Meeting Summaries

The First Annual Dallas Chapter Workshop On Signal Processing Technology

(by J. Picone, J. Naik)

The first annual workshop on signal processing technology, sponsored by the Dallas Chapter of the IEEE ASSP Society, was held on April 20, in Richardson, TX. Fifty participants attended a one-day course titled "The TMS 320: A Powerful Programmable Signal Processing Chip." Three universities and ten companies were represented at this workshop. The goals of the workshop were to introduce the novice to a digital signal processing chip architecture and instruction set, and discuss software and hardware issues associated with several basic applications in digital signal processing. This particular workshop focused on the TMS32010 digital signal processing chip.

The morning session dealt primarily with software issues. The session began with an overview of the chip architecture. This was followed by a detailed discussion of the instruction set of the device. Several coding exercises demonstrating such applications as digital FIR filter implementations gave the participants an opportunity to gain practical coding experience. The morning session concluded with a discussion of techniques for pipelining instructions.

The initial portion of the afternoon session was devoted to a discussion of the various development tools available for the device, followed by a demonstration of the Digital Filter Design Package marketed by Atlanta Signal Processors, Inc. Next, hardware interface techniques were demonstrated, with a focus on I/O issues. Finally, the technical program concluded with a presentation on the TMS32020, the second generation in the TMS 320 family. This was followed by a lively, informal discussion on the future trends in digital signal processing chip technology.

There was an overwhelming positive response from the participants to have the Dallas Chapter host future workshops along these lines, and we are currently making plans to institute an annual workshop. Any chapters interested in presenting a similar workshop to their members should contact Dr. Joseph Picone, The Dallas Chapter of the ASSP Society, Texas Instruments, Inc., P.O. Box 225016, MS 238, Dallas, TX 75266.

1985 International Conference on Acoustics Speech and Signal Processing March 26–29, 1985

Vijay K. Jain

bout 1400 attendees from all over the world participated in the tenth International Conference on Acoustics Speech and Signal Processing. Held March 26–29 at the Hyatt Regency and Hilton Hotels in Tampa, Florida, ICASSP-85 was the first four-full-days conference in its history. General Chairman Rex Dixon planned and succeeded in making it an 'attendee-oriented' conference.

From over 800 submissions, approximately 480 papers were selected by the technical program committee headed by Vijay Jain and composed of the members of the various ASSP technical committees and their chairmen. Parallelism of sessions was minimized and, with only minor exceptions, an attendee could attend all of the sessions in a major area: General Signal Processing, Spectrum Estimation and Modeling, Speech Processing, Multi-dimensional Signal Processing, VLSI, or Underwater Acoustics.

Harry Nienhaus was the Finance Chairman, and Dave Kiewit the Exhibits Chairman. More than 35 exhibits displayed the state-of-the-art in Signal Processing, speech, and other hardware and software. Registration was handled by David Snider, local arrangements by Allen Gondeck, and publicity was conducted by Donald Childers. Their superb efforts contributed greatly to the smooth running of ICASSP-85.

A summary of the technical program prepared by Jae Lim, Jim Cadzow, Vishu Viswanathan, John Woods, Peter Cappello, and Jack Ianniello is given below for each technical area.

Numbers and names in parentheses indicate session numbers and chairpersons.

DIGITAL SIGNAL PROCESSING

(2) **Digital Filter Design**
(J. F. Kaiser)

Various aspects for designing FIR, IIR, and other special filters such as the wave digital filters, filter banks, and median filters were presented. The major considerations were new insights, the design of filter structures that could be efficiently implemented, and the design of filters to meet additional passband and stopband constraints with prescribed arbitrary precision. The depth and range of the papers reflected the maturity this topic has obtained.

(12) **Deconvolution and Bandlimited Extrapolation** (T. Huang)

Two papers discussed deconvolution with the L_1 norm. Two more papers suggested time-domain deconvolution, one a conjugate gradient approach and the other a state-space approach. Other deconvolution papers dealt with noncoherent data, sequency domain theory, fuzzy set theory, and the use of known input-output data in lieu of a system model. Two extrapolation papers were presented, one comparing these methods and the other generalizing on existing methods.

(14) **Time-frequency Analysis and Synthesis** (L. Cohen)

The majority of papers were on the design and theory of analysis/synthesis filters. An interesting general framework, unifying understanding for the many previously proposed methods for analysis/synthesis of filter banks, was presented. A new connection between filter/detectors and the short-time Fourier transform magnitude was shown to provide an understanding and comparison of the spectrogram, the sliding DFT and other such methods.

(20) **Fast Transforms** (S. Burrus)

Efficient architectures/algorithms using prime factors were described for the DFT, the discrete Hartley transform, and the DCT. A family of radix-2 decimation-in-frequency algorithms were discussed for the DCT and the DST. Also discussed the implementation of "split-radix" length-2^n DFT algorithms, a realizable lower bound on the number of real multiplications required for a length-2^n complex DFT, and reductions in the arithmetic complexity of fast finite field algorithms.

(30) **Adaptive Filtering I**
(B. Widrow)

A new method for stabilizing the fast RLS algorithm, and an efficient step size adaptation technique were presented. Several papers addressed the convergence and noise capture properties of the constant modulus algorithm. Aspects of lattice structure adaptive filters were also discussed.

(32) **Adaptive Filtering II**
(M. Bellanger)

Parallel algorithms were presented for multiplication and inversion of Toeplitz matrices, and the solution of Toeplitz systems from an implementation point of view. An efficient block processing Fast Transversal filter was described. The Echo cancellation and line equalization problems were used as two typical applications for refined adaptive algorithms. A particular structure was shown and demonstrated to be quite efficient for noise cancelling. Also, the potential advantage in using Differentially Coded Signals was underlined. A unifying approach of adaptive systems as optimal processors was also discussed.

(42) **Signal Processing Hardware and Software** (F. Mintzer)

Many of the papers involved application of the commercially available general-purpose DSP chips. There were papers both on configuring high-performance systems, and on managing them in software. Multiprocessor architectures included a single instruction, multiple data architecture, a 100–280 MFLOPS parallel processing system, and a 10.4 GOPS system. The session was testimony to the fact that we may be entering an era where hardware is not the limiting consideration even in video processing applications.

(44) **Quantization and Nonlinear Systems** (L. Jackson)

Several trends were evident. The factored state-variable representation provides an analytic approach to represent and generate new filter structures. The idea for system theory of "balanced" state-space realization was shown to lead to low-noise and low-sensitivity structures. In addition, there was continuing interest in quantization effects in adaptive filter algorithms.

SPECTRUM ESTIMATION AND MODELING

(3) **Spectrum Analysis** (J. A. Cadzow)

Three papers discussed adaptive versions of Pisorenko's harmonic retrieval method. One paper extended the so-called MLM to cross-spectrum analysis, while another extended the Gershberg-Populis algorithm to spectrum analysis of time series with missing data. A paper suggested a class of data windows based on B-splines, and one presented asymptotic statistics on estimates of an AR(1) process. Two other papers considered minimum-energy extension of a limited autocorrelation sequence and a multichannel AR technique for single channel data.

(9) **Parameter Estimation**
(D. W. Tufts)

The lead paper presented the strong points and limitations of the use of the L_1-norm in parametric spectral analysis. An interesting paper presented the use of branch-and-bound techniques for the solution of nonlinear least-squares problems. The problem of approximations to eigenvectors and eigenvalues was also addressed in this session. Another interesting paper discussed an approximate factorization for unfactorable spectral models. Other papers addressed the generation of ARMA auto and cross covariance sequences and the detection of signals using AR models.

(17) **Estimation and Performance**
(L. Marple)

Three papers in spectral estimation dealt with the potential loss in spectral accuracy when using sample covariances, classification of narrowband signals, and sensitivity analysis of an SVD approach for estimating sinusoids. Next, signal subspace approach to the angle-of-arrival problem for wideband sources, and in the presence of sensor errors, was discussed in separate papers. New results in multivariate AR models and in detecting non-stationarities in an AR process were presented. In the last two papers AR modeling in the presence of additive noise, and the use of Bayes' recursion formula when noise is non-Gaussian were discussed.

(21) **Multi-channel and Multidimensional Spectral Analysis**
(R. Yarlagadda)

The lead paper presented a simple sufficient condition for extendibility and its application to 2-D spectrum estimation. The second paper presented a 2-D Prony method where the frequencies are obtained by computing the eigenvalues of a tridiagonal matrix. A multichannel AR spectrum estimation method was then discussed and shown to be optimum in the reflection coefficient domain. Derivation of simultaneous confidence bands for a class of 2-D noncausal parametric spectral estimates was also addressed. Other papers discussed the performance of frequency-wavenumber spectral estimators in a case of smooth, broad spectra, and universal characterization of m-d maximum entropy covariances.

(27) **Modeling of Time-Varying Signals**
(S. Kay)

Several papers dealt with the properties and applications of the Wigner-Ville distribution: its optimality for detection, its expansion into a singular system framework, and application to signal synthesis; also discussed was a band-selectable (or zoom) discrete Wigner distribution. Other papers dealt with innovations on bilinear signal representation, generalized ambiguity function, and stable time-varying models through log-area ratios.

SPEECH PROCESSING

(2) **Isolated Word Recognition I** (B. T. Lowerre)

Four papers dealt with application of hidden Markov models for recognition and training. Three papers dealt with speaker-independent recognition by means of phonetic segmentation, instead of using dynamic programming for time alignment. One paper was on a study of different distortion measures. There was also a paper on an isolated word recognition approach that used matrix quantization.

(7) **Narrowband Speech Coding** (B. S. Atal)

The lead paper presented a 300 bit/s segment vocoder that reconstructs its output by concatenating stored segments of speech waveform after appropriate time and pitch scaling. Of the remaining papers presented, two use the line spectrum pairs representation of LPC; three use vector quantization; three use the approach of multipulse-excited LPC; and one uses Markov-Huffman encoding of LPC parameters.

(11) **Speaker Recognition and Pitch Extraction** (C. L. Wayne)

In the speaker recognition area, one paper dealt with text-independent identification over telephone channels, one with recognition using vocoded speech, two with vector quantization approach, and one with a prototype access control system. In the pitch extraction area, two papers compared the performance of various algorithms; one presented a real-time implementation; and one presented a robust algorithm for noisy speech. There was also one paper on cochlear prosthesis, one on the effect of speech level on intellegibility testing, and one on objective measures of speech quality.

(13) **Speech Analysis and Reconstruction** (S. Seneff)

For multipulse LPC, one paper presented a frequency-domain formulation and another presented a new all-pole modeling method. Two approaches were presented for time-scale modification of speech; one based on a sinusoidal representation of speech and the other involving a new overlap-and-add procedure; the first approach also facilitates frequency scaling and pitch scaling. The remaining papers proposed different analysis/synthesis approaches: robust LPC analysis of noisy speech; LPC analysis using the L_1 norm; Fourier-Bessel representation of speech; time-dependent all-pole modeling; perceptually based spectrum estimation; and a generalized excitation model that allows each harmonic to be voiced or unvoiced.

(35) **Topics in Estimation and Radar** (W. F. Gabriel)

The lead paper discussed the prediction problem of a stochastic signal when constraints are imposed on the filter's impulse response. The second paper presented a new method for computing the magnitude squared coherence by estimating the parameters of a rational model. An AR model-order selection criterion was then addressed via significant reflection coefficients. Another interesting paper discussed an extrapolation algorithm based on a minmax criterion. Other papers addressed the use of AR models for bispectrum estimation and the polar-to-Cartesian interpolation problem.

(39) **System Identification** (B. Friedlander)

Bounding the response envelope of linear systems with bandlimited inputs was considered. Three papers presented work on identification algorithms: lattice recursions for the instrumental variable algorithm; a two-stage predictor applied to ARMA processes; and efficient linear phase system identification. Eigenstructures were analyzed in two papers: a statistical analysis of the SVD approach to determining effective rank; and a computation of the expected resolving power of the MUSIC algorithm for finite data. Another paper analyzed the effects of misestimation of the support bandwidth on the bandlimited extrapolation problem.

(19) **Speech Enhancement and Synthesis** (M. M. Sondhi)

The six speech enhancement papers dealt with different ways of combatting noise and interference, including speech transduction with multiple sensors, adaptive noise cancelling, robust all-pole modeling, and steering of microphone arrays to seek and track active talkers. Four of the five synthesis papers discussed synthesis by rule in terms of different sound units (diphones and demisyllables) and in terms of different representations (formant synthesizer, multipulse LPC, and articulatory model). The fifth paper treated the problem of converting the speech of one speaker to sound like that of another.

(23) **Isolated Word Recognition II** (G. E. Kopec)

Speaker-independent and speaker-trained systems, both small and large vocabularies, and both clean and noisy speech, were addressed. Four papers were explicitly concerned with performance evaluation, and discussed either large-scale experiments testing a particular system or general approaches to predicting performance. One particularly thorough evaluation of a system for telephone speech recognition involved about 2000 talkers. Three papers directly addressed the problem of large vocabulary recognition. Notable among these was a workstation-based real-time implementation of the IBM 5000-word office correspondence system.

(25) **Mediumband Speech Coding I** (M. Berouti)

Papers presented spanned bit rates from 4.8 kbits/s to 16 kbits/s. The first two papers presented coding methods that use the stochastic excitation model. Of the remaining papers, one presented a sinusoidal representation of speech; five discussed different computational approaches to multipulse-excited LPC; and two dealt with baseband coders.

(29) **Vocal Tract and Speech Analysis** (D. G. Childers)

Three papers examined the glottal source and presented methods for various measurements. Two papers dealt with vocal tract shape estimation from acoustic measurements. Another paper described a new articulatory model for speech production. Nine papers were presented on speech analysis, covering various topics including modeling of speech processing in the auditory periphery, cubic spline modeling of speech spectra, a new form of Kalman estimator, a recursive method for estimating excitation pulses, and formant analysis in helium speech.

(31) **Continuous Speech Recognition** (M. Pichney)

Papers were presented giving results for small vocabulary continuous speech recognition tasks. Most papers were interested in actual speech recognition, with a couple of papers on segmentation and one on keyword recognition. Most papers presented improvements to network-based recognition methods: DP matching or Markov modeling, with one paper presenting an expert system approach. Two of the papers presented speaker-independent results, with the rest giving speaker-dependent results.

(37) **Speech Processing Hardware Implementation** (J. J. Wolf)

Programmability was shown to be of major importance in all implementations presented in this session. A programmable signal processor was the vehicle of implementation of a voice messaging system, a 32 kbit/s ADPCM/PCM transcoder, a silence-detecting subband coder, and an LPC-based speech recognition feature measurement chip. Several signal processors were used in a real-time implementation of the new CCITT 32 kbit/s ADPCM standard, an architecture for real-time vector quantization of speech, and an architecture for speech recognition data flow machines. There was also a paper describing a multiple (100) microprocessor parallel architecture for speech recognition.

MULTIDIMENSIONAL SIGNAL PROCESSING

(4) **Single-frame Image Coding** (J. Biemond)

The first three papers covered a predictive approach for hierarchical line encoding, an ordering predictive coding method, and a hybrid image encoding technique, respectively. The next three papers on vector quantization elicited a lot of questions from the attendees. A fast matrix quantizer was then presented which attempts to incorporate a model for the human visual system. This was followed by an AT-based tree encoding procedure with reduced computational complexity, and a simple hybrid image encoding technique which preserves edge information.

(10) **Image Sequence Coding** (B. G. Haskell)

Most of the papers in the Image Sequence Coding session dealt with some aspect of motion compensated interframe prediction algorithms for full motion video. The prediction error is then coded and transmitted using various intraframe coding methods, which also try to adapt to the amount of motion in the scene. Videotapes were shown of video scenes coded at 50 kbs and 256 kbs. Quality was reasonably good as long as the speed of motion was not excessive.

(18) **Image Estimation and Restoration** (R. W. Schafer)

A group of three papers was concerned with nonlinear smoothing filters which preserve edges. Two papers were concerned with reconstruction from projections and holograms. In the image restoration and enhancement area, two papers described new methods of estimating the blurring psf. Two papers were presented on Kalman filtering approaches to noise-smoothing and deblurring. Two papers were presented on iterative techniques with new constraints. Two papers dealt with the use of optimization methods in image restoration. Finally a multiband least-squares technique was presented for image enhancement.

(24) **Feature Extraction, Segmentation, and Scene Analysis** (J. T. Tou)

Fifteen high quality papers were given in this session covering new approaches to feature extraction, image segmentation, object detection, motion analysis, 3-D object representation and shape estimation, and a knowledge-based systems approach to image processing. The papers reflected the advances in this subject matter.

(28) **Reconstruction and Computerized Tomography** (M. Hayes)

Reconstruction from zero (threshold) crossings, and incorporation of moment bounds on the image into the iterative POCS method were discussed. The stability of the discrete phase retrieval problem was treated. In the fourth paper, a bandlimited signal extrapolation method allowing linear constraints was presented. Papers in computer tomography (CT) included image reconstruction incorporating the conversion of fan beam to parallel beam projections, nearest neighbor and inverse distance interpolation for polar to rectangular conversion in Fourier-domain based CT reconstruction, and application of CT to reconstruct the beating heart.

(34) **Multidimensional Filtering** (J. Woods)

A diverse group of papers was presented on a wide range of problems in multidimensional filter design and application. Included were two papers on the nonlinear median and morphological filters. There were two papers on IIR filter realization, two papers on separating YIQ signals in NTSC encoded images, one paper on lattice-filter models for 2-D random fields, one paper on symmetery in FIR filter design, and one paper on image rotation where filters can be employed to reduce aliasing distortion.

(40) **Image Processing Hardware, and M-D Transforms** (R. M. Mersereau)

In the hardware sub-session, papers described an associative memory module for pattern recognition and a pipeline architecture for computing projections of images. A paper presented an architecture for implementing median filters and a high-speed architecture for 2-D recursive filters. The transforms sub-session dealt with a vector radix algorithm for the fast Hartley transform, an implementation for the prime factor DFT on an arbitrary lattice, fast 2-D DCT algorithms based on polynomial transforms, and the computation of Hankel transforms.

VLSI FOR SIGNAL PROCESSING

(6) **VLSI Signal Processors** (G. Edwards)

The papers discussed chips and chip sets that vary in process technology, arithmetic (e.g., fixed-point, floating point, and the residue number system), and target application area. These papers indicate rapid advancement in signal processors.

(8) **VLSI Algorithms and Systolic Arrays** (S. Y. Kung and J. Hesson)

The papers ranged from bit-level designs to designs based on powerful processors, and addressed issues such as programmability, reconfigurability, synchronization, and methods of specification and transformation.

(26) **VLSI for Speech and Image Processing** (J. Kumar and A. N. Venetsanopoulis)

The Speech subsession included papers on both recognition and synthesis. The Image subsession included papers on advances in 2-D filtering and image processing control.

(36) **VLSI Architecture** (Paul Toldalagi)

This wide ranging session included papers on architectures that implement filters, a decoder, A-to-D conversion, a multidimensional access memory, and integer product modulo, a Fermet number. One paper discussed a technique for the optimal scheduling of certain signal flow graphs onto a synchronous multiprocessor architecture.

(38) **VLSI Design Methodologies and Transform Architectures** (Dick Lyon and Earl Swartzlander)

The subsession on Design Methodologies comprised four papers on optimization approaches appropriate for designing custom VLSI signal processing chips. The techniques ranged from optimization of cells in an array to optimization of filter coefficients for complexity reduction within the constraints of an application's specifications. The subsession on Transform Architectures comprised four papers presenting chip designs and techniques for implementing fast transforms in VLSI. The properties of the various techniques were discussed in terms of performance, generality, and design time.

UNDERWATER ACOUSTICS

(5) **Time Delay Estimation and Geophysical S. P.** (J. P. Ianniello)

Target tracking with Kalman filters using robust adaptive methods was discussed. The next paper analyzed time delay estimation in the presence of jitter. Time delay estimation for spread spectrum signals was considered next. Another paper examined time of arrival resolution in multipath communication systems. Physical modelling was used in a presentation to optimize predictive deconvolution of seismic array data. A random ocean model using a linear systems theory approach was described. Also given was a paper on use of coherence estimates to construct 2-D filters which diminish or remove coherent noise from correlograms.

(15) **Array Processing and Beamforming I** (D. H. Johnson)

The first four papers focused on eigenanalysis techniques. Rather than simple application of old results, these concentrated on extension of the theory and on derivative ideas. There was a rather detailed paper concerning approximation of the probability density function of a detection statistic. Another paper stood out as a novel approach to high resolution beamforming. The last two papers discussed conventional beamforming.

(22) **Geophysical DSP and Medium Models** (Y. T. Chan)

An adaptive least-squares lattice algorithm to the rejection of under-ice acoustic reverberation, was followed by a deconvolution technique to reconstruct an original pulse from noisy tomography multipath data. Bayesian estimation of object image from wavefield measurements was presented. The last two papers gave methods of inverting a beamformer output, and a technique to determine the modulation type of a radio frequency signal.

(33) **Detection and Estimation** (R. G. Pridham)

Two of the papers involved application of AR (autoregressive) techniques to detection of broadband signals. This represents a new approach to the problem and resulted in considerable audience interest. Another group of three papers addressed the nongaussian detection problem. One of these considered detection in an environment limited by Weibull distributed reverberation and is relevant to the problem of imaging the sea bottom. The remaining papers treated various other detection and estimation problems of current interest.

(45) **Time Delay Estimation and Source Localization** (T. S. Durrani)

The first paper gave a unified approach to source motion analysis and multipath modeling. Two papers addressed TDE: one with non-stationary signals, and the other on the effect of correlated array motion on source location. An EM algorithm for multiple source location was introduced. Another paper described a new application of the adaptive line enhancer as a preprocessor. The next two papers discussed threshold reduction in the target motion analysis problem, and a dynamic programming approach to low SNR target tracking. The last paper developed a fast m.l.e. algorithm time delay estimation.

(46) **Array Processing and Beamforming II** (M. Wax)

The first paper considered SVD estimation of multiple two dimensional sinusoids, followed by a paper on robust resolution of closely spaced signals. The next paper gave performance limits of high resolution beamformers at low SNR. The next two papers discussed adaptive beamforming in correlated or coherent noise. Several papers examined sensitivity issues, including how to best distribute elements in a sparse array to estimate the spatial spectrum, and the design of end fire array shadings.

AUDIO AND ELECTROACOUSTICS

(16) **Audio and Electroacoustics** (J. L. Flanagan)

Papers addressed loudspeaker arrays for directivity control, and the analysis of electroacoustics transducers. A particularly interesting paper discussed the design and optimization of a digital FM receiver using DPLL techniques. Other papers included hall acoustics, and a new auditory model for the evaluation of sound quality of audio systems.

(41) **Phonetic Analysis and Data Bases** (J. P. Haton)

Three papers treated vowel behavior in several languages. Three speech recognition algorithms were proposed: one for use in benchmarking and speech data base analysis, another for connected digits in Japanese, and the third one for matching a phonetic lattice with a graph. Three papers were on acoustic correlates, cues for aspirated plosives, and bispectrum-based voiced/unvoiced decision. The remaining five papers dealt with lexical stress determination, speaker sampling for enhanced diversity, collection of phoneme samples, phonetic knowledge representation, and probabilistic grammar for phonetic to French transcription.

(43) **Mediumband Speech Coding II** (A. Gersho)

Papers spanned bit rates from 8 kbits/s using subband coding (four papers), vector quantization (four papers), adaptive delta modulation, baseband coding, multipulse LPC, and a new technique called critical point coding. Papers on subband coding dealt with backward adaptive prediction, dynamic bit allocation, and application of multirate coding to digital speech interpolation. Papers on vector quantization dealt with efficient bit allocation for multiple VQ's, use of normalized residual, embedded coding of speech, and design of contour VQ's.