

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

Robert M. Gray

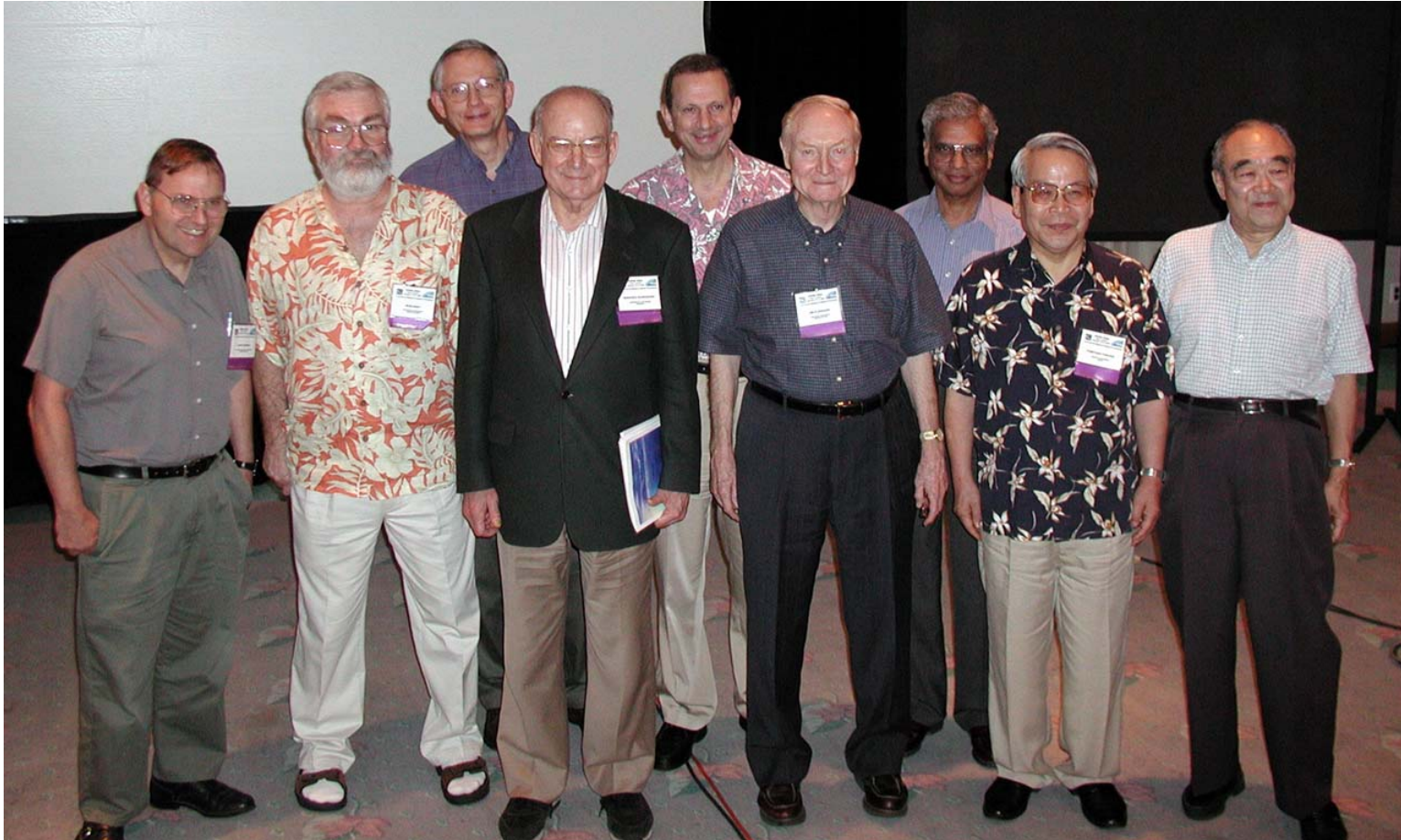
Information Systems Laboratory, Department of Electrical Engineering
Stanford, CA 94305

The author's work in speech was partially supported by the National Science Foundation. Thanks to J. D. Markel, A.H. "Steen" Gray, Jr., John Burg, Charlie Davis, Mike McCammon, Danny Cohen, Steve Casner, Richard Wiggins, Vishu Viswanathan, Jim Murphy, Cliff Weinstein, Joseph P. Campbell, Randy Cole, Rich Dean, Vint Cerf, and Bob Kahn.

<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

Fix m . Observe a data sequence $\{X_0, X_1, \dots, X_{m-1}\}$.

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

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

Modeling/density estimation $\dot{!}$ What is the probability density function (pdf) that “best” models X^m ?

Spectrum Estimation $\dot{!}$ 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 $X^m = (X_0, X_1, \dots, X_{m-1})^t$

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

¿ What is best $\tilde{X}_m = p(X^m)$ yielding minimum $E[(X_m - \tilde{X}_m)^2]$?

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

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

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

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

\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

Linear predictor $\tilde{X}_m = -\sum_{l=1}^m a_l X_{m-l} \Rightarrow \text{MSE} =$
 $E[\epsilon_m^2] = a^t R_{m+1} a$, where $\epsilon_m = X_m - \tilde{X}_m$ and
 $a \triangleq (a_0 = 1, a_1, \dots, a_m)^t$, **whether or not Gaussian!**.

\Rightarrow optimal a for linear prediction (LP) is $\text{argmin}_{a:a_0=1} a^t R_{m+1} a$
 $=$ 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 \Rightarrow
 \cdot *covariance method*

If R_{m+1} Toeplitz, Levinson-Durbin algorithm \Rightarrow
 \cdot *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, \dots, X_{n-1} ? Can estimate:

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

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

and “plug in.”

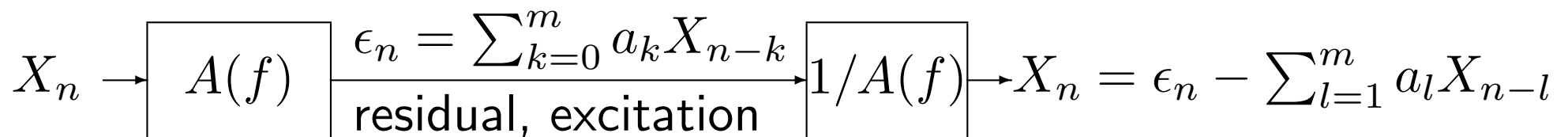
\hat{R}_m Toeplitz, \bar{R}_m not.

As $n \rightarrow \infty$, $\bar{R}_{m+1} \approx \hat{R}_{m+1}$

Processes and Filters

For $n = m, m + 1, \dots$ 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* $\Leftrightarrow A(f) = \sum_{n=0}^m a_n e^{-i2\pi n f}$



Limit: $m \rightarrow \infty$, the orthogonality principal \Rightarrow prediction error becomes uncorrelated — *white noise*!

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

A little algebra \Rightarrow

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

For large n $x^t A_n^t A_n x \approx a^t \hat{R}_{m+1} a$

Maximize $f_{X^n|a}(x) \Leftrightarrow \operatorname{argmin}_{a:a_0=1} a^t \hat{R}_{m+1} a$

LP problem again!

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

Maximum Entropy View

Suppose have estimate \hat{R}_{m+1} of correlations to lag m of stationary random process X_n .

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

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

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: Basic information theory \Rightarrow solution is Gaussian m -th order autoregressive process with variance

$$\sigma_{X_m|X^m}^2 = \frac{|R_{m+1}|}{|R_m|} = \alpha_m = \min_{a:a_0=1} a^t R_{m+1} a,$$

and power spectral density $S(f) = \alpha_m / |A(f)|^2$

LP problem again

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 m th order autoregressive processes by minimum distortion (nearest neighbor) rule.

Example: Modified *Itakura-Saito distortion measure* by

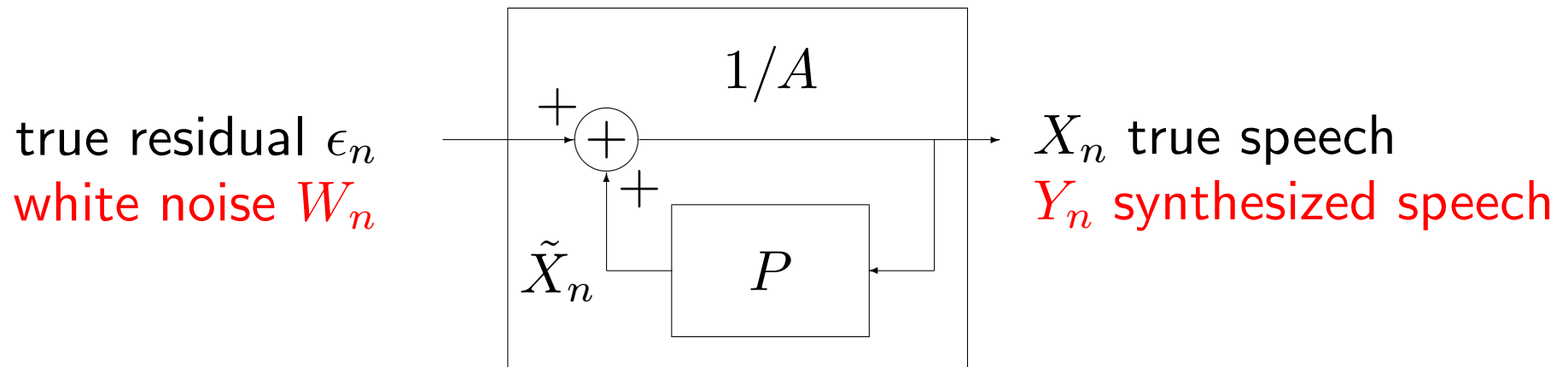
$$d(S, S_Y) = \int_{-1/2}^{1/2} \left(\frac{S(f)}{S_Y(f)} - \ln \frac{S(f)}{S_Y(f)} - 1 \right) = \frac{a^t R_{m+1} a}{\sigma_Y^2} - \ln \frac{\alpha_m}{\sigma_Y^2} - 1$$

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 $a = \operatorname{argmin}_{a:a_0=1} a^t R_{m+1} a$
and $\sigma_Y^2 = \alpha_m$. LP problem again!

Linear Predictive Coding

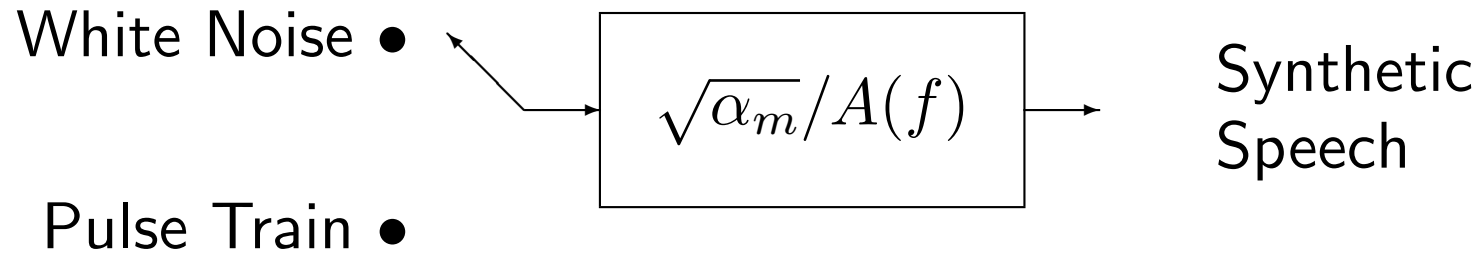


Matching correlation: $r_X(n) = r_Y(n), n = 0, 1, \dots, m$

\Rightarrow LPC *model*: m th-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 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.

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 [5] 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 [11, 12].

From [5]:

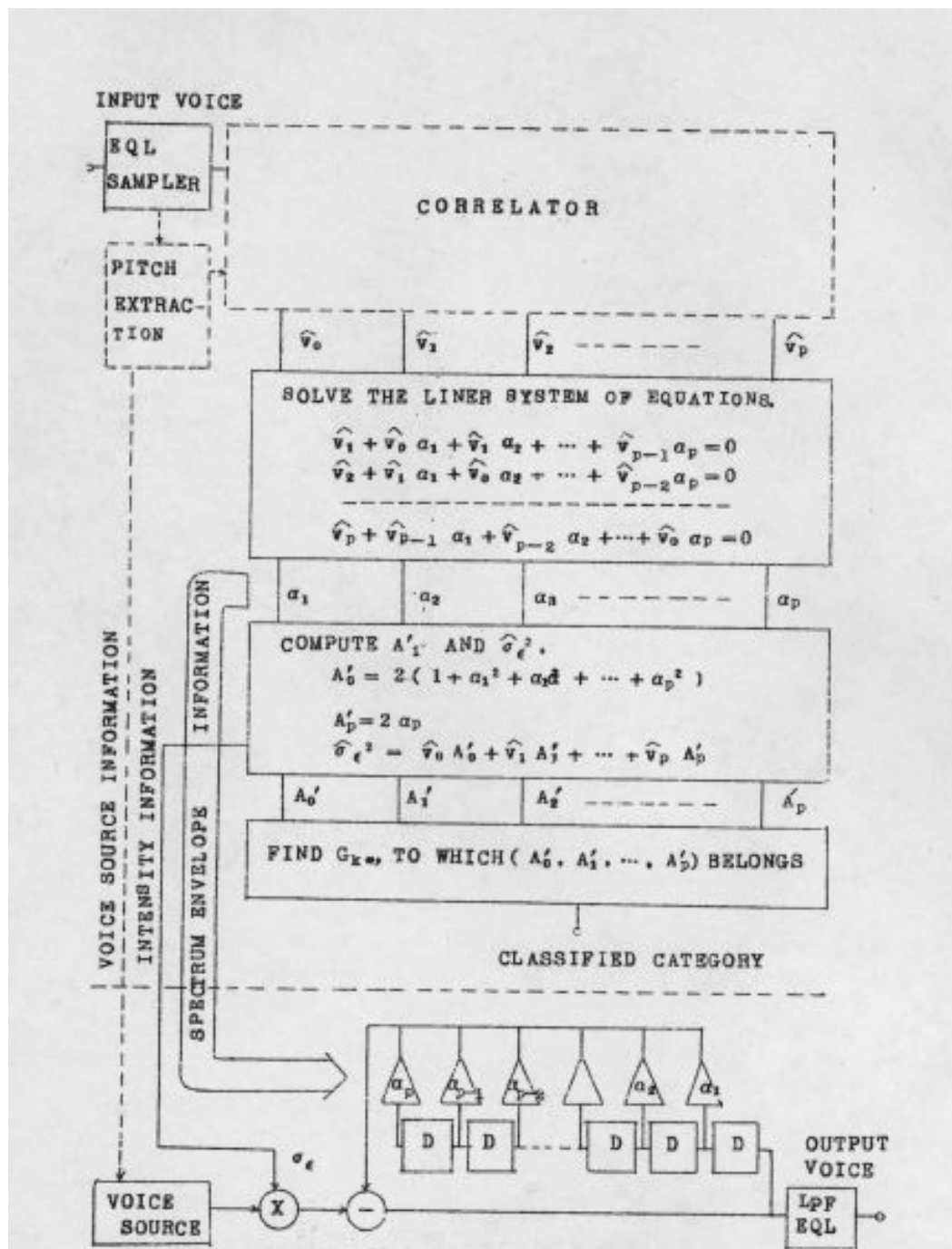


图 5. 新しいパラメータ伝送方式

1967



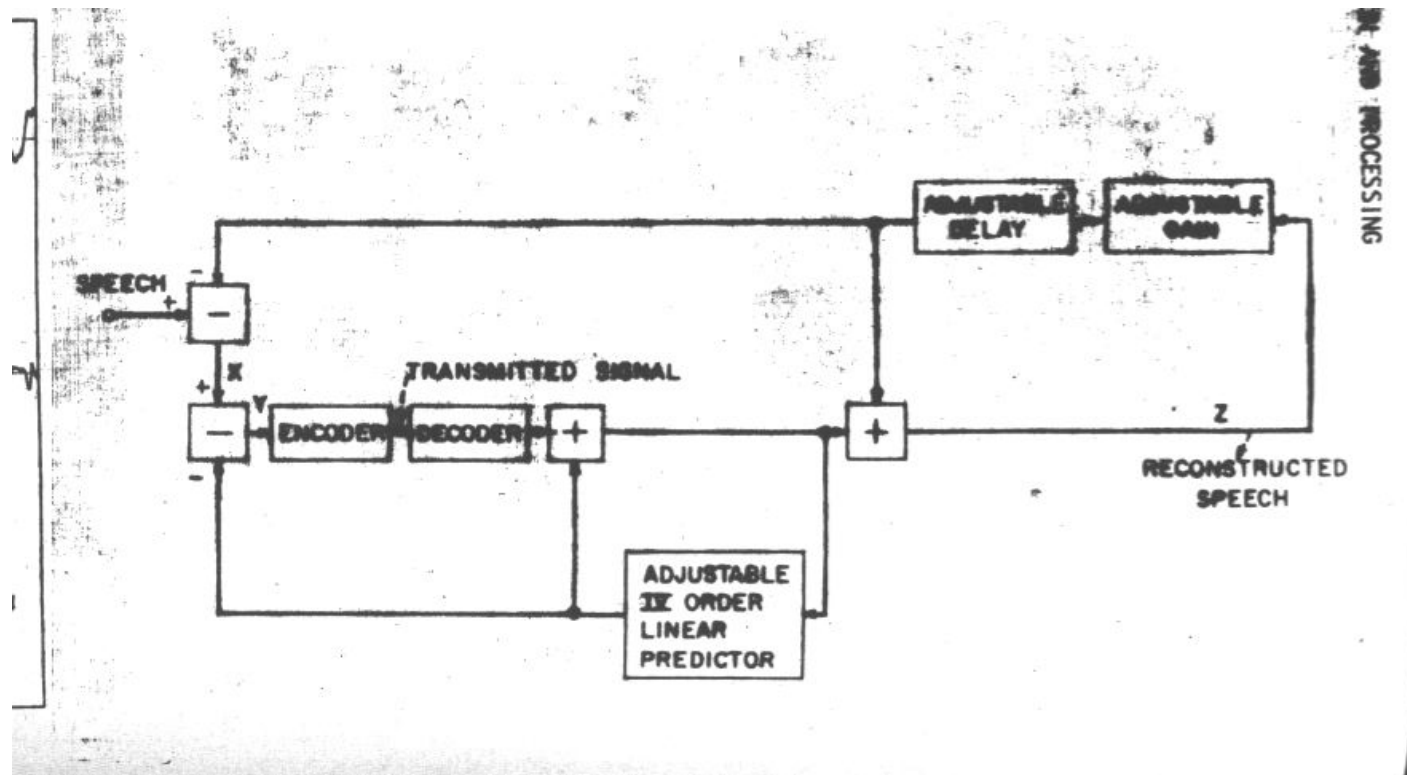
Ed Jaynes and John Burg.

October John Burg presents maximum entropy approach [9] and wins best presentation award at the meeting of the Society of Exploration Geophysicists. Focus is on prediction error properties. Variational, not parametric.

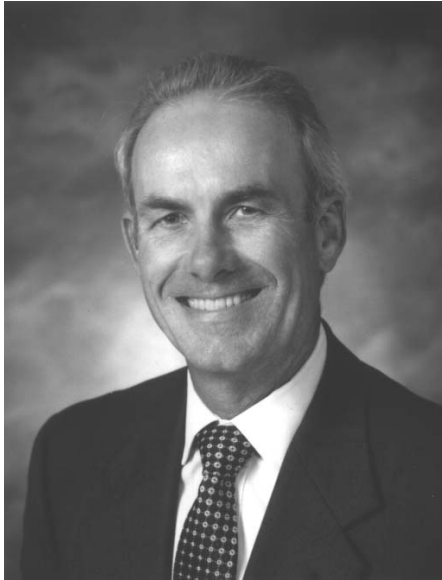
1967

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.

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

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) — the “node” of the ARPANET. BBN gets contract from ARPA to build and deploy 4 in January 1969. [72]

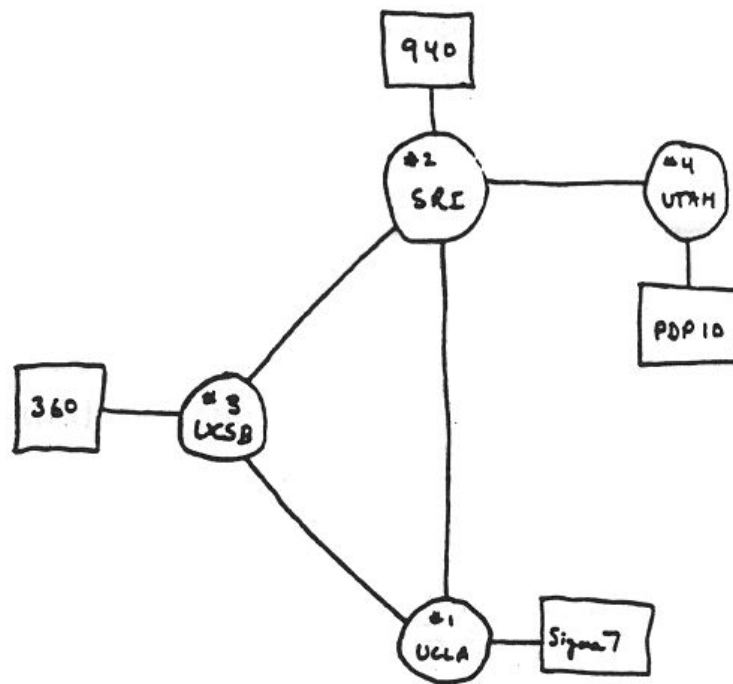
Culler cofounds Culler-Harrison Inc (CHI), which builds early array processors which are adopted and commercialized by FPS, will replace SPS-41 array processors.

1969

Itakura and Saito[12] introduce partial correlation (PARCOR) [1969] variation on autocorrelation method, finds partial correlation [1] coefficients. 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.



THE ARPA NETWORK

DEC 1969

4 NODES

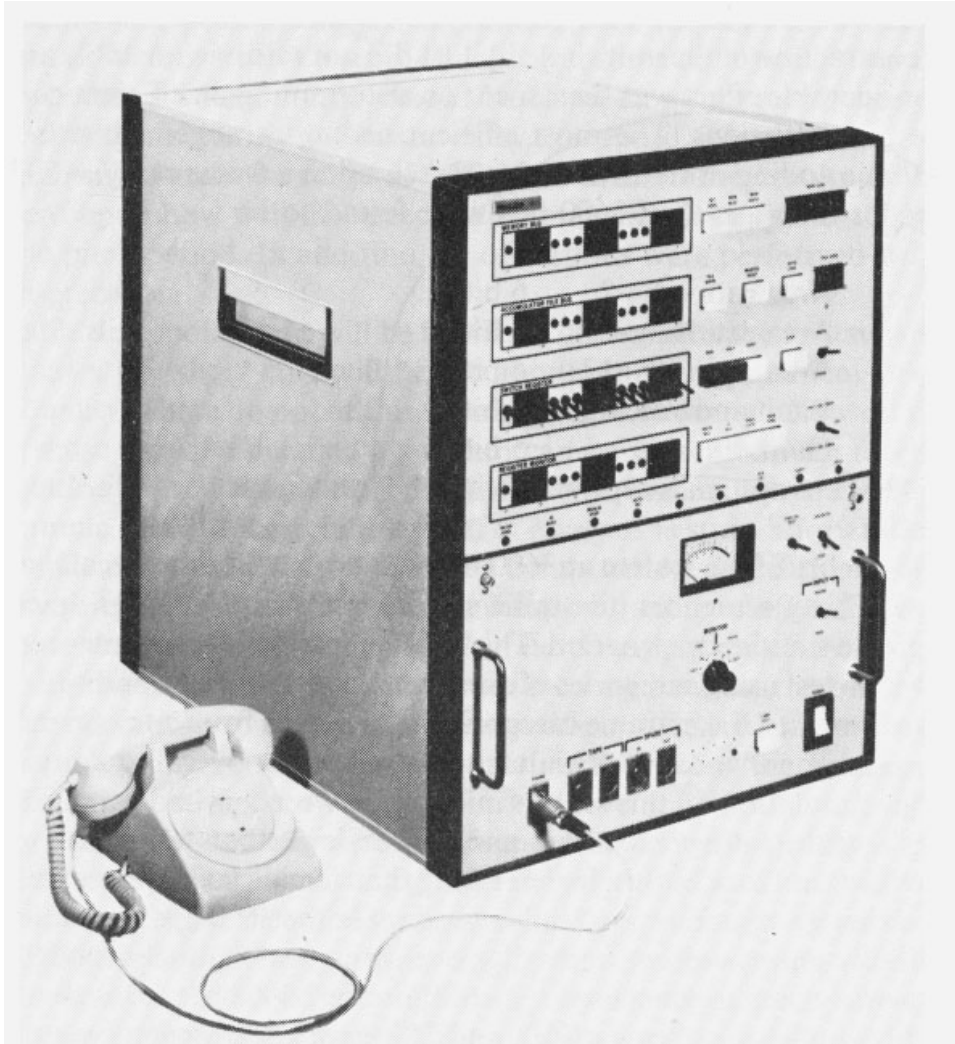
FIGURE 6.2 Drawing of 4 Node Network
(Courtesy of Alex McKenzie)

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[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 (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

Network Speech Compression Note #3
NIC 19946

RECEIVED
DEC 6 1973

Marcia Keeney
SRI-ARC
November 14, 1973

NETWORK SPEECH COMPRESSION GROUP MEMBERSHIP LIST

Steve F. Boll
University of Utah
Computer Science Dept.
3160 Merrill Engineering Building
Salt Lake City, Utah 84112

SFB
(801) 581-8576
UTAH-10

Dan Cohen
University of Southern California
Information Sciences Institute
4676 Admiralty Way
Marina Del Rey, California 90291

DC
(213) 822-1511
USC-ISI

Glenn J. Culler
Culler-Harrison, Inc.
150-A Aero Camino
Goleta, California 93017

GJC
(805) 968-1813
CHI2

Robert E. Kahn
Advanced Research Projects Agency
1400 Wilson Boulevard
Arlington, Virginia 22209

REK2
(202) 694-5921 or 694-5922
ARPA-TIP

D. T. (Tom) Magill
Stanford Research Institute
333 Ravenswood Avenue
Menlo Park, California 94025

DTM
(415) 326-6200 ext 2664
SRI

John Makhoul
Bolt Beranek and Newman Inc.
50 Moulton Street
Cambridge, Mass. 02138

JM
(617) 491-1850 ext 234
BBN-TENEX

John D. Markel
Speech Communications Research Lab, Inc.
800 Miramonte Drive
Santa Barbara, California 93109

JDM
(805) 965-3011
SCRL

Joseph Tierney
MIT Lincoln Laboratory
Lexington, Mass. 02173

JT3
(617) 862-5500 ext 277
IND

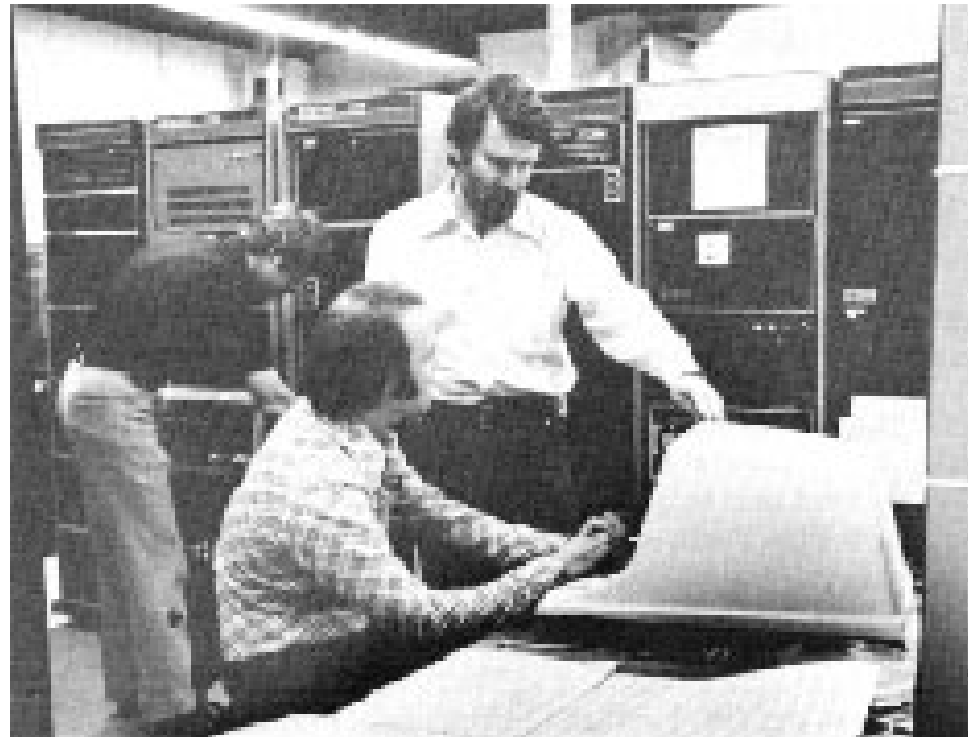
Original Members of
NSC: ISI, University
of Utah, BBN, MIT-
LL, SRI. Soon joined
by SCRL, CHI.

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.



Danny learns of LPC.

1973 Continued

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

Remember, DSP chips did not yet exist!

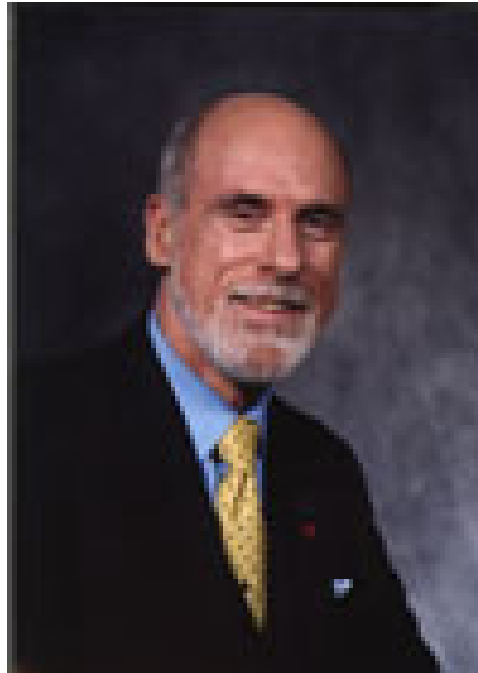
LL tradition is that first realtime 2400 bps LPC on the FDP done by Ed Hofstetter using Markel/Gray LPC formulation.

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.

1974

January Danny Cohen and Vint Cerf meet in Palo Alto, begin long discussion 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, 73].
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 would be a step backward for realtime applications on the ARPANET!

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

★ **December 1974** 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.

Effectively “voice over IP” before IP existed.

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



1976

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

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).”

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.”

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.

1977

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

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.[66, 74]

Patent never enforced.

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.

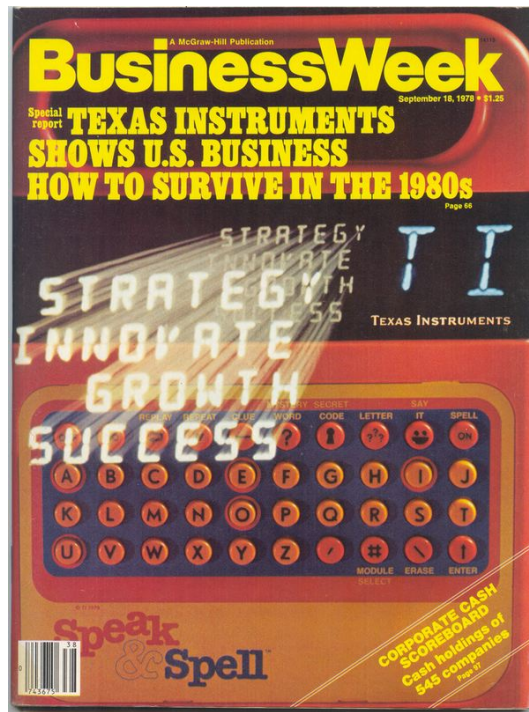
1978

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 nearly three decades ago!!*

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

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.”
- Barry Leiner et al. [73] “. . . 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.

- Vint Cerf and Bob Kahn won the 2004 ACM Turing Award for TCP/IP.

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