California Coding: Early LPC Speech in Santa Barbara, Marina del Rey, and Silicon Valley 1966–1982

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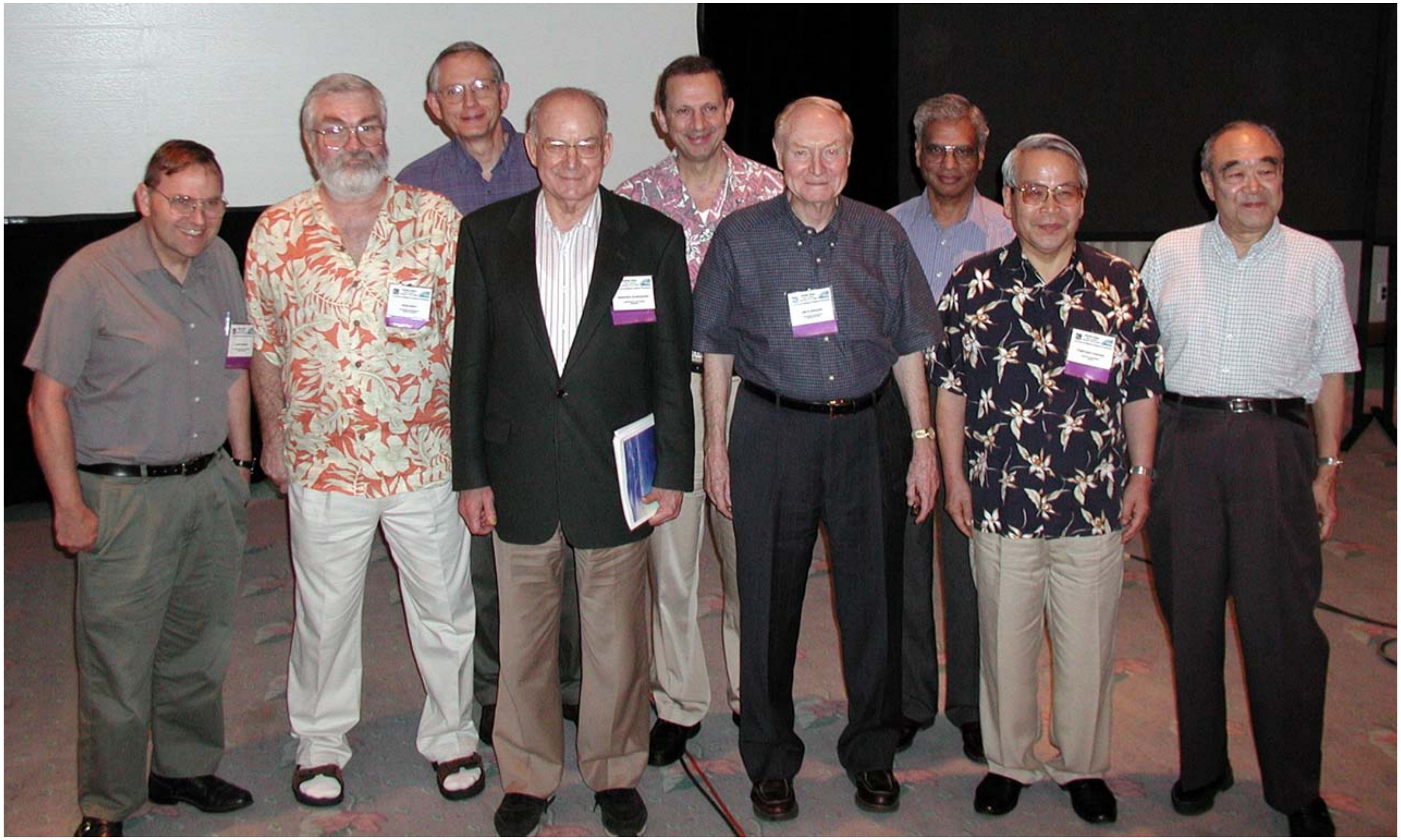
Origins



Special Workshop in Maui (SWIM), 12 January 2004

Part I: Technical Overview

Fix m. Observe a data sequence $X^m = \{X_0, X_1, \ldots, X_{m-1}\}$.

Optimal 1-step prediction ¿ What is the optimal predictor of the form $\tilde{X}_m = p(X_0, \ldots, X_{m-1})$?

Optimal 1-step linear prediction ¿ What is the optimal linear predictor of the form $\tilde{X}_m = -\sum_{l=1}^{m} a_l X_{m-l}$?

Modeling/density estimation ¿ What is the probability density function (pdf) that "best" models X^m?

Spectrum Estimation ¿ What is the "best" estimate of the power spectral density or covariance of the underlying random process?

The Application

Speech Coding ¿ How apply linear prediction to produce low bit rate speech of sufficient quality for speech understanding and speaker recognition?

Wide literature exists on all of these topics in a speech context and they are intimately related.

See, e.g., J. Makhoul's classic survey [35] and J.D. Markel and A.H. Gray's classic book [41].

Clearly problems ill-posed unless define terms like "optimal" and assume some structure.

Optimal Prediction

Random vector $X^m = (X_0, X_1, \ldots, X_{m-1})^t$
Correlation $r_{i,j} = E[X_i X_j]$, $R_n = \{r_{i,j};\ i,j = 0,1,\ldots,n-1\}$

¿ What is best $\tilde{X}_m = p(X^m)$ given X^m in sense of minimizing mean squared prediction error $E[\epsilon_m^2]$, $\epsilon_m = X_m - \tilde{X}_m$?

Answer: $\tilde{X}_m = E[X_m | X^m]$, $\alpha_m = \sigma^2_{X_m|X^m}$.

If X^{m+1} Gaussian, $\Rightarrow$ optimal predictor $= E[X_m|X^m] = (r_{m,0}, r_{m,1}, \ldots, r_{m,m-1}) R_m^{-1} X^m \stackrel{\Delta}{=} (a_{m-1}, \ldots, a_2, a_1)^t X^m$,
MMSE $= \alpha_m = |\ R_{m+1}\ |\ /\ |\ R_m\ |$

$\Rightarrow$ Form and performance are determined entirely by R_{m+1}!
$\Rightarrow$ optimal predictor is *linear!!*
$\Rightarrow$ optimal linear predictor = optimal predictor

Optimal Linear Prediction

NonGaussian case: Performance = performance in Gaussian case with same autocorrelation: $a \stackrel{\Delta}{=} (a_0 = 1, a_1, \ldots, a_m)$
$\tilde{X}_m = -\sum_{l=1}^{m} a_l X_{m-l}$, has MSE $E[\epsilon_m^2] = a^t R_{m+1} a \Rightarrow$
optimal linear predictor = optimal predictor for Gaussian

Moral: Gaussian assumption provides short cut proofs in nonGaussian problems — no calculus and get global optimality!

Efficient inversion to find a: Cholesky decomposition =
.
covariance method

If R_{m+1} Toeplitz, Levinson-Durbin algorithm =
.
autocorrelation method

Common approach: Calculus or orthogonality principle $\Rightarrow$ normal equations (Wiener-Hopf, Yule-Walker):
m linear equations in m unknowns.

¿ What if don't know R_m, observe long sequence of actual data $X_0, X_1, \ldots, X_{n-1}$? Under suitable conditions can estimate:

$$\hat{r}_k = \frac{1}{n-m}\sum_{l=m}^{n-1} X_l X_{l-|k|};\ \hat{R}_{m+1} = \{\hat{r}_{i-j};\ i,j = 0,1,\ldots,m\}$$

$$\overline{r}_{i,j} = \frac{1}{n-m}\sum_{l=m}^{n-1} X_{l-i} X_{l-j};\ \overline{R}_{m+1} = \{\overline{r}_{i,j};\ i,j = 0,1,\ldots,m\}$$

and "plug in." $\hat{R}_m$ Toeplitz, $\overline{R}_m$ not.

As $n \to \infty$, $\overline{R}_{m+1} \approx \hat{R}_{m+1}$ LP $\Leftrightarrow \operatorname{argmin}_{a:a_0=1} a^t R_{m+1} a$

Processes and Filters

For $n = m, m+1, \ldots$ find linear least squares estimate $\tilde{X}_n = -\sum_{l=1}^{m} a_l X_{n-l}$: Previous formulation $\Rightarrow$ optimal a, MMSE α_m.

LTI filter with input X_n, response a_k: *prediction error filter* or *inverse filter* $\Leftrightarrow A(f) = \sum_{n=0}^{m} a_n e^{-i2\pi n f}$

$$X_n \to \boxed{A(f)} \xrightarrow[\text{residual, excitation}]{\epsilon_n = \sum_{k=0}^{m} a_k X_{n-k}} \boxed{1/A(f)} \to X_n = \epsilon_n - \sum_{l=1}^{m} a_l X_{n-l}$$

Limit: $m \to \infty$, the orthogonality principal $\Rightarrow$ prediction error becomes *white*!

For finite m can interpret LP as trying to make prediction error *as white as possible*.

Maximum Likelihood View

Suppose process X_n is order m Gaussian autoregressive source: W_n is iid Gaussian, variance σ^2 and

$$X_n = \begin{cases} -\sum_{k=1}^{m} a_k X_{n-k} + W_n & n = 0, 1, \ldots \\ 0 & n < 0 \end{cases}$$

Observe $X_0, \ldots, X_{n-1}$. ¿ What is maximum likelihood estimate of a, i.e., what a maximizes $f_{X^n|a}(x)$? $\quad A_n X^n = W^n$ where

$$A_n = \begin{bmatrix} 1 & & & & & 0 \\ a_1 & 1 & & & & \\ \vdots & a_1 & 1 & & & \\ a_m & & \ddots & \ddots & & \\ & \ddots & & & & \\ 0 & & a_m & \ldots & a_1 & 1 \end{bmatrix}$$

$$\Rightarrow R_W^{(n)} = A_n R_X^{(n)} A_n^* = \sigma^2 I \;\Rightarrow\; R_X^{(n)} = \sigma^{-2} A_n^{-1} A_n^{-1*}$$

so the pdf for $X = X^n$ is

$$f_{X^n|a}(x) = \left(\frac{1}{2\sigma^2\pi}\right)^{\frac{n}{2}} e^{-\frac{1}{2\sigma^2} x^t A_n^t A_n x}$$

For large n $x^t A_n^t A_n x \approx \mathcal{E}_n = a^t \overline{R}_{m+1} a$

ML estimate does (approximately for large n) same minimization as LP solution a and $\sigma^2 = \alpha_m$.

ML method produces a *model* or *density estimate*: a Gaussian autoregressive process fit to data.

Maximum Entropy View

Suppose have estimate $\hat{R}_m$ of correlations to lag m of stationary random process X_n.

¿ What m-step Markov random process maximizes the Shannon differential entropy rate:

$$h(X) = \lim_{n\to\infty} \frac{1}{n} h(X^n)$$

where

$$h(X^n) = -\int f_{X^n}(x^n) \log f_{X^n}(x^n)\, dx^n\,?$$

Since assume Markov, $h(X) = h(X_m|X^m)$.

Note: No Gaussian assumption, stated as a *variational problem*.

Answer: If n and R_n are fixed, then largest differential entropy is (surprise!) obtained by Gaussian density as

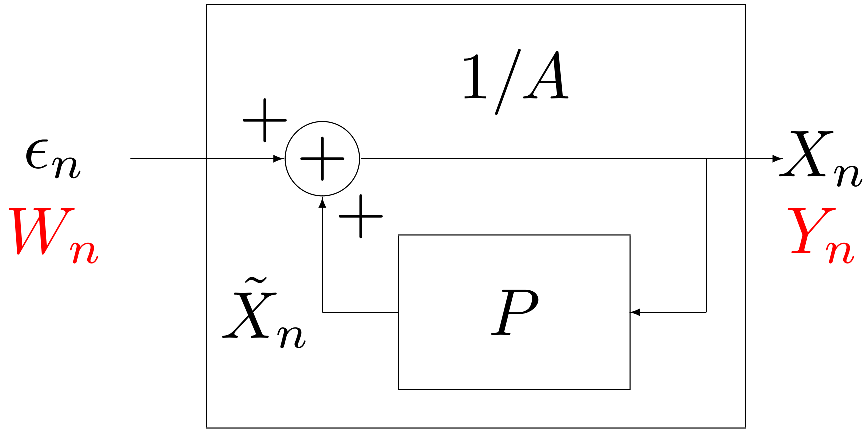
$$h(X^n) = \frac{1}{2}\log(2\pi e)^n \mid R_n \mid .$$

$\Rightarrow$ MAXDET problem, which has a long history and large literature. [73].

As $n \rightarrow \infty$, wish to maximize $h(X_m \mid X^m)$, accomplished by a Gaussian density with variance [61]

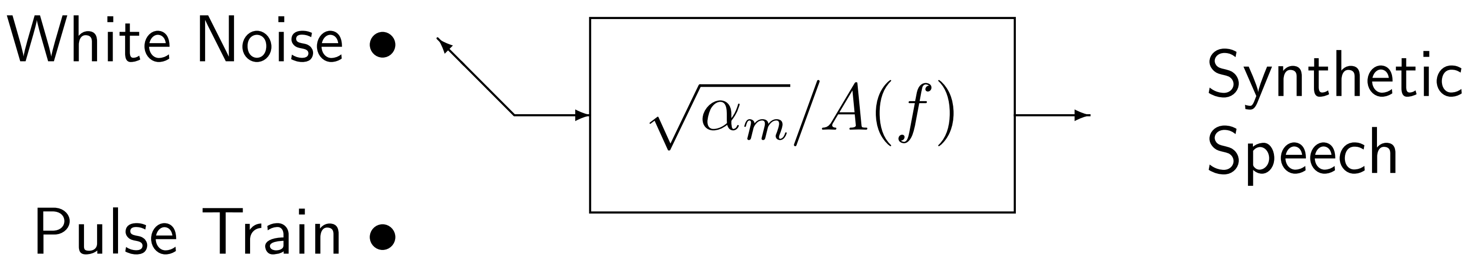
$$\sigma^2_{X_m|X^m} = \frac{\mid R_{m+1} \mid}{\mid R_m \mid} = \alpha_m,$$

achieved by mth order autoregressive process with psd
$S(f) = \alpha_m / \mid A(f) \mid^2$ LP problem again



$$\sigma_W^2 = \alpha_m \ , \ r_X(n) = r_Y(n), n = 0, 1, \ldots, m$$

$\Rightarrow$ LPC *model*



Simplistic: no voicing or pitch estimation details

Minimum Distortion Model Selection

Given process $\{X_n\}$ with autocorrelation R or psd S

Assume a distortion measure $d(R, R_Y)$ or $d(S, S_Y)$

Choose best model in the class $\mathcal{A}_m$ of all mth order autoregressive processes by minimum distortion (nearest neighbor) rule.

Example: Modified *Itakura-Saito distortion measure* by

$$d(S, S_Y) = \int_{-1/2}^{1/2} \left(\frac{S(f)}{S_Y(f)} - \ln \frac{S(f)}{S_Y(f)} - 1 \right) = \frac{a^t R_{m+1} a}{\sigma_Y^2} - \ln \frac{\alpha_m}{\sigma_Y^2} - 1$$

Example of Kullback's *minimum discrimination information* for density/parameter estimation. [3, 55, 62]

Relative entropy rate between Gaussian processes (Pinsker [4])

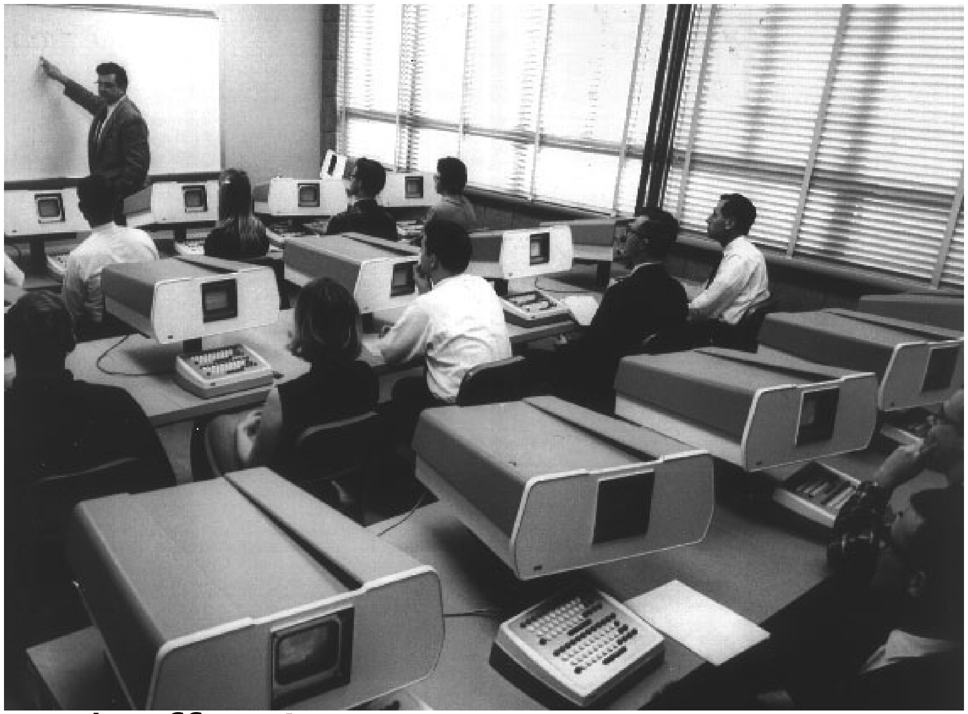
Minimized by choosing the a to minimize $a^t R_{m+1} a$ and $\sigma_Y^2 = \alpha_m$.

LPC

Find model (α_m, A) as before. Coding occurs when the final model is selected from a discrete set, e.g., quantize separate parameters or parameter vector. Local synthesis at decoder.

Part II: History – 1966

UCSB Glen Culler introduces On-Line System (OLS or Culler-Fried) — allows real time signal processing at individual student terminals. Founds Culler-Harrison, Inc (CHI) which will eventually develop the FPS array processor. Reknowned for building fast and effective computer systems, and many of his students and associates will found companies. CHI ideas will be successfully adopted and commercialized by FPS, will replace SPS-41.



1966

In December Saito and Itakura at NTT [5] describe an

approach to automatic phoneme discrimination and, as part of the development, develop the ML approach to speech coding: LP parameters transmitted to decoder with voicing information. Decoder synthesizes from noise or pulse train driving autoregressive filter.

See also 1968 & 1969 papers [11, 12].

From [5]:

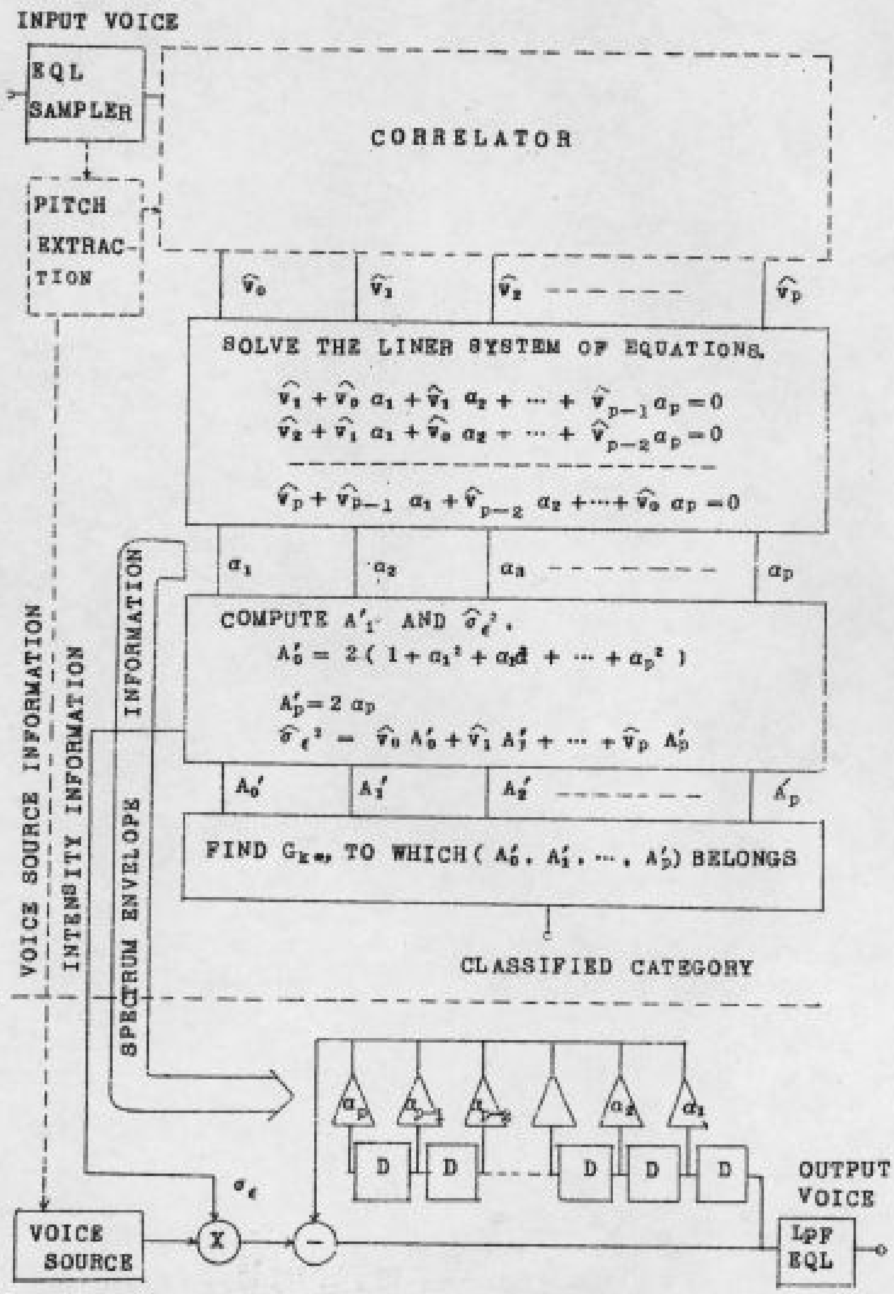


図 5. 新しいパラメータ伝送方式

—39—

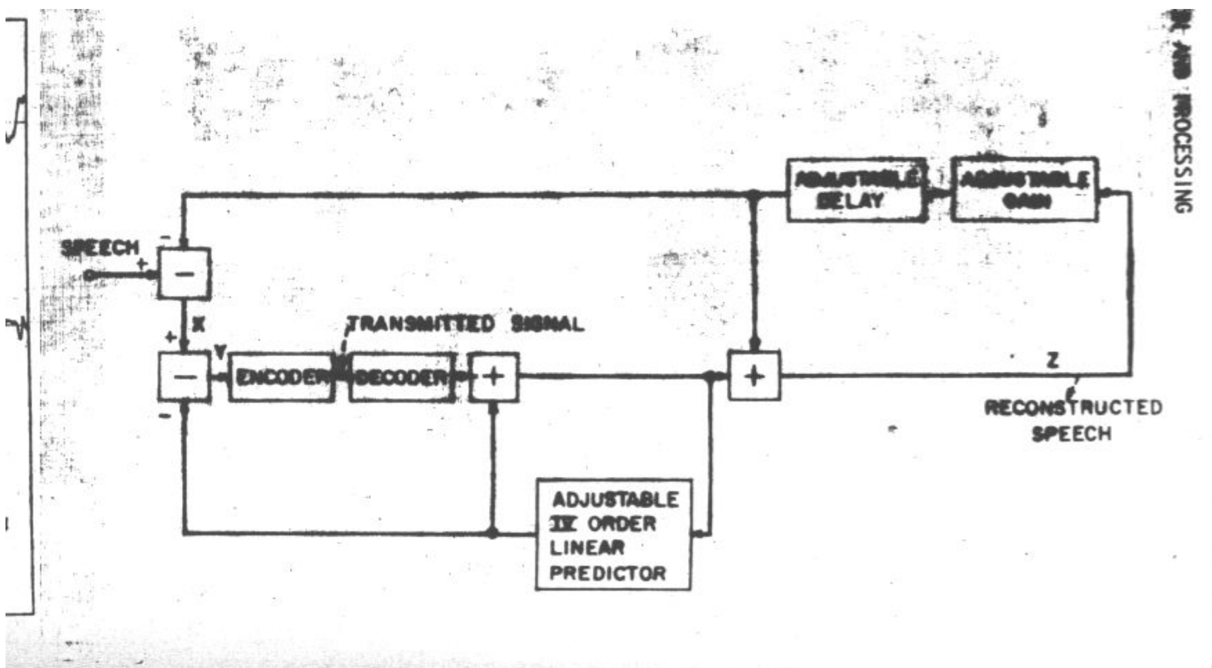
1967

October John Burg presents maximum entropy approach [9] and wins best presentation award at the meeting of the Society of Exploration Geophysists. Dave Sakrison complains about buzz words "maximum entropy." Focus is on prediction error properties. Variational, not parametric.

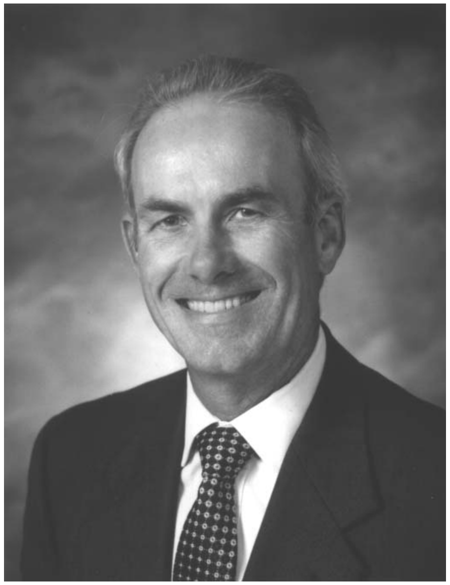
1967

November B.S. Atal and M.R. Schroeder [6]: LP coefficients used to form prediction residual, which is also coded. Adaptive predictive coding (APC), *residual excited* LPC. No explicit modeling. Elaborated in 1968 [7, 8] using covariance method.

From [6]

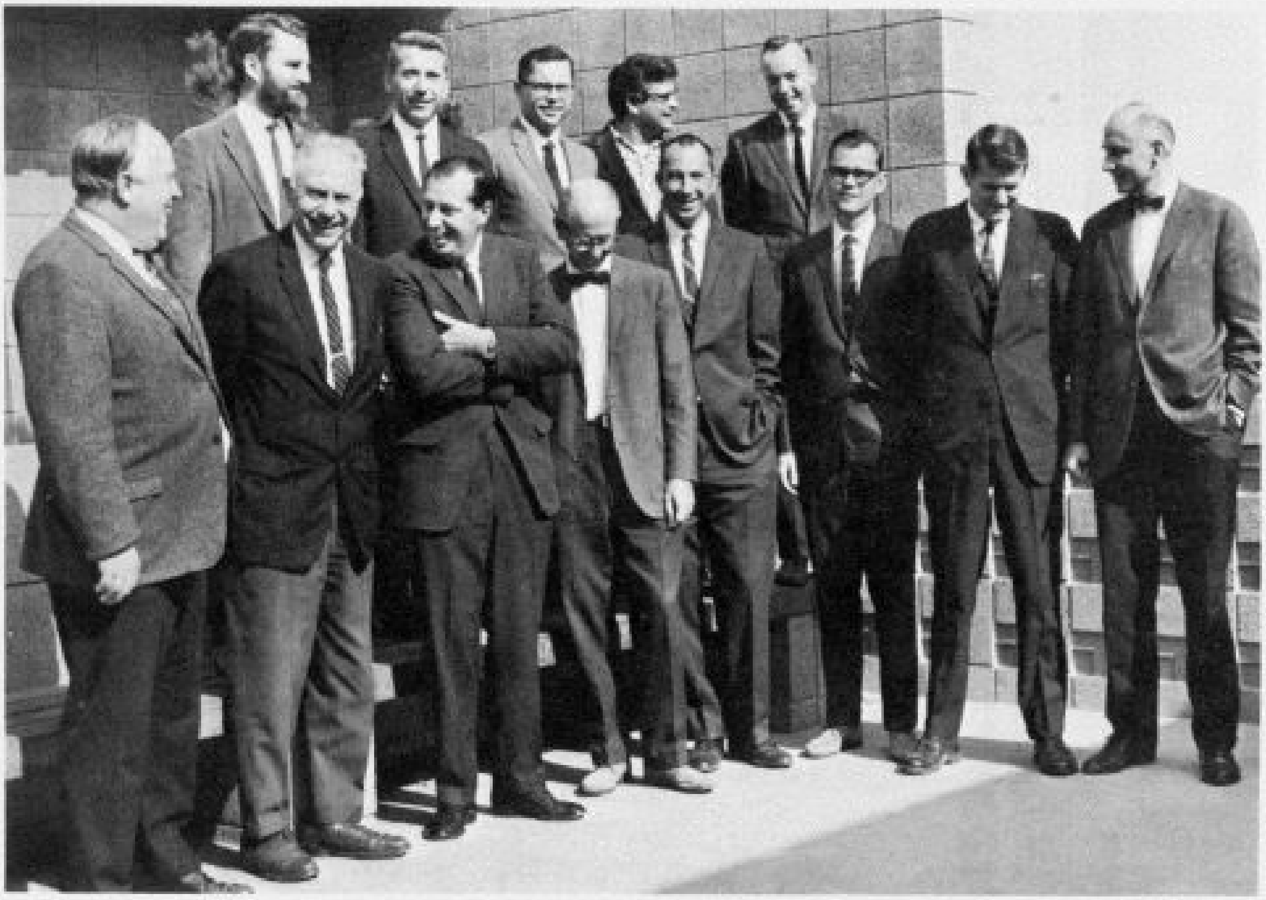


1968



John Markel drops required language course in French for PhD program at Arizona State. Moves to UCSB (Fortran is accepted there). Joins Speech Communications Research Lab (SCRL). Reads Flanagan's book and sets goal to someday write and publish a book in the same series with the same publisher.

1968



Front row: Philip Ordung, Albert Conrad, Jorge Fontana, George Matthaei, Roger Wood, Kenneth Kotzebue, John Skalnik, Glen Wade. **Second row:** Augustine Gray Jr., Glen Heidbreder, John Baldwin, Glen Culler, James Howard.

UCSB Faculty in 1968, including A.H. Gray, Jr., and Glen Culler

1968

John Burg [10] presents "A new analysis technique for time series data" at NATO Advanced Study Institute — the Burg algorithm. Finds reflection coefficients from original data using a forward-backward algorithm. Later dubbed "covariance lattice" approach in speech[46, 56].

Glen Culler contributes to Interface Message Processor (IMP) specification (with Shapiro, Kleinrock, Roberts). BBN gets contract to build and deploy 4 in January 1969. [72]

1969

Itakura and Saito[12] introduce partial correlation (PARCOR) [1969] variation on autocorrelation method, finds partial correlation [1] coeffients. Similar to Burg algorithm, but based on classical statistical ideas and lower complexity.

May Glen Culler proposes online speech processing system aimed at real-time speech encoding based on a signal decomposition that would now be called a Gabor wavelet analysis. [13]

November B.S. Atal presents LPC speech coder at Annual Meeting of the Acoustical Society of America. [14]. Abstract published in 1970, full paper with Hanauer in 1971[16], uses covariance method.

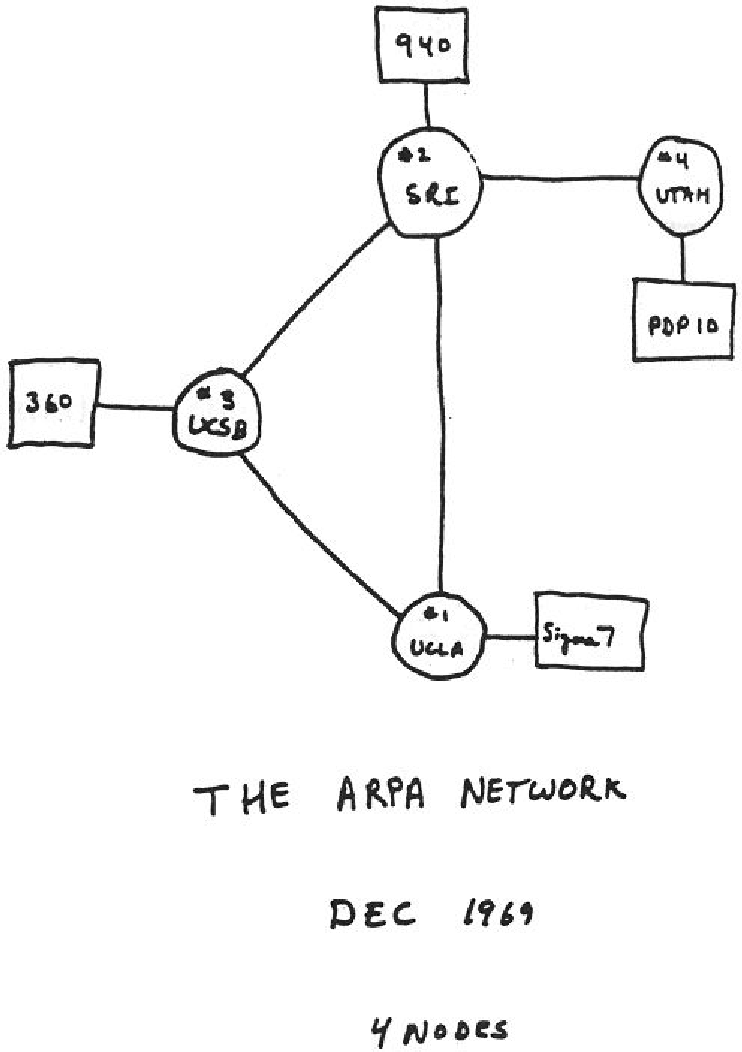


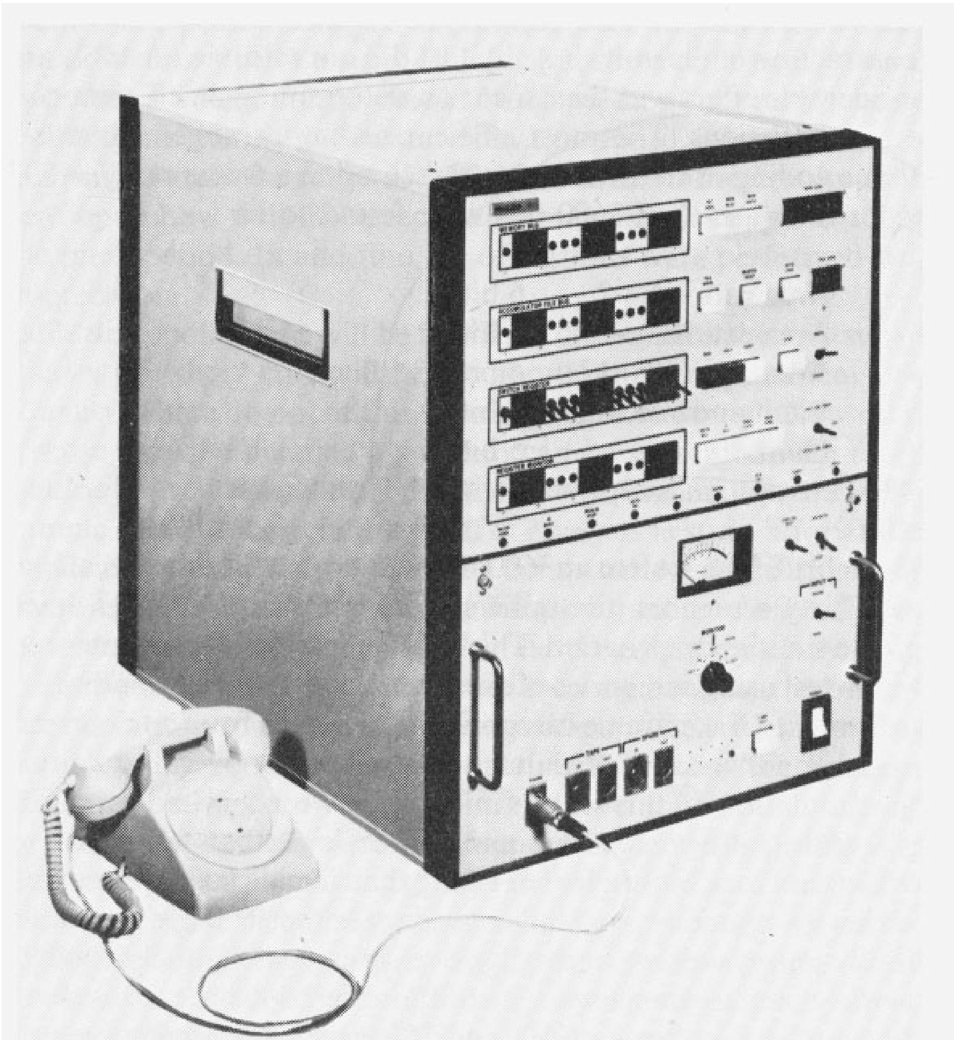
FIGURE 6.2 Drawing of 4 Node Network (Courtesy of Alex McKenzie)

Thanks to Culler, UCSB becomes the third node (IMP) on the ARPAnet (joining #1 UCLA, #2 SRI, #4 University of Utah)

Note no two computers were the same (Sigma-7, SDS-940, IBM-360, and DEC-PDP10)

Drawing by Jon Postel of ISI

1971



Real time LPC using Cholesky/covariance at Philco-Ford in PA. LONGBRAKE II Final Report in 1974[33]. 16 bit fixed point LPC. Four were sold (Navy and NSA), they weighed 250 lbs ©. Used PFSP signal processing computer. [41]

1972

Bob Kahn (ARPA) with Jim Forgie (LL) and Dave Walden (LL) initiate first efforts towards packet speech on net. Simulated pieces of 64 Kbps PCM speech packets on ARPANET to understand how might eventually fit packet speech into net. Concluded major change in packet handling and serious compression would be needed.

Danny Cohen working at Harvard on realtime visual flight simulation. Bob Kahn (ARPA) suggests to Danny that similar ideas would work for real time speech communication over developing ARPAnet. Suggests Danny consider moving to Information Sciences Institute (ISI) in Marina del Rey.

1973

Danny moves to ISI, works with Steve Casner, Randy Cole, and others and with SCRL on real time operating systems. Danny forms Network Secure Communications (NSC) group as ARPA Project directed by Kahn. (Later called Network Speech Compression and Network Skiing Club because of a preference for winter meetings in Alta.) Danny learns of LPC.

DSP chips did not yet exist. Every node on ARPAnet had different equipment and software. Focus on interface.

LL tradition is that first realtime 2400 bps LPC on the FDP done by Ed Hofstetter using Markel/Gray LPC formulation.

ARPA Network Information Center
Stanford Research Institute
Menlo Park, California 95025

RECEIVED
DEC 6 1973

Marcia Keeney
SRI-ARC
November 14, 1973

Network Speech Compression Note #3
NIC 19946

NETWORK SPEECH COMPRESSION GROUP MEMBERSHIP LIST

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Original Members of NSC: ISI, University of Utah, BBN, MIT-LL, SRI. Soon joined by SCRL, CHI. Others attend as well, including TI, NRL, Harris, NSA, Bell Labs

1973 Continued

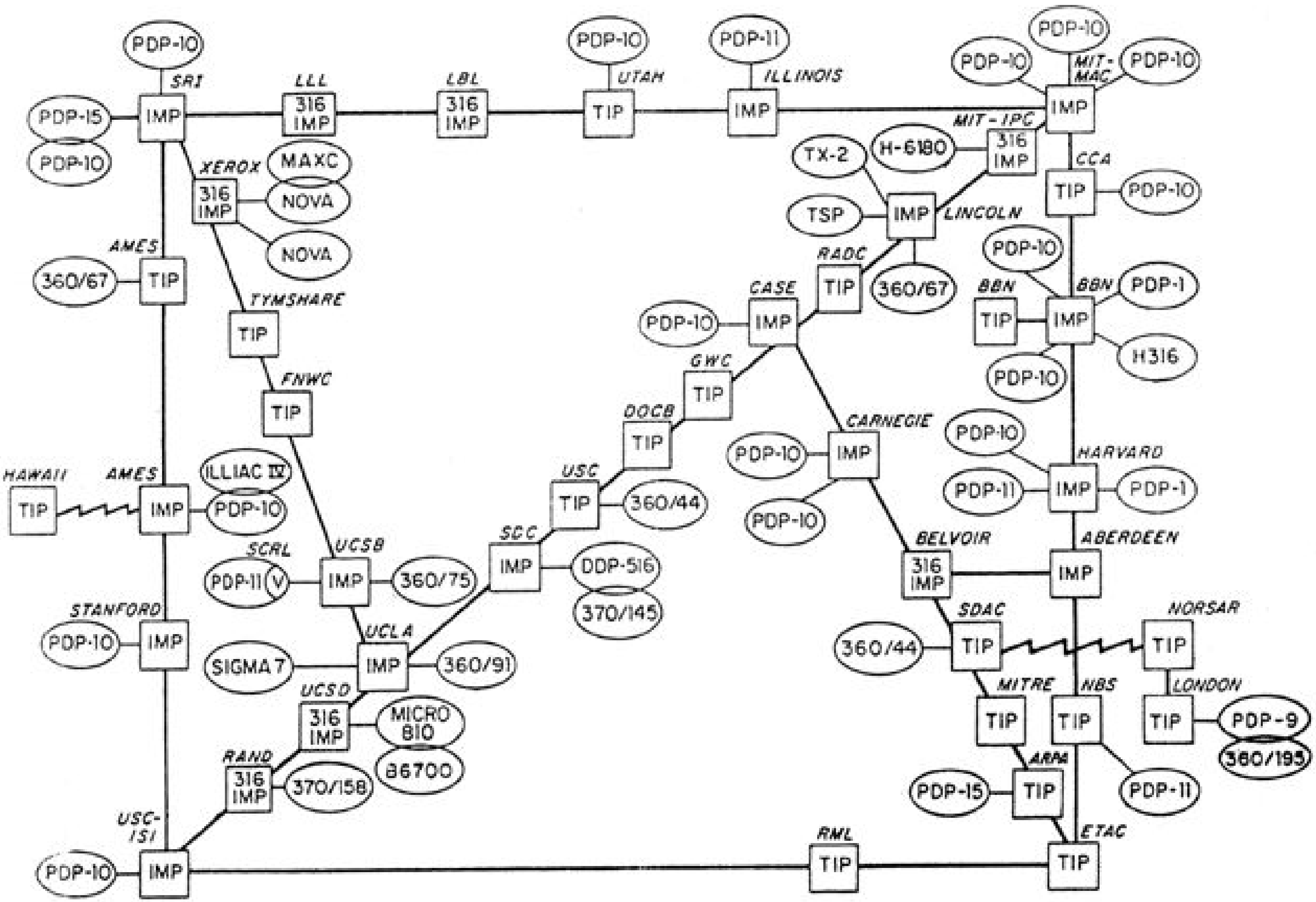
John Markel becomes Vice President of SCRL. Markel, Gray, and Wakita [21, 22, 23] publish SCRL reports and papers describing their implementations of Itakura's algorithms plus several of their own. Provide Fortran code for LPC and associated signal processing algorithms.

ISI adopts basic Markel/Gray software [22] as vocoder technique for network speech project. SPS-41 chosen for implementation (LL and CHI excepted) . Divided software development among ISI, BBN, LL, SRI

John Burg meets Bishnu Atal, learns of LPC.

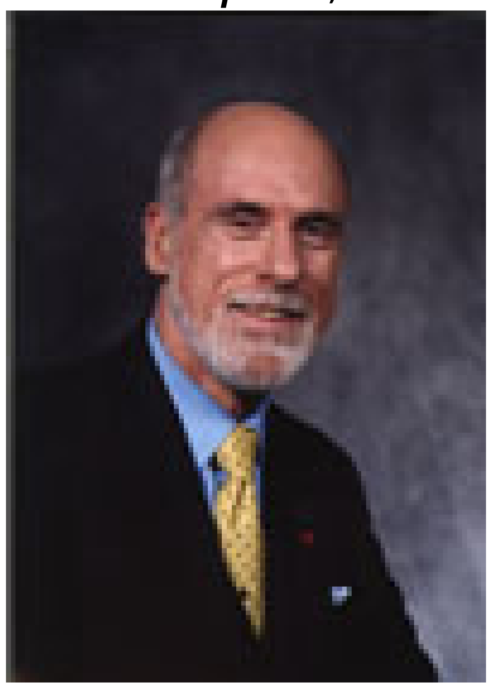
Danny Cohn raises issue of developing protocols supporting real-time applications with Kahn, who refers him to Vint Cerf.

ARPA NETWORK, LOGICAL MAP, SEPTEMBER 1973



1974

January Danny Cohen and Vint Cerf meet in Palo Alto, begin long disscussion regarding handling realtime applications vs. reliable data. Danny describes difference as the difference between milk and wine: *you have to deliver the milk quickly before it spoils even if it spills, but wine can take a LOT longer.*



TCP invented by Bob Kahn and Vint Cerf [26, 74]. Published in May.

1974 Continued

Network Voice Protocol (NVP) developed and written by Danny Cohen in coordination with others in NSC [30, 39, 43, 66]. Independent of TCP, uses only ARPANET message header.

Cohen realized that TCP was unsuitable for real time communication because of packet and reliability constraints and argued for separation of IP from the original TCP.

Implementation of TCP is a step backward for realtime applications on the ARPANET!

August NVP successfully tested using CVSD 16 Kbps, between ISI and LL. Poor quality at achievable rates.

December First realtime two-way LPC packet speech communication. 3.5 kbs over ARPAnet between CHI and MIT-LL. [29, 34, 37, 66, 67] Uses basic M&G LPC algorithms [37, 23, 25] coupled with NVP. CHI: MP-32A signal processor + AP-90 array/arithmetic coprocessor, LL: TX2 and FDP.

Development completed on Secure Terminal Unit (STU) I. produced 1977–1979. APC using Levinson, $35K @. Speech coding at NSA led by Tom Tremain, active participant in NSC

1975

John Burg's company Time and Space Processing (TSP) begins work on real time speech box using the Burg algorithm. Charlie Davis manages hardware and algorithm implementations.

John Markel notes many algorithms published for "optimal quantization" of LPC parameters. Observes any system is optimal according to some criterion. Other Markelisms:
Never believe demonstrations based on the training data.
Never trust quality if the original is played first.

October Chaffee and Omura (UCLA) algorithm for designing a codebook of LPC models based on rate distortion theory ideas for speech coding at under 1000 bps. [31, 36]

1976

Linear Prediction of Speech by J.D. Markel and A.H. Gray, Jr. published, fulfilling Markel's goal.

Comment on M&G from Joseph P. Campbell of LL (former NSA employee): "it was considered basic reading, and code segments (translated from Fortran) from this book (e.g., STEP-UP and STEP-DOWN) are still running in coders that are in operational use today (e.g., FED-STD-1016 CELP)."

1976

Texas Instruments begins development of Speak & Spell toy: Larry Brantingham, Paul Breedlove, Richard Wiggins, and Gene Frantz.

Prior to TI, Wiggins worked on speech algorithms at MITRE in cooperation with LL and visited Itakura and Atal at Bell, NSC, Makhoul and Viswanathan at BBN, George Kang at NRL. While at TI visits Markel at SCRL and ISI in summer of 1977.

January • First LPC conference over ARPANET based on LPC and NVP successfully tested.: CHI, ISI, SRI, LL 3.5 kbps

March 1976

NVP published by Danny Cohen. Excerpt: “The Network Voice Protocol (NVP), implemented first in December 1973, and has been in use since then for local and transnet real-time voice communication over the ARPANET at the following sites:

- Information Sciences Institute, for LPC and CVSD, with a PDP-11/45 and an SPS-41.
- Lincoln Laboratory, for LPC and CVSD, with a TX2 and the Lincoln FDP, and with a PDP-11/45 and the LDVT.
- Culler-Harrison, Inc., for LPC, with the Culler-Harrison MP32A and AP-90.
- Stanford Research Institute, for LPC, with a PDP-11/40 and an SPS-41.”

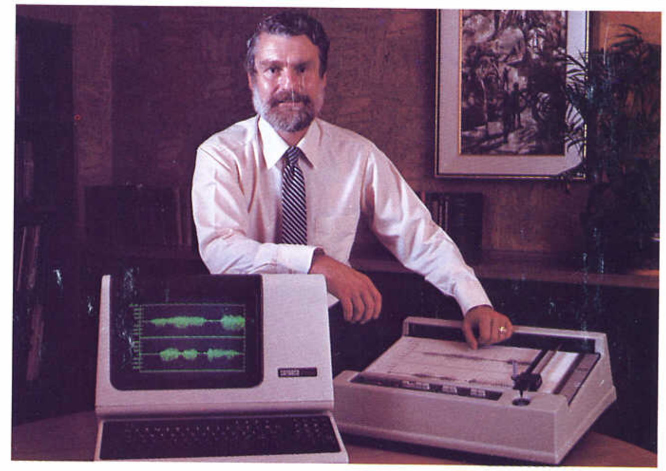
“The NVP’s success in bridging among these different systems is due mainly to the cooperation of many people in the ARPA-NSC community, including Jim Forgie (Lincoln Laboratory), Mike McCammon (Culler-Harrison), Steve Casner (ISI) and Paul Raveling (ISI), who participated heavily in the definition of the control protocol; and John Markel (Speech Communications Research Laboratory), John Makhoul (Bolt Beranek & Newman, Inc.) and Randy Cole (ISI), who participated in the definition of the data protocol. Many other people have contributed to the NVP-based effort, in both software and hardware support.”

1977

John Markel founds Signal Technology Inc. (STI). First product is Interactive Laboratory System (real time signal processing packages based on SCRL software).

WHY YOU SHOULDN'T WRITE YOUR OWN SIGNAL PROCESSING SOFTWARE.

BY A.H. "STEEN" GRAY, Jr., Ph.D.
Vice President, Signal Technology, Inc.



I've been in this business long enough to know that some things never change. The "make or buy" quandary as it applies to software is a good example. "We've got some expensive programmers; let them earn their keep." How many times have you heard that when you've suggested buying a program package that seems perfectly tailored to your application?

HOW TO ANSWER

If your line of work is signal processing, the answer should be reasonably simple. Just say, "It would take us ten years and a bundle of money to come up with a package as good as the one already available from some experts out in California." That

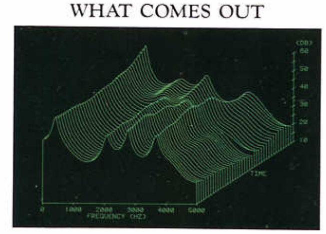
might be just a slight exaggeration, but your point would be well taken. You see, we do have the last word in interactive signal processing software. It's called ILS, and with over 200 installations worldwide, it is often referred to (and not just by us) as the "world standard."

WHAT ILS IS

ILS is a highly modular set of FORTRAN programs that make up a sophisticated interactive software system with standard file structures, documentation and ongoing support. To date, it is performing with excellent results in a wide variety of industries and technologies, including: **speech—noise and vibration—acoustics—biology—medicine—simulation—digital filtering—sonar—radar—seismic—** and some we aren't being told about.

Many of our users also find DACS, our Data Acquisition and Conversion Software, and APAS, our Array Processor Application Software, of great value in their applications.

WHAT COMES OUT



With any compatible computer system and the appropriate terminals, ILS will give you: **pattern analysis—digital filtering—signal editing—3-D displays—modeling—correlation—convolution—spectral density—signal displays—coherence—** and maybe even a picture of your Aunt Sally, if that's what you want.

THE POINT IS

The important thing is, ILS is available *now*. It has been proven in a multitude of applications around the world. And it can cost you a lot less in time and money to buy it from us rather than developing a comparable package yourself.

FREE DEMO

If you have a compatible graphics terminal and modem, we can arrange to give you an on-line demonstration of ILS. Simply call (toll free) and ask for our ILS marketing representative at (800) 235-5787. We also have a video tape that illustrates many of the features of ILS and its capabilities. We'll be happy to send it to you.

IF YOU'RE STILL WITH US

If you've read this far, you probably have some kind of interest in signal processing. I'd be delighted to send you a reprint of the series of three articles on digital filtering that I co-authored with John Markel. Write to me at the address below.

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1977 Continued

Development begins for STU II (LPC/APC) (1977-1980), produced 1982–1986, $13K @

April Bell Labs applies for patent for "'packet transmission of speech" four years after ARPA/NSC LL/CHI demonstration. Granted USA Patent 4,100,377 in 1978.[66, 75]

May: TSP demonstrates Model 100 LPC box at Armed Forces Communication and Electronics Association annual show. $12K/box. Continued as viable product through mid 80s.

August At ISI, Cohen, Cerf, and Jon Postel discuss the need to handle real time traffic – including speech, video, and military applications. Agree to extract IP from TCP. Create user datagram protocol (UDP) for nonsequenced realtime data.

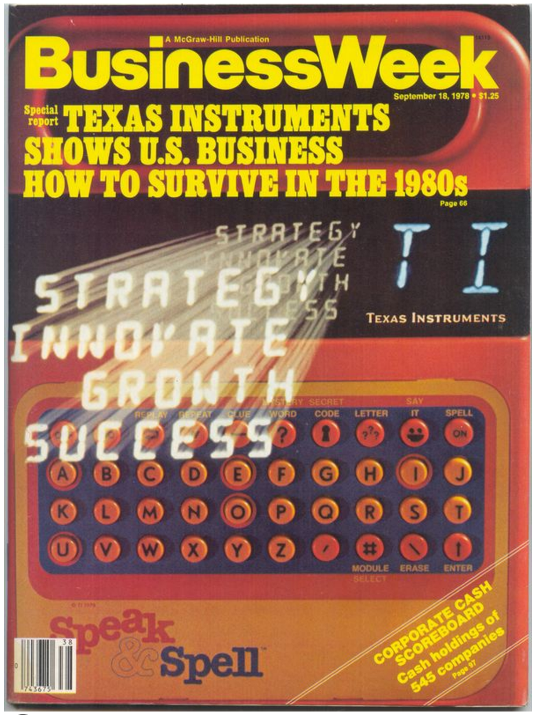
1978

January: IP officially extracted from TCP in version 3[47].
Eventually ATM developed for similar reasons.
Finally TCP/IP suite stabilizes with version 4, still in use today.

Irony in current popular view of VoIP as novel, *IP was in fact specifically designed to handle packet speech and other realtime data!!*

April–May LPC conferencing over ARPANET using variable frame-rate (2–5 kbps) among CHI, ISI, and LL (Vishwanath et al. of BBN developed variable-rate LPC algorithm)

June Texas Instrument *Speak & Spell* toy hits the market. 1st consumer product incorporating LPC and 1st single chip speech synthesizer and early DSP chip.



Speech synthesis from stored LPC words and phrases using TMC 0280 one-chip LPC speech synthesizer. Seminal to the development of DSP chips. Before announcement, Wiggins calls Markel, Makhoul, Atal, and Gold to acknowledge their contributions to speech and to announce the Speak & Spell. Markel asked where his royalties were — Wiggins sent him a Speak & Spell.

1978

June Steen Gray gives talk on "Speech compression and speech distortion measures" and plays tapes of 800bps vector quantized LPC speech at 1978 Information Theory Workshop in Lake Tahoe. They sound awful. John Markel phones to explain the original speech was awful.

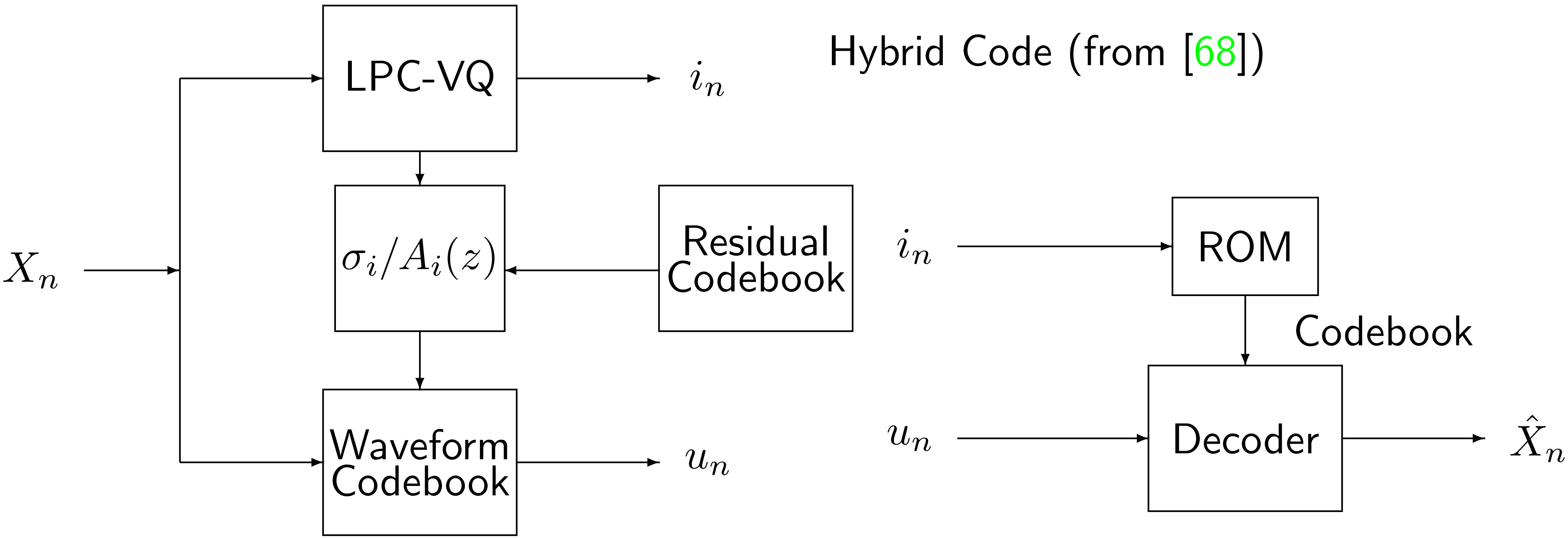
November–December Gray, Buzo, Matsuyama, Gray, and Markel present clustered LPC VQ using Itakura-Saito distortion at conferences [50, 51]. In 1979–1980 expand in [52, 53, 57].

1979

ARPANET/Atlantic SATNET conferencing using LPC10 at 2.4kbps: ISI, LL, NDRE (Norway), UCL (London)

1981

Larry Stewart [60, 65, 68] designs trellis codes (low complexity VQ) for closed loop prediction residuals for coded LPC model. "Hybrid" code using LPC and excitation codebooks. Simulations using legendary Altos systems at Xerox PARC.



1982

800 bps and 350 bps LPC VQ vocoders with DRTs of 87 and 79 at STI [63, 64]

LL's single-card device using 3 NEC7720 chips, 18 sq.in. and 5.5 w shown to NSA Director. In combination with NSA's in-house development of a DSP-chip-based 2.4 kbps modem$\Rightarrow$ that secure desktop telephones were feasible, and led directly to the decision to go ahead with the STU-III development–eventually displaced TSP and other LPC boxes.

Epilog

• Residual codebook excitation combined with perceptual weighting filters, long term prediction, and randomly generated codebook VQ to produce CELP (1985) [69] + postfiltering from Chen and Gersho (UCSB, 1987) [70] ⇒ Fed-Std-1016 CELP coder (Tremain, Campbell, Welch [71]), which included some old Markel/Gray software.

STU III: development begun in 1984, production begun 1987, $2K each. STU III still in use, now being replaced by MELP at 2400 bps and G729 CS ACELP at 8,000 bps.

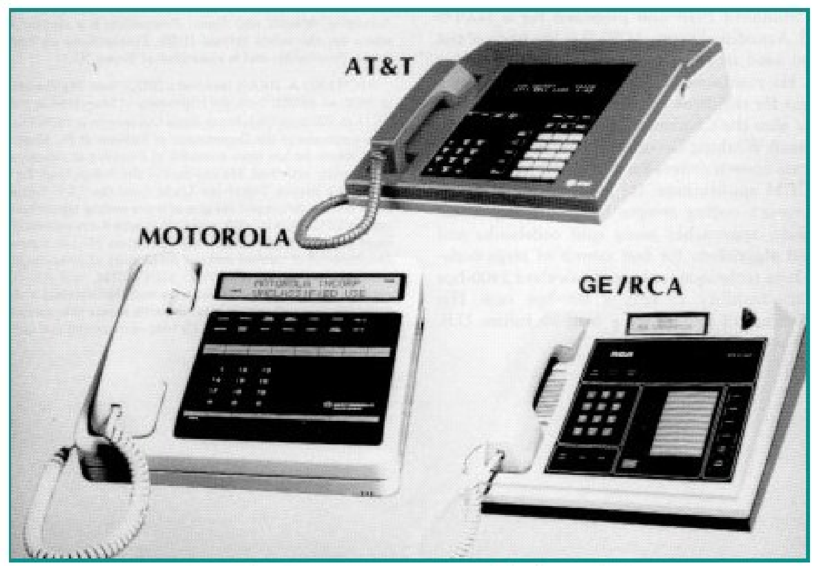


Fig. 4. The STU-III secure voice terminal family, circa 1986

Voice coding at NSA died in 2003.

•Randy Cole: "it's hard to understate the influence that the NSC work had on networking. . . . the NSC effort was the first real exploration into packet-switched media, and we all know the effect that's having on our lives 30 years later."

•[74] ". . . some of the early work on advanced network applications, in particular packet voice in the 1970s, made clear that in some cases packet losses should not be corrected by TCP, but should be left to the application to deal with. This led to a reorganization of the original TCP into two protocols, the simple IP which provided only for addressing and forwarding of individual packets, and the separate TCP, which was concerned with service features such as flow control and recovery from lost packets."

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