speech pathologies

3. SIGNAL PROCESSING
- 3.1 ONE-DIMENSIONAL DSP
  - 3.1.1 DFT and Other Transforms
  - 3.1.2 Fast Algorithms
  - 3.1.3 Quantization Effects
  - 3.1.4 Spectral Analysis
  - 3.1.5 System Identification (Signal Modeling)
    - 3.1.5.1 Parametric Methods
      - 3.1.5.1.1 Adaptive Filtering
      - 3.1.5.1.2 Least Squares Methods
    - 3.1.5.2 Nonparametric Methods
  - 3.1.6 Filter Design and Applications
    - 3.1.6.1 FIR
    - 3.1.6.2 IIR
    - 3.1.6.3 Time Varying Filters
    - 3.1.6.4 Deconvolution
    - 3.1.6.5 Singular Value Decomposition Methods
  - 3.1.7 Applications
    - 3.1.7.1 Echo Cancellation
    - 3.1.7.2 Communication Networking
    - 3.1.7.3 Aids for the Handicapped
  - 3.1.8 Detection and Estimation
    - 3.1.8.1 Time Delay Estimation
  - 3.1.9 Radar
- 3.2 MULTI-DIMENSIONAL DSP
  - 3.2.1 Image Coding
  - 3.2.2 Image Processing
  - 3.2.3 Multi-Dimensional Spectral Analysis and Synthesis
  - 3.2.4 Filter Design and Applications
- 3.3 VLSI AND HARDWARE IMPLEMENTATIONS
  - 3.3.1 Analysis-Synthesis
  - 3.3.2 Coding
  - 3.3.3 Filtering
  - 3.3.4 Speech Recognition
  - 3.3.5 Speech Synthesis

Table II
Past Presidents of the ASSP Society

Year	President
1949–50	L. Beranek
1951–52	B. Bauer
1953	M. Camras
1954	V. Salmon
1955	W. Kock
1956	D. Martin
1957	H. Olson
1958	F. Slaymaker
1959	A. Bereskin
1960	H. Knowles
1961	C. Harris
1962	R. Benson
1963	F. Comerci
1964	W. Ihde
1965	I. Kerney
1966	D. Brinkerhoff
1967–68	W. Lang
1969–70	J. Flanagan
1971–72	R. Kaenel
1973	J. Bouyoucos
1974–75	L. Rabiner
1976–77	H. Helms
1978–79	R. Schafer
1980–81	C. Rader
1982–83	N. R. Dixon

in December 1972. The DSP committee became so fascinated by the potential coverage of the IEEE Press (because of its broad membership and low costs of publication) that it decided to request the IEEE Press to expand its charter by publishing a book which was a bibliography of books and papers in the field of digital signal processing. This request was received favorably by IEEE Press and in January 1973 the DSP Red Book [31] on literature in DSP was published. The book was subsequently updated in 1975 [32], and a supplement was issued by ASSP Society itself in 1979 [33].

The initial success of the IEEE Press reprint collection led the DSP committee to gather titles for a second DSP reprint volume (the Blue Book) which was published in 1976 [34]. About the time of publication of the Blue Book, the DSP Committee felt that the DSP field had matured sufficiently that it was time to try to publish a set of algorithms (in the form of portable FORTRAN code) in the areas of filter design, spectral analysis, filter implementation, finite word length effects, linear prediction, and multirate systems. The IEEE Press was again requested to experiment with a new form of publication. Again the request was favorably received by IEEE and in 1979 the DSP Program book was published [35] along with an option to purchase a computer tape with the source code of all published programs. The success of these five ASSP Society IEEE Press books in DSP is well known (in fact sales of these ASSP volumes accounted for 10% of all cumulative IEEE Press sales as of the end of 1982) and was, of course, a constant source of pride and encouragement to the DSP Committee.

The second major technical committee, the one on Speech Communication, got off to a slow start since it had no predecessor committee on which to build. In its early years its prime function was to organize a major conference on Speech Communication, held in April 1972 in Newton, Massachusetts. Selected papers from this speech meeting were published in a transactions special issue in June 1973. The speech committee also took a role in stimulating IEEE Press reprint books (although they didn't work on them as committee projects), ultimately leading to four speech-related reprint volumes [36–39]. Perhaps the most important role played by the speech committee was the initiation of the ICASSP meetings which today play such a large role in the society. This happened as an outgrowth of the plans of the Speech Committee to hold another Speech Communication meeting in 1976 in Philadelphia. The committee, with a bit of arm twisting by appropriate Ad-Com members, agreed to expand the function of the meeting into a general ASSP Conference. The initial meeting was a great success and the tradition of ICASSP meetings was begun.

The period from 1972 to the present has been marked by

Table III
Recipients of Major ASSP Awards, 1954–1982

Year	Society Award	Achievement Award
1954		B. Bauer
1955		H. Olson
1956		H. Roys
1957		M. Camras
1958		D. Martin
1959		A. Bereskin and P. Goldmark
1960		W. Snow
1961		J. Macdonald
1967	H. Dudley*	M. Corrington
1968		B. Knowles
1970		W. Lang
1971		J. Flanagan
1972	F. Cooper*	B. Gold
1975	J. Flanagan	B. Atal, J. Cooley
1976		C. Rader, A. Oppenheim
1977	H. McDonald	J. Kaiser, A. Gray, J. Markel
1978	H. Schuessler	L. Rabiner
1979	A. Oppenheim	R. Schafer
1980	L. Rabiner	T. Parks
1981	J. Kaiser	K. Steiglitz
1982	R. Schafer	J. Makhoul

*Forerunner of Society Award called Pioneer in Speech Communication Award several milestones in the way in which the Society has been conducting business. In August 1972 the Society began publishing six issues of the transactions per year (rather than four as done previously) to reflect the desire to be somewhat more timely and to accommodate the increasing number of papers being sent to the journal. In February 1974 the group name changed from Audio and Electroacoustics (AE) to Acoustics, Speech, and Signal Processing (ASSP) as it is called today. This change reflected a change in charter that had begun almost a decade earlier.

The last major structural change in ASSP occurred on January 1, 1976 when the group officially became a society. To understand the significance of this event one must understand the climate of the IEEE in the early 1970's. The Computer Group, which was the largest group in the IEEE, was interested in obtaining a higher degree of status and independence than that normally accorded an IEEE Group. Hence the IEEE created the category of the affiliated Society. To attain Society status, a group had to demonstrate that it met certain requirements on Technical Activities, Publications, Technical Meetings, Membership, Finances, and Long Range Technical Planning. The fact that the ASSP Group had met these conditions by 1976 showed just how far we had come since the early 1960's.

The period from 1976 to the present has been marked by a steady growth in membership. (Fig. 1), publication, and scope of the Society. The major technical areas of the ASSP Society, as we know it today, are shown in Fig. 2. All three arms of the society are flourishing. The acoustics area, the oldest segment of the society, has taken on new life in the past decade with advances in audio and instrumentation, and in the area of underwater acoustics. Speech processing is on the verge of attaining even higher prominence in the society with advances in technology taking research systems out of the laboratory and into the commercial world. Digital signal processing has attained maturity and today the activity often lies in specialized areas such as VLSI structures, two-dimensional theory, modern spectral analysis techniques, multidimensional processing, etc. Based on the 1982 and 1983 ICASSP meetings, a classification of the societies major areas of interest, courtesy of Jont Allen, is given in Table I. To round out this historical picture, Tables II and III list the past presidents of the ASSP Society since its inception in 1949, and the recipients of the Major ASSP Awards since 1954.

As we look to the future we can only see increased growth in all aspects of our society. The ASSP Society has matured but our period of dynamic growth has brought us to the threshold of a new era in signal processing. I personally look forward to seeing where it will eventually take us.

ACKNOWLEDGMENTS

As with any historical account, this paper suffers from errors of omission and commission. In an attempt to simplify matters I have singled out only a few individuals and a few research projects that have had a major impact on the growth of the society and its technical fields. However all work builds on previous experience and many important signal processing ideas were proposed independently and before the two milestones mentioned in this paper.

Several individuals have helped in providing comments and criticisms of this paper. These include Drs. James Flanagan, Aaron Rosenberg, Steve Levinson, Jim Kaiser, Jont Allen, N. S. Jayant, Ronald Schafer, Charlie Rader, Ben Gold, and Bill Lang. I thank each of these individuals for their help. Special thanks go to Charlie Rader for providing a detailed set of recollections of the workings of the ASSP Society in the 1960's.

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