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Packet speech on the Arpanet: A history of early LPC speech and its accidental impact on the Internet Protocol

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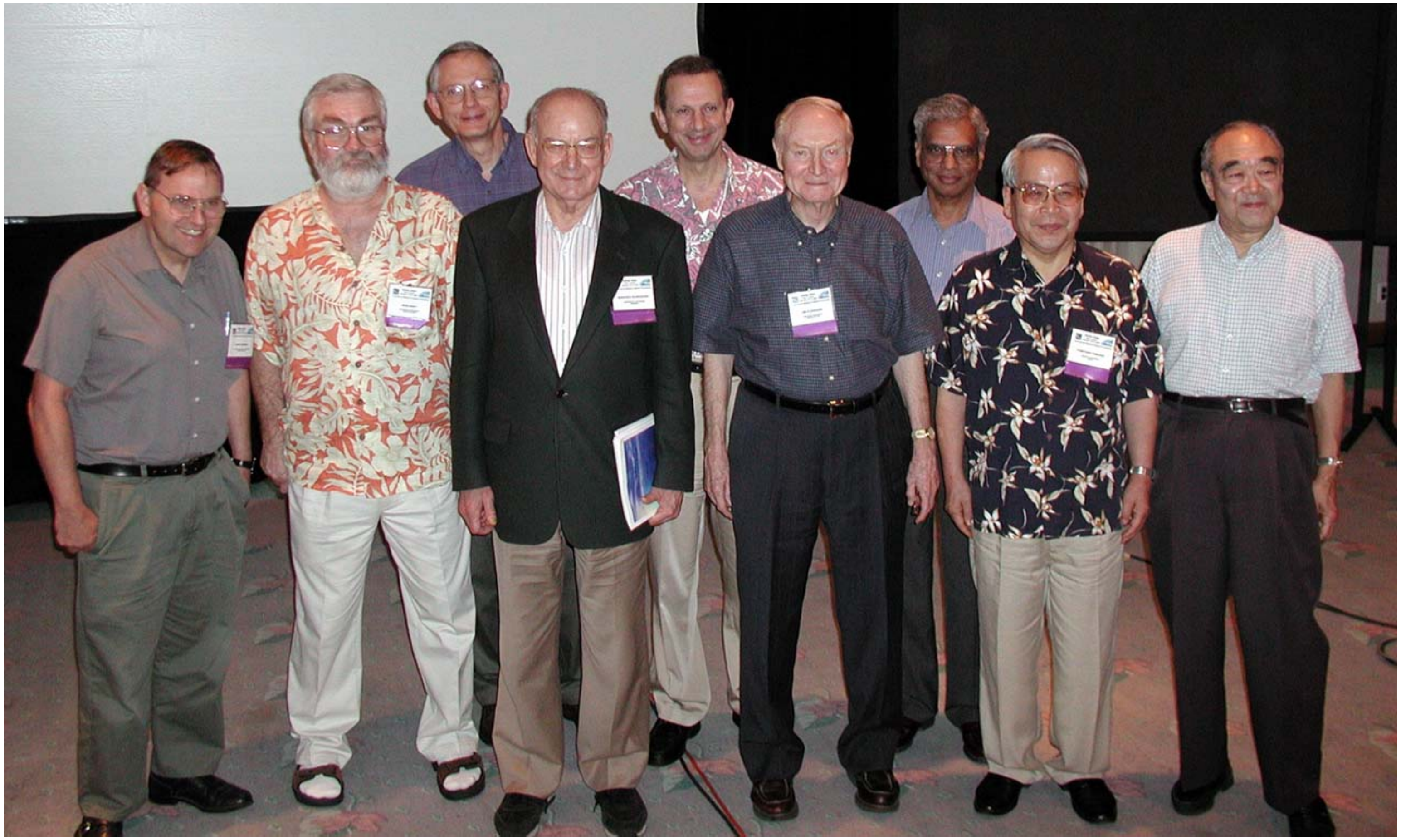
http://ee.stanford.edu/ gray/lpcip.html http://ee.stanford.edu/ gray/dl.html

Origins of this talk



Special Workshop in Maui (SWIM), 12 January 2004

Part I: Linear Prediction & Speech

Observe data sequence $\{X_0, X_1, \ldots, X_{m-1}\}$. Guess X_m

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 underyling 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?
E.g., reproducing waveform (waveform coding) or use model to synthesize (voice coding).

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 [34] and J.D. Markel and A.H. Gray Jr's classic book [39].

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 $r_{i,j} = E[X_i X_j]$, $R_n = \{r_{i,j};\ i,j = 0,1,\ldots,n-1\}$

¿ Best $\tilde{X}_m = p(X^m)$ yielding minimum $E[(\underbrace{X_m - \tilde{X}_m}_{\epsilon_m})^2]$?

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

If X^{m+1} Gaussian, $\Rightarrow E[X_m|X^m] = (a_{m-1}, \ldots, a_2, a_1)^t X^m$

where $(a_{m-1}, \ldots, a_2, a_1)^t = (r_{m,0}, r_{m,1}, \ldots, r_{m,m-1}) R_m^{-1}$

$$\alpha_m = \mid R_{m+1} \mid / \mid R_m \mid$$

$\Rightarrow$ Form and performance are determined entirely by R_{m+1}!

$\Rightarrow$ optimal predictor is *linear!!*

$\Rightarrow$ optimal linear predictor $=$ optimal predictor

Optimal Linear Prediction

$$\tilde{X}_m = -\sum_{l=1}^{m} a_l X_{m-l} \Rightarrow \text{MSE} = \boxed{E[\epsilon_m^2] = a^t R_{m+1} a},$$

where $a \triangleq (a_0 = 1, a_1, \ldots, a_m)^t$, whether or not Gaussian!.

$\Rightarrow$ optimal a for linear prediction (LP) is $\boxed{\operatorname{argmin}_{a:a_0=1} a^t R_{m+1} a}$

$=$ same as optimal for Gaussian! a and α_m as before.

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

Efficient inversion to find a: Cholesky decomposition $\Longrightarrow$

.

covariance method

If R_{m+1} Toeplitz, Levinson-Durbin algorithm $\Longrightarrow$

.

autocorrelation method

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

¿ What if don't know R_{m+1}, but observe long sequence of actual data $X_0, X_1, \ldots, X_{n-1}$? 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 $n \to \infty$, $\overline{R}_{m+1} \approx \hat{R}_{m+1}$

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* $\Leftrightarrow A(f) = \sum_{n=0}^{m} a_n e^{-i2\pi n f}$

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

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

Choose A to make prediction error *as white as possible*.

LP again

And there are more formulations with same solution:

- *Maximum likelihood* Assume $\{X_n\}$ Gauss autoregressive process. Given observations, what is maximum likelihood estimate of parameters describing Gaussian distribution?

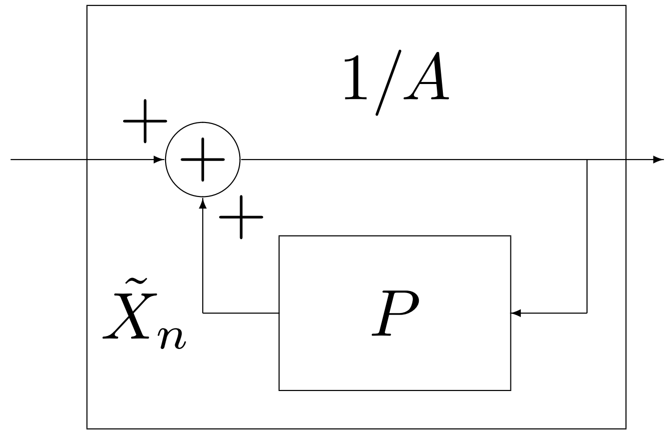
- *Maximum entropy* Suppose estimate $\hat{R}_{m+1}$ of correlations up to lag m of process $\{X_n\}$. What mth order Markov random process maximizes the Shannon differential entropy rate? (Variational problem, no Gaussian *assumption*.)

- *Minimum distortion* Minimum distortion fit of spectra: Itakura-Saito/Kullback-Leibler/minimum discrimination information.

- *Correlation matching* Given set of m autocorrelation values, what is the best estimate of the remaining coefficients?

Linear Predictive Coding (LPC)

true residual ϵ_n
white noise W_n

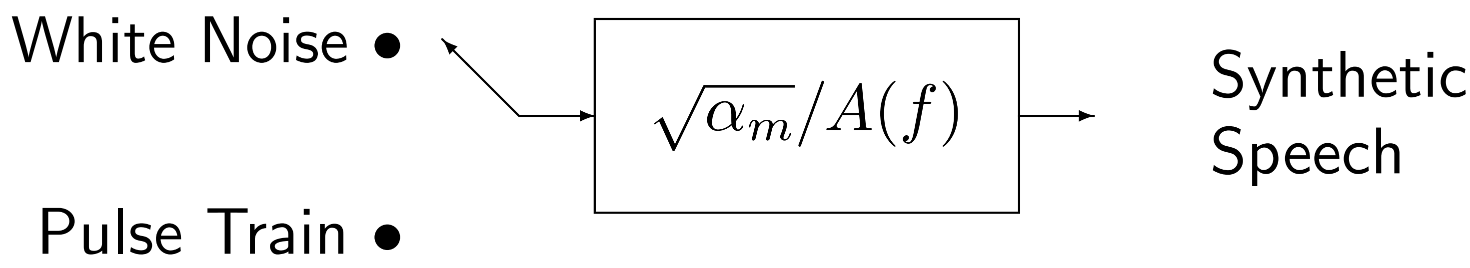


X_n true speech
Y_n synthesized speech

$\Rightarrow$ LPC *model*: mth-order autoregressive model with A solving LP problem.

Simplistic: no voicing or pitch estimation details.

Switch excitation between white noise (unvoiced sounds) and pulse train (voiced sounds)



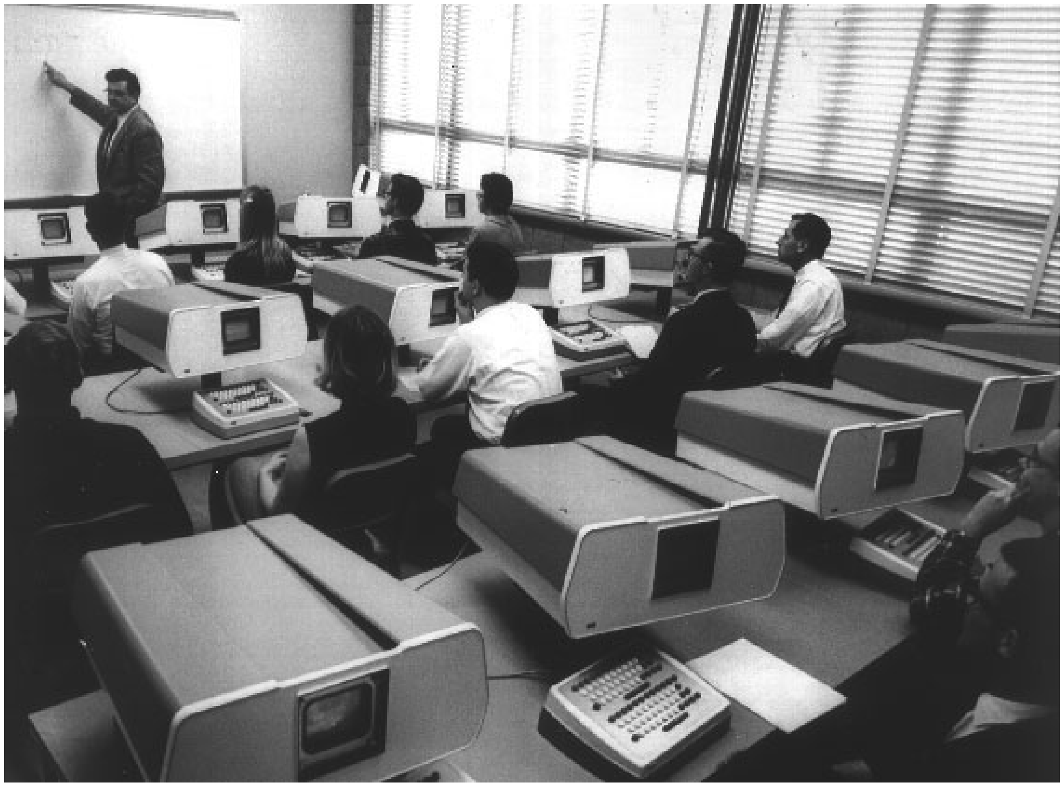
LPC

Estimate autocorrelation or covariance of observed data and find LP model (α_m, A). Coding occurs when the final model is selected from a discrete set, e.g., quantize separate parameters or parameter vector. Local synthesis at decoder.

Classic *vocoder* instead of a *waveform coder*.

Part II: History – 1966

UCSB Glen Culler introduces On-Line System (OLS or Culler-Fried system) — allows real time signal processing at individual student terminals, e.g., DFTs of real sampled speech. Culler is reknowned for building fast and effective computer systems.



1966

In December Saito and Itakura at NTT [4] describe an approach to automatic phoneme discrimination and develop the ML approach and minimum distortion approach to speech coding: LP parameters extracted using autocorrelation method & transmitted to decoder with voicing information. Decoder synthesizes from noise or pulse train driving autoregressive filter.

See also 1968 & 1969 papers [10, 11].

From [4]:

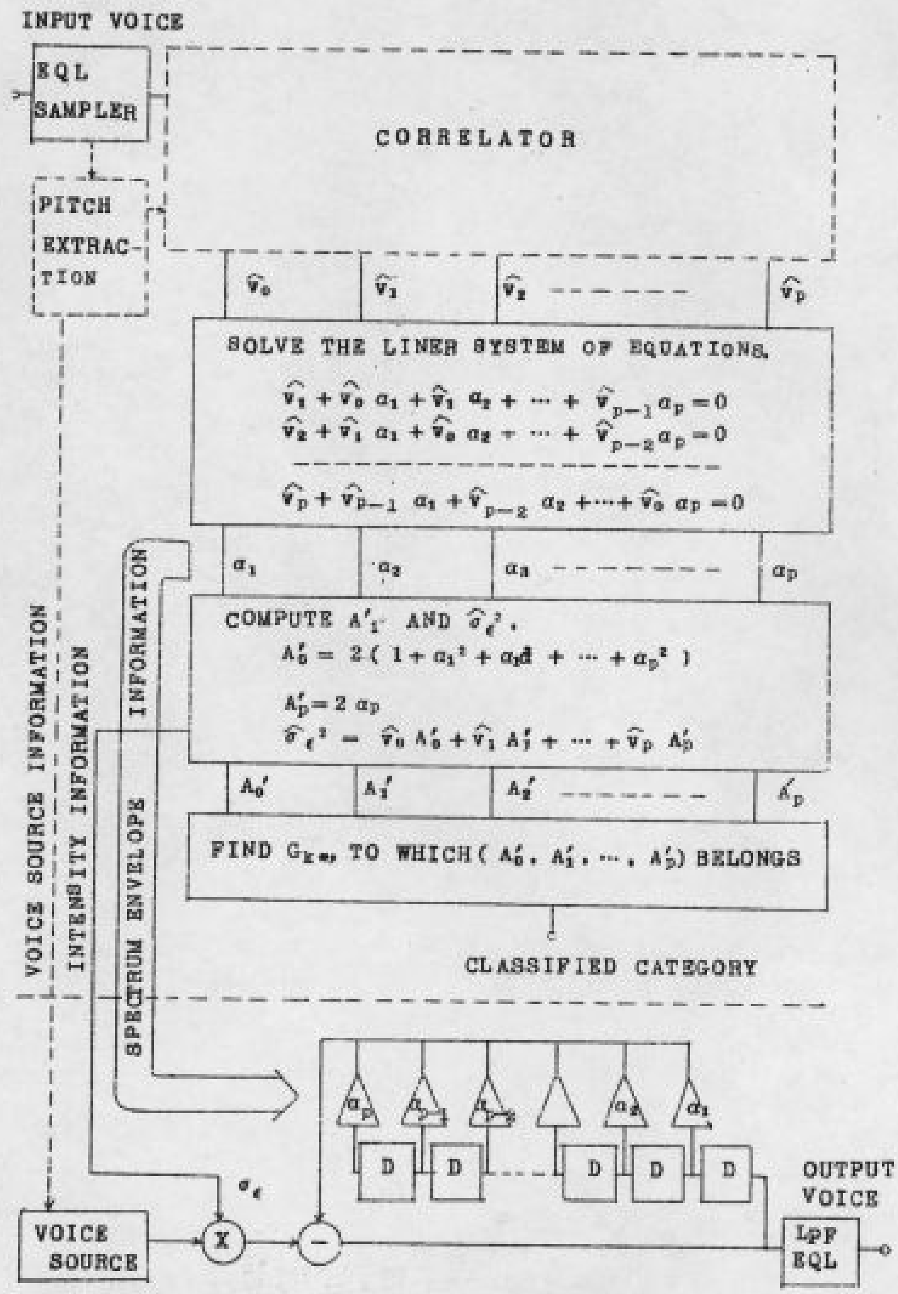


図 5. 新しいパラメータ伝送方式

—39—

1967

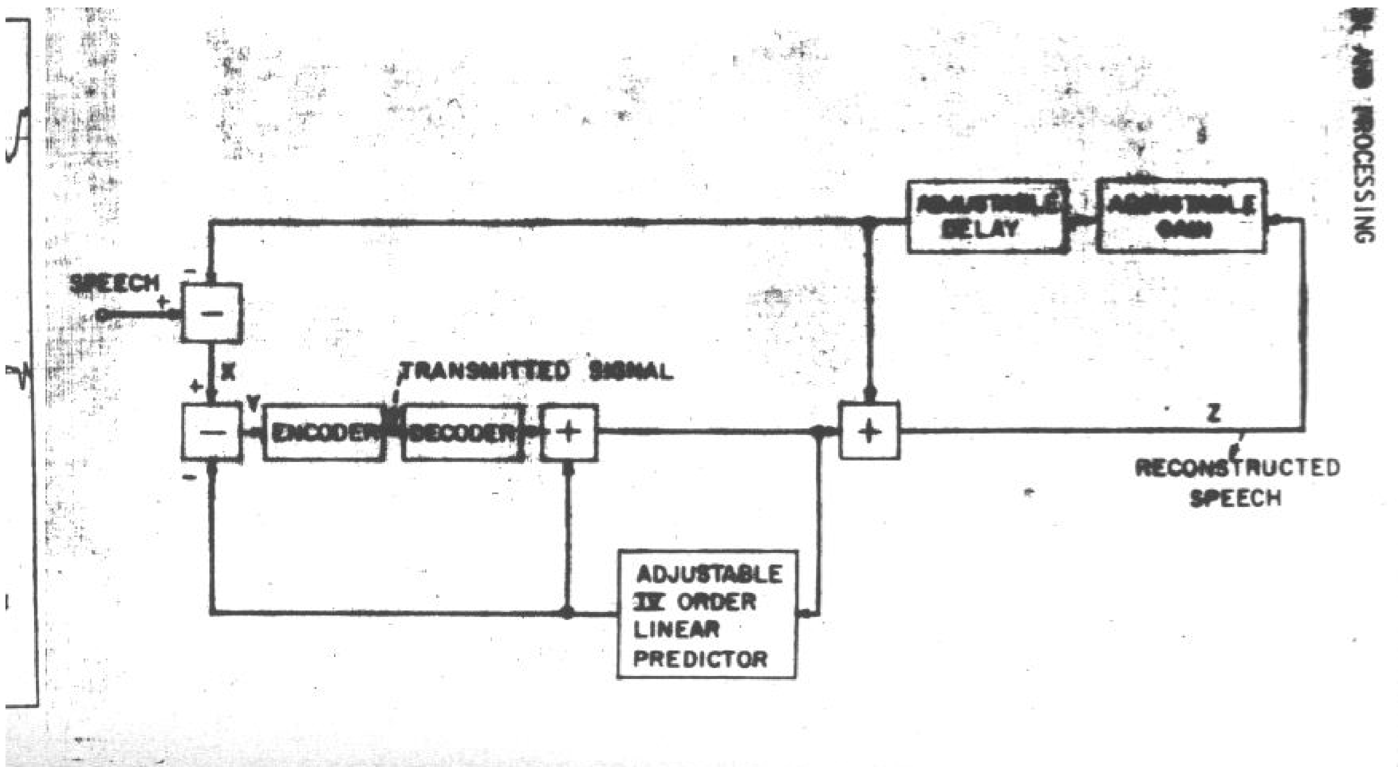
Ed Jaynes and John Burg.

October John Burg presents maximum entropy approach [8] and wins best presentation award at the meeting of the Society of Exploration Geophysists. Focus is on prediction error properties. Variational, not parametric.

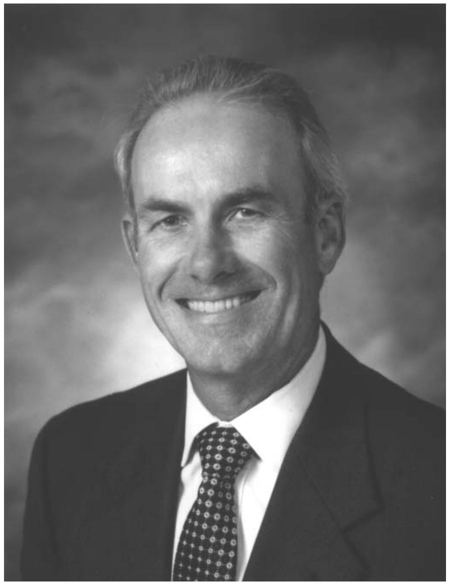
1967

November B.S. Atal and M.R. Schroeder [5]: LP coefficients used to form prediction residual, which is also coded. Adaptive predictive coding (APC), *residual excited* LPC. No explicit modeling. Elaborated in 1968 [6, 7] using covariance method.

From [5]



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.

Begins working with A.H. Gray Jr and Hisashi Wakita on implementations of Itakura's approach.

1968

John Burg [9] 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[44, 53].

Glen Culler contributes to Interface Message Processor (IMP) specification (with Shapiro, Kleinrock, Roberts) — the “node” of the ARPANET. BBN gets contract from ARPA to build and deploy 4 in January 1969. [63]

Culler cofounds Culler-Harrison Inc (CHI), which builds early array processors which are adopted and commercialized by FPS, will replace SPS-41 array processors.

1969

Itakura and Saito[11] 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. [12]

November B.S. Atal presents LPC speech coder at Annual Meeting of the Acoustical Society of America. [13]. Abstract published in 1970, full paper with Hanauer in 1971[15], uses covariance method.

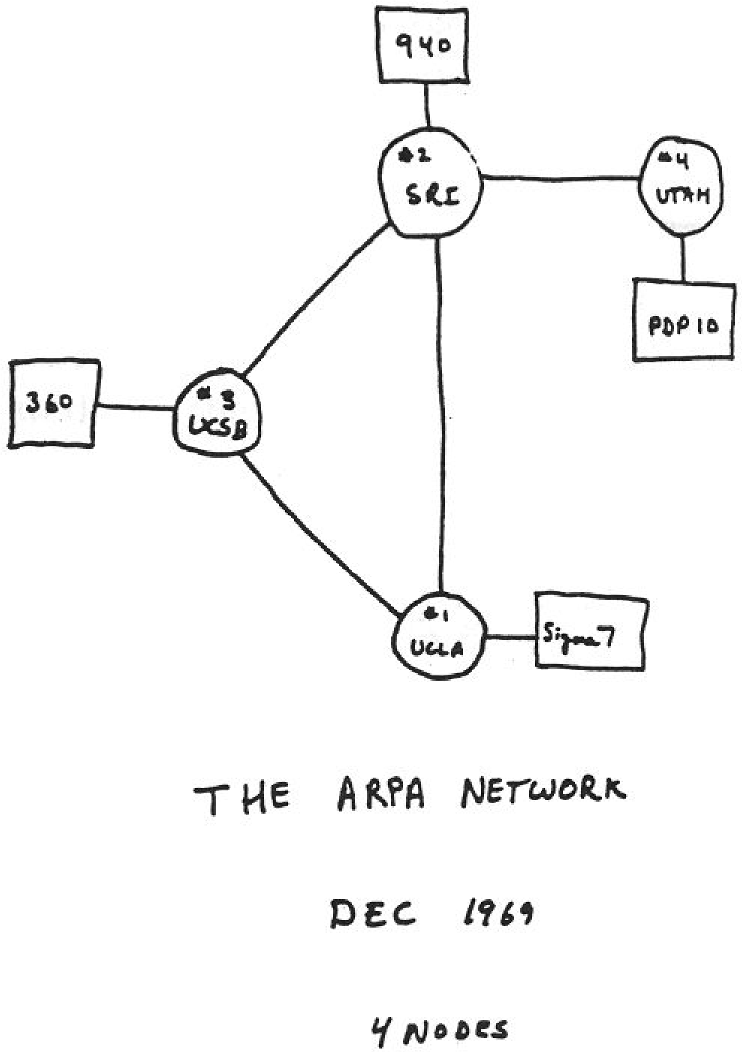


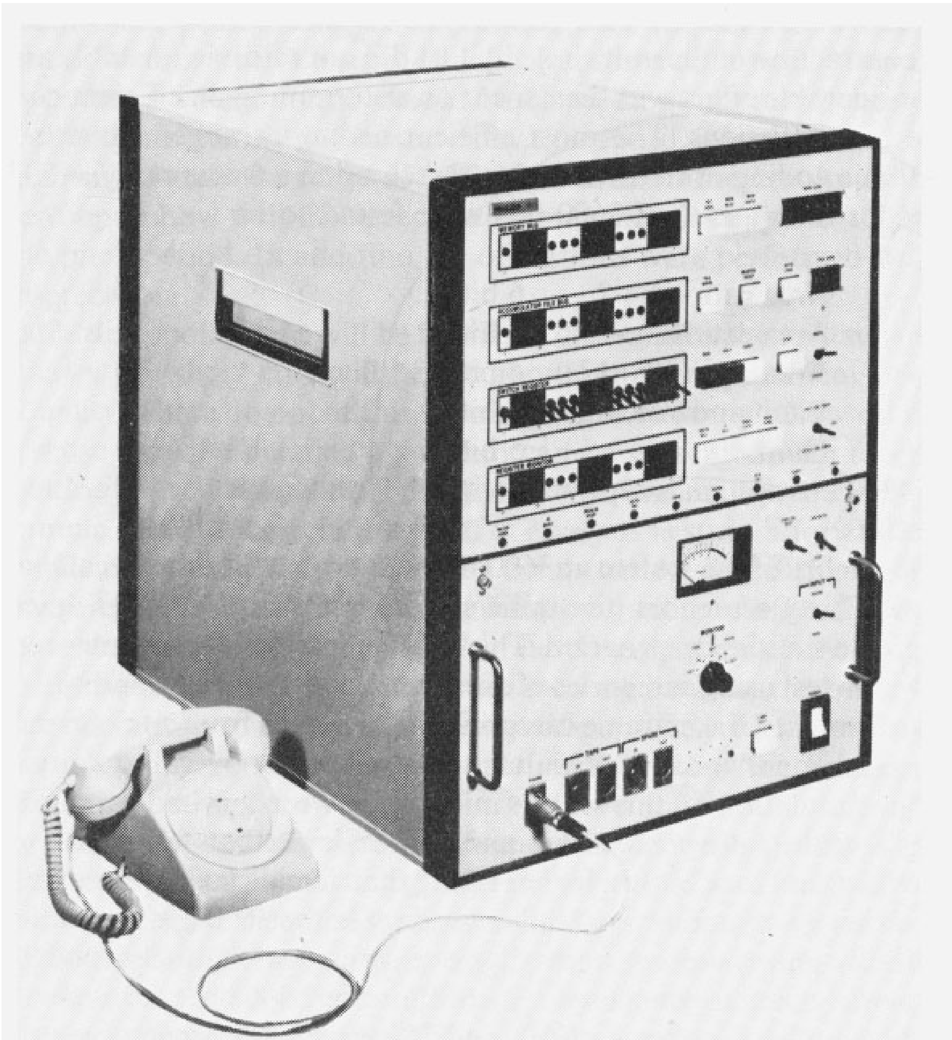
FIGURE 6.2 Drawing of 4 Node Network (Courtesy of Alex McKenzie)

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

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[32]. 16 bit fixed point LPC. Four were sold (Navy and NSA), they weighed 250 lbs @. Used PFSP signal processing computer. [39]

1972

Bob Kahn (ARPA) with Jim Forgie (LL) and Dave Walden (BBN) 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 suggests to Danny that similar ideas would work for real time speech communication over developing ARPAnet and described his project at the USC 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. Kahn forms Network Secure Communications (NSC) group. (Later called Network Speech Compression and Network Skiing Club because of a preference for winter meetings in Alta.) Every node on ARPAnet had different equipment and software. Focus on interface.

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

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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 [20, 21, 22] 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.

Danny works with SCRL on real-time operating systems, learns of LPC.

1973 Continued

ISI adopts basic Markel/Gray software [21] as vocoder technique for network speech project. SPS-41 chosen for implementation (LL and CHI excepted) . Divided software development among ISI, BBN, LL, SRI

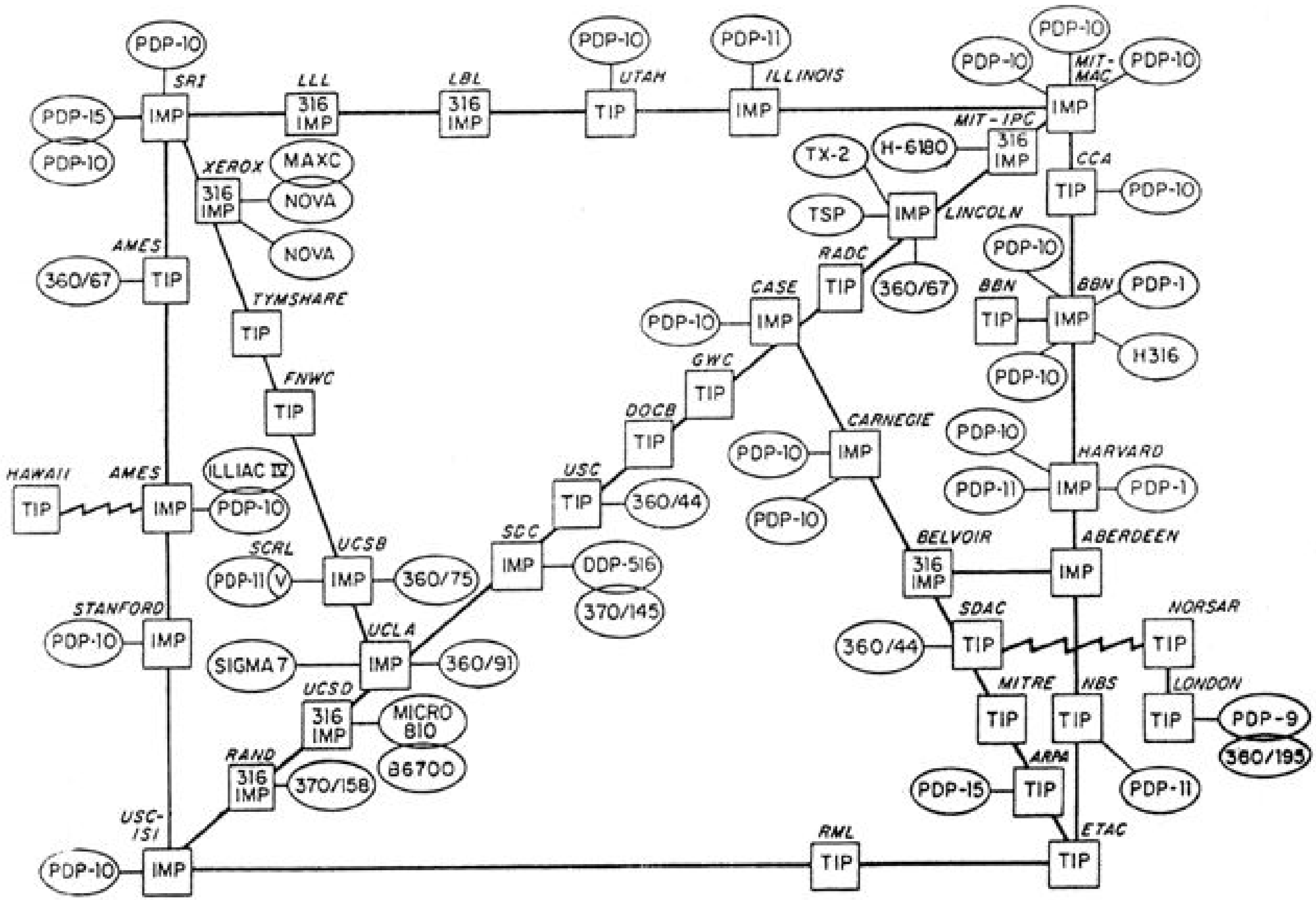
Remember, *DSP chips did not yet exist!*

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

John Burg meets Bishnu Atal, learns of LPC.

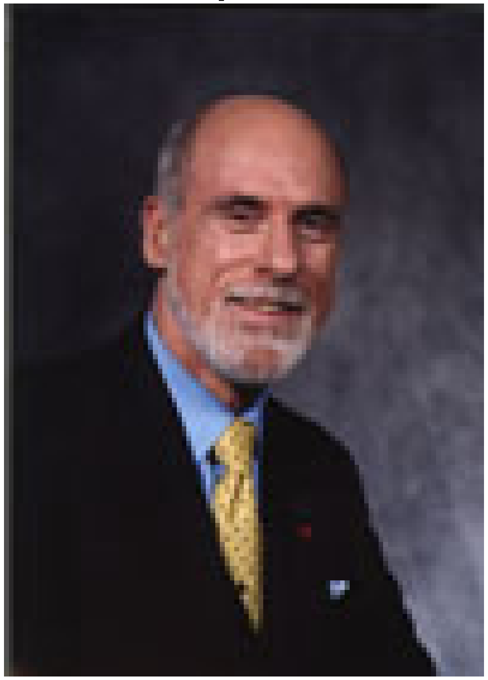
Danny Cohn raises issue of developing protocols supporting real-time applications with Kahn, who refers him to Vint Cerf.

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 [25, 64]. Published in May.

1974 Continued

Network Voice Protocol (NVP) developed and written by Danny Cohen et al. [29, 37, 41, 60]. Independent of TCP, uses only ARPANET message header. Cohen realized that TCP was unsuitable for real time communication because of packet and reliability constraints, argued for separation of IP from TCP.

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

$\star$ **December** First realtime two-way LPC packet speech communication. 3.5 kbs over ARPAnet between CHI and MIT-LL. [28, 33, 36, 60, 61] Uses basic M&G LPC algorithms [36, 22, 24] coupled with NVP. CHI: MP-32A signal processor + AP-90 array/arithmetic coprocessor, LL: TX2 and FDP.

1976

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

March NVP Published: "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."

note who is not mentioned . . .

1976 Continued

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.

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

1977

April James Flanagan at 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.[60, 65]

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.

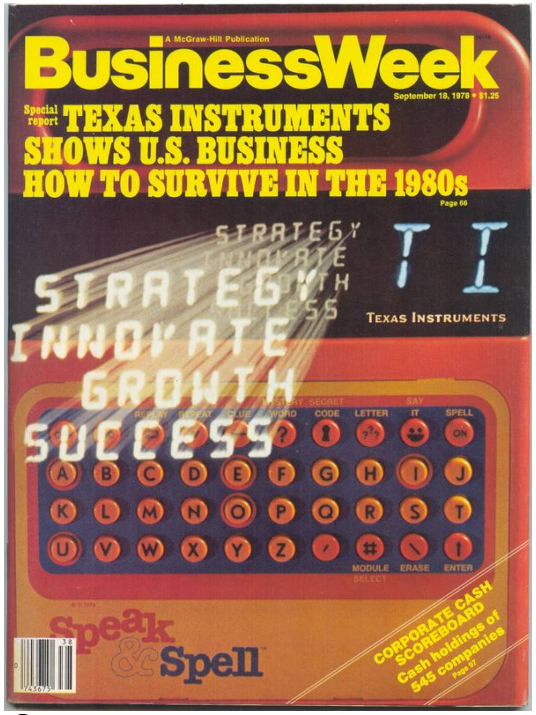
1978

January: IP officially extracted from TCP in version 3 [45]. 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 Instruments 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.

Epilog

- Randy Cole: "it's hard to overstate 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."

- Barry Leiner et al. [64] ". . . 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."

- In 2000 Glen Culler received the National Medal of Technology from President Clinton for "pioneering innovations in multiple branches of computing, including early efforts in digital speech processing, invention of the first on-line system for interactive graphical mathematics computing, and pioneering work on the ARPAnet." Culler died in May 2003.

- Cerf and Kahn won the 2004 ACM Turing Award.
- Danny Cohn is a Sun Fellow and a member of the NAE.

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