STREAMING IN 1970s

NVP & ST: the very first real-time streaming protocols

Yuriy Reznik
Brightcove, Inc.

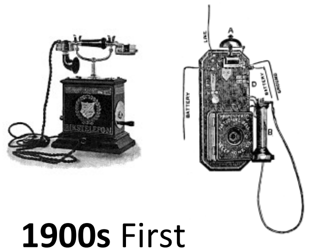
BRIGHTCOVE

TIMELINE

1850-1950s

The era of circuit-switching networks

1880s Telegraphs and first ideas and demonstrations of voice transmission

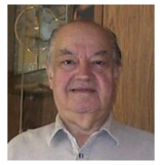


1900s First commercial telephones and telephone systems

1959 AT&T Picturephone

1960s

The invention of packet-switching networks.



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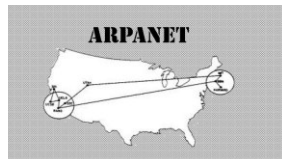

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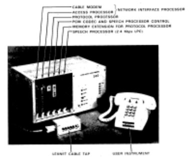

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1980 split of TCP:
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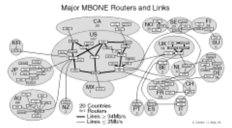
1990s

The era of UDP-based (RTP+RTSP) streaming



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2000s

The era of progressive downloads and TCP- and HTTP-based streaming (HLS, DASH)

2002-2007 Flash adds video playback capability. Popularizes progressive downloads.

2005 YouTube launched. Uses progressive downloads.

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2024 L3D DASH

2012-2021 QUIC development and adoption (RFC 9000)

2022-present Media over QUIC (MoQ) development

TIMELINE

Main focus of this talk

1850-1950s	1960s	1970s–1980s	1990s	2000s	2010s–2020s
The era of circuit-switching networks	The invention of packet-switching networks.	The rise of ARPAnet. Design of first systems for voice transmission over packet networks. NVP and ST protocols. TCP and its evolution.	The era of UDP-based (RTP+RTSP) streaming	The era of progressive downloads and TCP- and HTTP-based streaming (HLS, DASH)	Evolution toward low-delay and non-TCP-based streaming.

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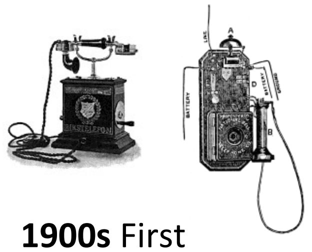

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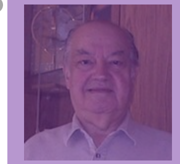

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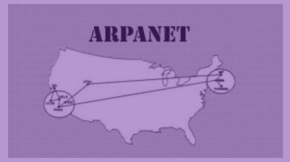

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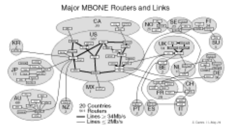

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1850-1950s | 1960s | 1970s | 1980s | 1990s | 2000s | 2010s | 2020s

PACKET SWITCHING

- Motivation 1 (Paul Baran, RAND, 1962)
  - It is needed to increase survivability (resilience) of tactical communication systems

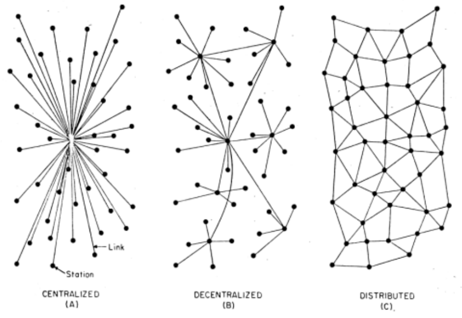


FIG. 1 — Centralized, Decentralized and Distributed Networks

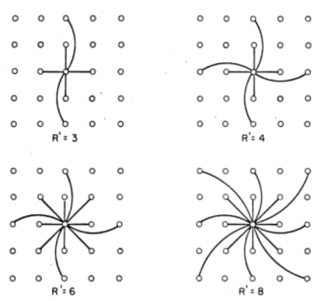


FIG. 2 – Definition of Redundancy Level

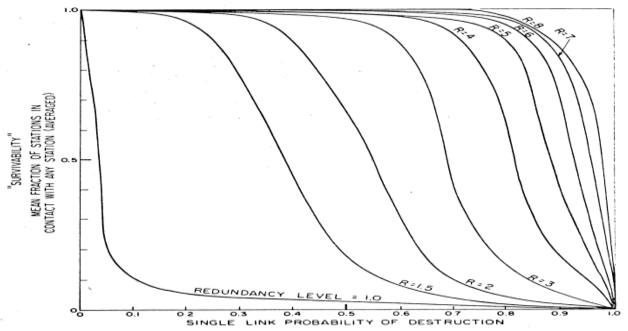


FIG. 5 — Perfect Switching in a Distributed Network — Sensitivity to Link Destruction, 100% of Nodes Operative.

- Motivation 2 (Lenny Kleinrock, MIT, 1959)
  - Circuit switching is problematic because *data communications* is bursty, that is, it is typically dominated by short bursts of activity with long periods of inactivity.
  - Any static assignment of network resources, as is the case with circuit switching, would be extremely wasteful of those resources, whereas dynamic assignment (on demand sharing) would be highly efficient.

Origins of packet switching:

L. Kleinrock, "Message Delay in Communication Nets with Storage," Ph.D. dissertation, MIT, Cambridge, MA, 1962.

P. Baran, "On Distributed Communication Networks," Rand Paper P-2626, Sept. 1962.

D. W. Davies, "Proposal for a Digital Communication Network," unpublished memo, June 1966,

D. W. Davies et al., "A Digital Communication Network For Computers Giving Rapid Response at Remote Terminals," ACM Symp. Op. Sys. Principles, Gatlinburg, TN, Oct. 1967.

D. W. Davies, "The Principles of a Data Communication Network for Computers and Remote Peripherals," Proc. IFIP Hardware, Edinburgh, 1968.

L. Kleinrock, "An early history of the internet [History of Communications]." IEEE Communications Magazine 48, no. 8 (2010): 26-36.


Leonard Kleinrock
CUNY, MIT, UCLA
1934 -

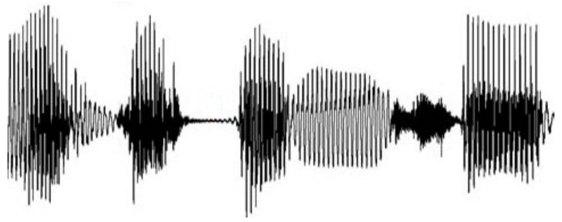

Paul Baran
UCLA, RAND Corp.
1926 - 2011


Donald Davies
Imperial College, NPL
1926 - 2011

SPEECH COMMUNICATION

- **Speech ≠ data** !

- A mix of voiced and silence intervals
- The average rate is ~ 40-50% from peak
- Max. duration of a talkspurt ~ 20sec



- Impacts of delays:
  - If the overall round-trip delay is less than about l/4 second, conversations are carried out in a "normal" fashion with considerable feedback from "listener" to "talker" taking place.
  - When greater delay is experienced, people switch to a more formal mode in which feedback utterances are suppressed, and the listener generally waits until the talker indicates that he has finished before saying anything.
  - User satisfaction declines with increasing delay, but systems remain usable for delays as long as several seconds.
- Impacts of packet losses:
  - The loss of small (50 msec or less) chunks of speech produces a degradation of quality, but sentence intelligibility tends to be preserved up to fairly high percentage losses.
  - Larger chunks of speech represent whole syllables or words, and their loss can change the meaning of sentences.

Some references

James W. Forgie, ST - A Proposed Internet Stream Protocol, IEN 119, 7 September 1979 https://www.rfc-editor.org/ien/ien119.txt

C. Weinstein and J. Forgie, "Experience with Speech Communication in Packet Networks," JSAC 1/6,1983.

L. Delgrossi, and L. Delgrossi, “The Internet Stream Protocol,” in Design of Reservation Protocols for Multimedia Communication, Springer, 1996, pp.49-63.

A.M. Kondoz, “Digital speech: coding for low bit rate communication systems,” John Wiley & Sons; 2005.

R.M. Gray, “Linear Predictive Coding and the Internet Protocol,” Now publishers, Boston, 2010.

NVP = NETWORK VOICE PROTOCOL

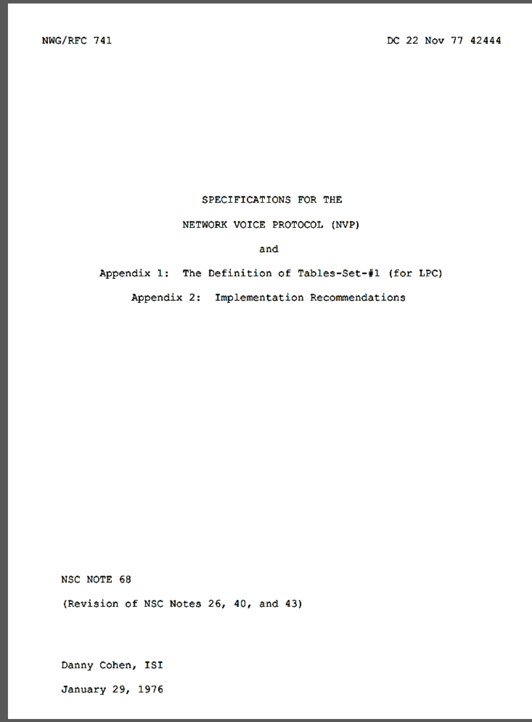
- Developed in 1973-1976 by Danny Kohen, USC / ISI as part of ARPA “Packet Speech” project
- NVP defined:
  - call initiation and termination, including negotiation of voice encoder compatibility and handling of ringing and busy conditions
  - packetization of voice for transmission, with the time stamps and sequence numbers needed for speech reconstitution at the receiver
  - speech playout with buffering to smooth variable packet delays
- Design features
  - Separate control and data messages
  - Avoidance of retransmissions
  - Independence from lower-level protocols (ARPANET NCP)
- First demonstrated operation
  - USC-MIT voice call in August 1974

First voice over packet network system

Prototype telephone for testing the Network Voice Protocol, MIT Lincoln Laboratory, 1974

D. Cohen, Specification for the Network Voice Protocol (NVP), RFC 741, January 29, 1976.

Danny Cohen
Harvard, Caltech, USC/ISI, Sun
1937-2019



NWG/RFC 741 DC 22 Nov 77 42444

SPECIFICATIONS FOR THE

NETWORK VOICE PROTOCOL (NVP)

and

Appendix 1: The Definition of Tables-Set-#1 (for LPC)

Appendix 2: Implementation Recommendations

NSC NOTE 68

(Revision of NSC Notes 26, 40, and 43)

Danny Cohen, ISI

January 29, 1976

NVP – EXAMPLE CONTROL SEQUENCES

Handshake

```
Here is an example for a connection:

   (377)  C: 1,<WHO>,<WHOM>,340    Please talk to me on 340/341.

   (340)  A: 2,1                   I refuse, since I'm busy.

Another example:

   (377)  C: 1,<WHO>,<WHOM>,360    Please talk to me on 360/361.

   (360)  A: 6,350                 OK.  You talk to me on 350/351.

   (350)  C: 1,<WHO>,<WHOM>        I want to talk to you.

   (360)  A: 3,1,1,2               Can you do CVSD?  (ANSWERER tries
                                   to be the NEGOTIATION MASTER)

   (350)  C: 12,1                  I want to be it.

   (360)  A: 13,1                  That's OK with me.
```

Codec capability negotiation

```
   (350)  C: 3,1,1,2               Can you do CVSD?

   (360)  A: 5,1,1                 No, but I can do LPC.

   (350)  C: 3,1,1,3               Can you do RELP?

   (360)  A: 5,1,1                 No, but I can do LPC.

   (350)  C: 3,1,1,1               How about LPC?

   (360)  A: 4,1,1                 LPC is fine with me.

   (350)  C: 3,2,1,150             Can you use 150 microseconds
                                   sampling?

   (360)  A: 4,2,150               I can use 150 microseconds.
```

Bitrate negotiation

```
   (350)  C: 3,4,3,976,1040,2016   Can you use 976, 1040, or 2016
                                   bits/msg?

   (360)  A: 4,4,976               I can use 976.

   (350)  C: 3,5,1,10              Can you send 10 coefficients?

   (360)  A: 4,5,10                I can send 10.
```

Frame length negotiation

```
   (350)  C: 3,6,1,64              Can you use a 64 sample
                                   transmission?

   (360)  A: 4,6,64                I can use 64.

   (350)  C: 3,7,2,1,2             SIMPLE or OPTIMIZED acoustic
                                   coding?

   (360)  A: 4,7,2                 OPTIMIZED!

   (350)  C: 3,8,1,1               Can you do SIMPLE info coding?

   (360)  A: 4,8,1                 I can do SIMPLE.

   (350)  C: 3,9,1,58              mu = 0.90625?

   (360)  A: 4,9,58                Fine with me.

   (350)  C: 3,10,1                Table set #1?

   (360)  A: 4,10,1                Of course!
```

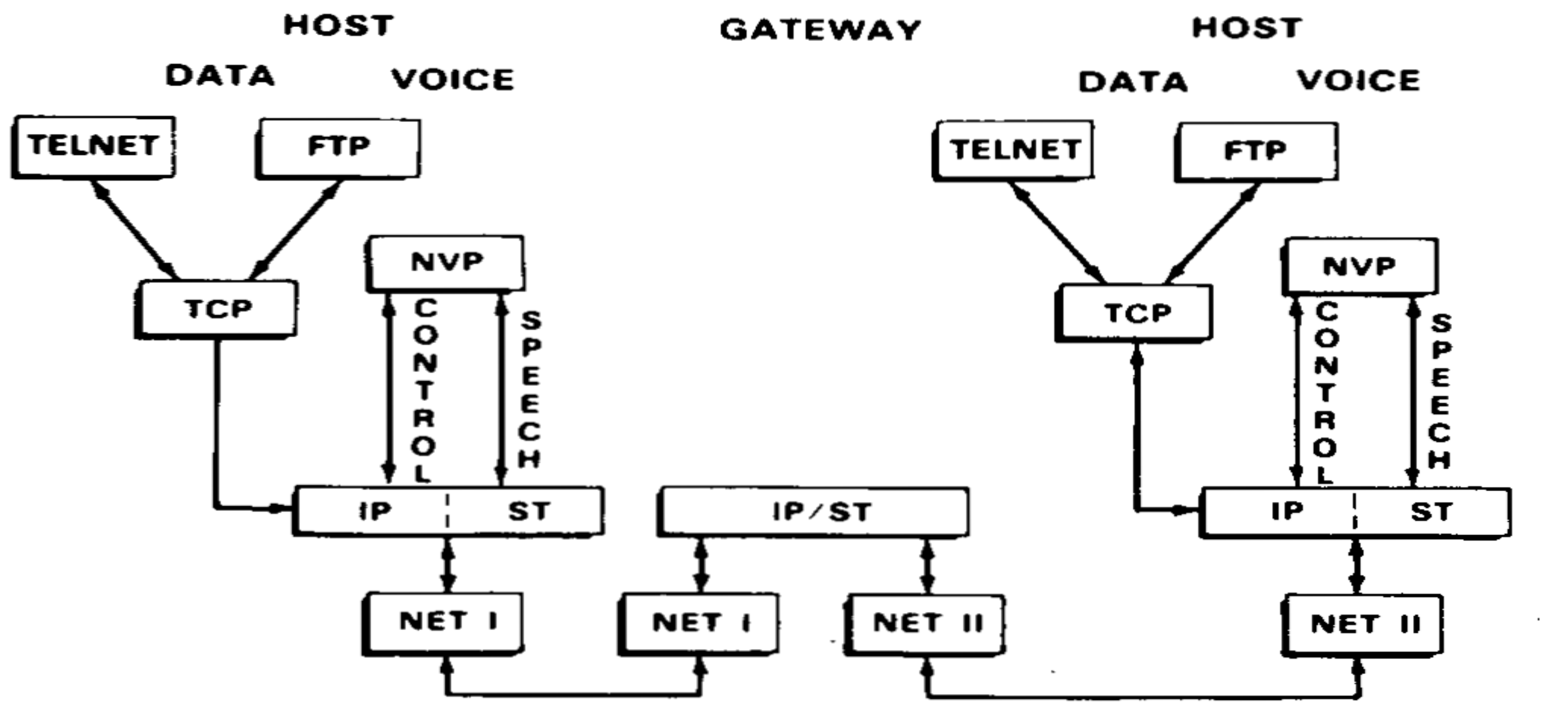
Session start

```
   (350)  C: 6                     I am ready.  (Note:  No "RINGING"
                                   sent)

   (350)  C: 8                     And you?

   (360)  A: 6                     I am ready, too.

     .......                       Data is exchanged now,

     .......                       on 351 and 361.

   (350)  C: 10,1234               Echo it, please.

   (360)  A: 11,1234               Here it comes!

     .......

   (360)  A: 10,3333               Now ANSWERER wants to measure

   (350)  C: 11,3333               ...the delays, too.

     .......

   (???)    X: 2,3                 Termination by either user.
```

ST = STREAM PROTOCOL

BRIGHTCOVE

- Developed in 1973-1979 by Jim Forgie, Danny Kohen, and Estil Hoversten
- Replaced IP with ST packets, forcing restricted (typically fixed for session duration) routing



- Key design objectives:
  > Statistically defined delay and bandwidth of speech connections
  > Minimization of header overhead of the traditional IP

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MIT, MIT Lincoln Lab
1929 - 2011

Danny Cohen
Harvard, Caltech, USC/ISI, Sun
1937-2019

Estil Hoversten
Iowa State, MIT
1936-2021

BRIGHTCOVE

ST – MOTIVATIONS

- **Conventional datagram networks are unsatisfactory for speech communication** except under conditions of light overall load or where speech constitutes a small fraction of the overall load and can be given priority service.
- The difficulty with datagram nets comes from their **inability to provide the controlled delay** and **guaranteed data rate** required for speech. Delay increases with offered load, slowly at light load, but dramatically as average load approaches capacity. Flow control strategies tend to be aimed at buffer management and fairness goals, both of which will operate to restrict the effective data rate available to an individual user as load increases. Traffic control strategies are mainly concerned with congestion control and are primarily defensive, resulting in offered datagrams being held off or refused when difficulties are detected. Unfortunately for the speech user, by the time congestion is detected, it is already too late. For satisfactory speech service, congestion due to overload must be prevented. **Since a datagram net has no knowledge of the a priori requirements of users, it cannot develop traffic control strategies to meet these requirements.**
- Another disadvantage of datagrams for speech is their **packet efficiency**. The speech content of an individual user packet can be anything from 50 or so bits up to 1200 or 1300 bits depending upon the speech digitization technique in use. **The need to carry full source and destination addresses as well as other packet-handling information in each packet penalizes datagrams relative to other packet and circuit switching techniques.** In the internet case the penalty is worse since two sets of header information have to be carried. For example, IP datagrams on SATNET carrying 40-msec chunks of 16-kbps speech (a reasonable chunk size and popular data rate) would have a packet efficiency of about 56% and would require utilization factors of about 80% to break even with respect to circuit switching. It is unlikely that delay characteristics would be satisfactory at this level of load.

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FIRST ARPANET VOICE TRANSMISSIONS

- 1971: First experiments using PCM speech and simulated network. MIT LL
- August 1974: First voice call over ARPAnet between USC/ISI and MIT LL
  - Codec: 9.6 kbits/CVSD
- December 1974: First call over ARPAnet using LPC vocoder
  - Sites: MIT LL and CHI (UCSB)
  - Codec: LPC, 3.5kbits
- January 1975: First multi-way voice conferencing over ARPAnet
  - Sites: CHI, ISI, LL, and SRI
  - Codec: LPC, 3.5kbits

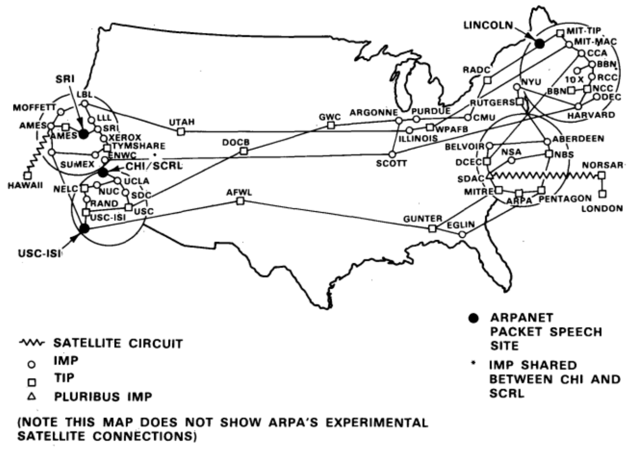
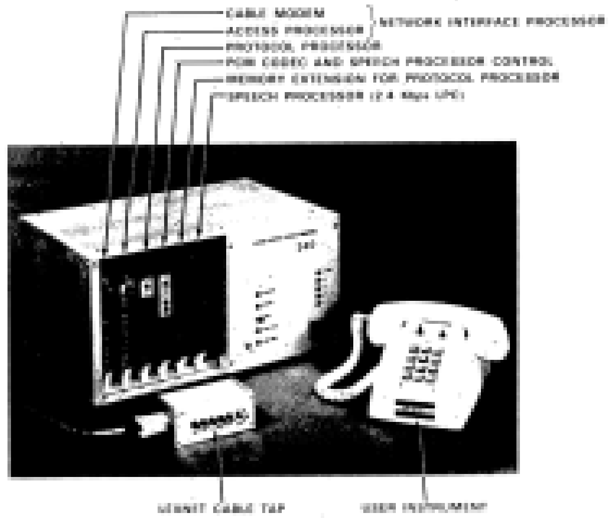
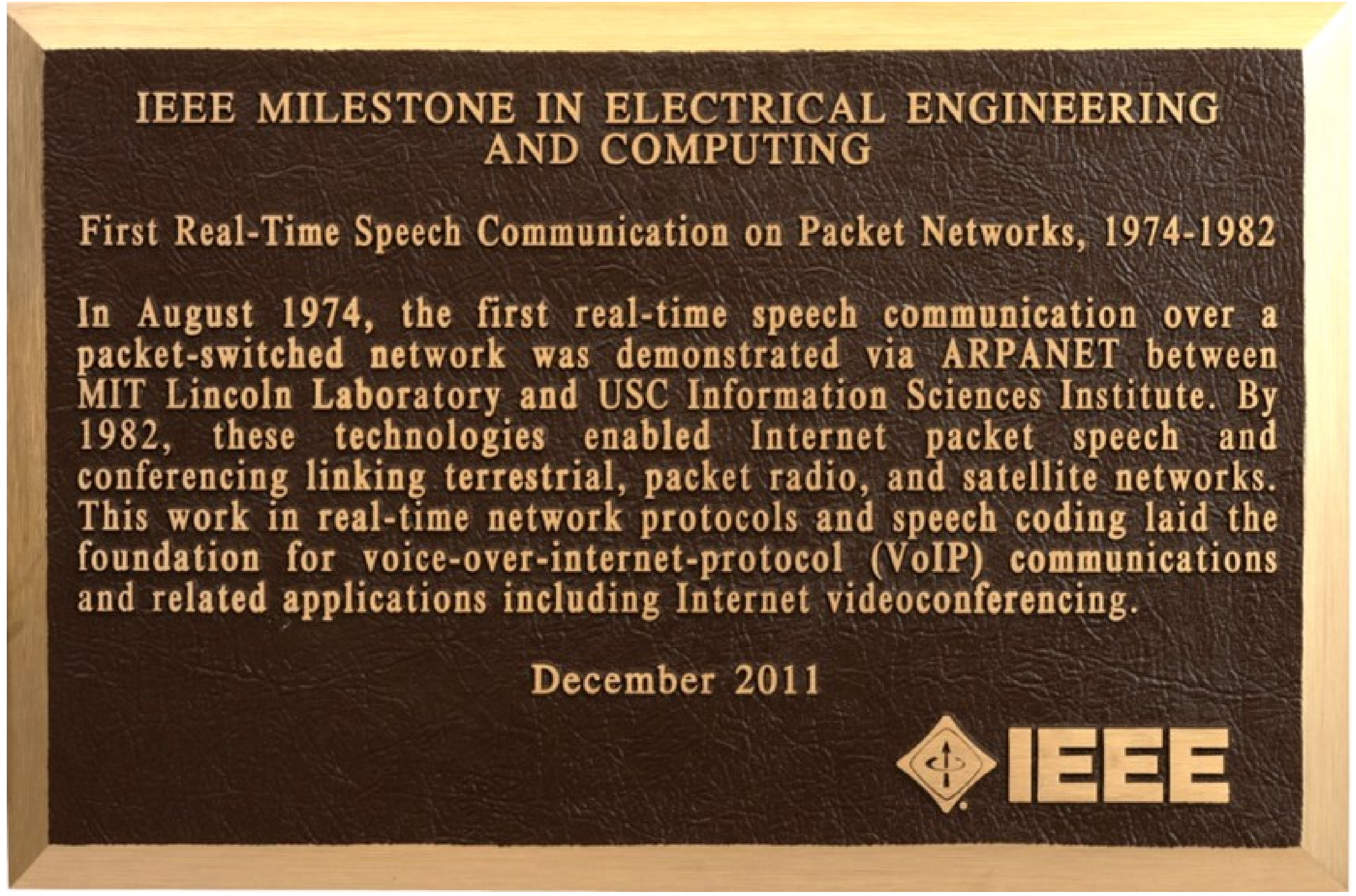


Fig. 5. Geographic map of the ARPANET, as of June 1975, showing locations of ARPANET packet speech sites.





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TIMELINE

UDP-based streaming

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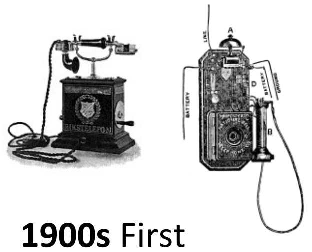

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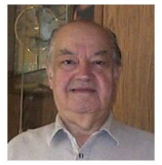

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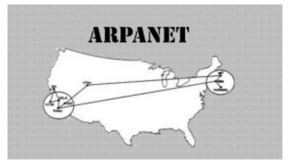

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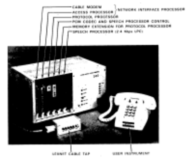

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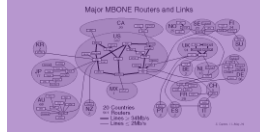

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TIMELINE

Longer delay, TCP+HTTP-based streaming

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The era of circuit-switching networks

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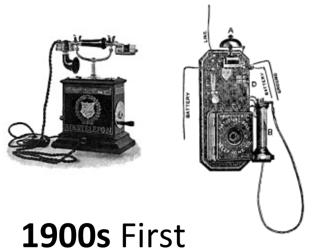

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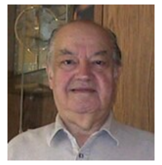

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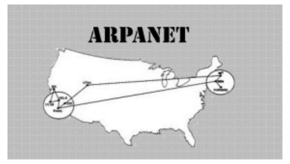

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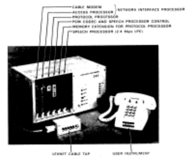

1974 , First voice call over packet-switched network. MIT-USC

1974 V. Cerf & R. Kahn, "A protocol for Packet Net. Comm." (TCP v1)
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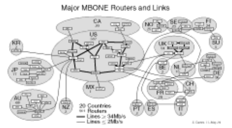

1993, Live streaming over MBONE network.

1995 RealAudio, First purposedly-built streaming application with pre-roll delay

1995-1999 Other early streaming systems.

1996 RTP: RFC 1889
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The era of progressive downloads and TCP- and HTTP-based streaming (HLS, DASH)

2002-2007 Flash adds video playback capability. Popularizes progressive downloads.

2005 YouTube launched. Uses progressive downloads.

2007 Move Networks HTTP-based streaming

2008 HTML 5 released Beginning of plugin-free trend

2009 Apple HTTP live streaming (HLS)

2010 Microsoft Smooth Streaming

2012 MPEG DASH ISO/IEC 23003-1

Evolution toward low-delay and non-TCP-based streaming.

2011 WebRTC Browser-supported VoIP and video conferencing

2017 Low-latency DASH (DVB report)
2020 Low-latency HLS
2020 Low-latency modes of DASH (DASH-IF IOP guidelines)
2024 L3D DASH

2012-2021 QUIC development and adoption (RFC 9000)

2022-present Media over QUIC (MoQ) development

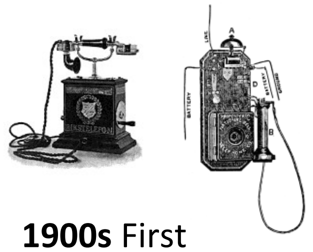
BRIGHTCOVE

TIMELINE

Back to low-delay (and UDP-based?) streaming

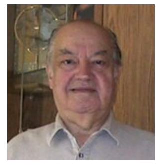
1850-1950s	1960s	1970s	1980s	1990s	2000s	2010s	2020s
The era of circuit-switching networks	The invention of packet-switching networks.	The rise of ARPAnet. Design of first systems for voice transmission over packet networks. NVP and ST protocols. TCP and its evolution.		The era of UDP-based (RTP+RTSP) streaming	The era of progressive downloads and TCP- and HTTP-based streaming (HLS, DASH)		Evolution toward low-delay and non-TCP-based streaming.

1880s Telegraphs and first ideas and demonstrations of voice transmission



1900s First commercial telephones and telephone systems

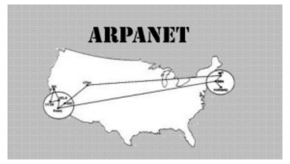
1959 AT&T Picturephone



1962 Paul Baran, "On Distributed Communication Networks" RAND

1963 Lenny Kleinrock: "Message delay in communication nets with storage", MIT

1966 Donald Davies, "Proposal for a Digital Communications Network," NPL



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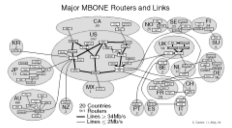

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PARALLELS WITH 1970S

- We again realize that for low-delays we have to send data in smaller chunks
  - From video segments – to shorter segments or frames
- The progress in device compute capabilities and video signal processing (super-resolution, frame-rate up-conversion, GenAI) allows more effective concealment and regeneration of missing parts than it was possible before.
  - GenAI today = LPC speech synthesis of 1970s
- This enables the use of UDP (or QUIC with no retransmissions) for video, and promises some improvements in the efficiency of video transmissions.
- However, we still have largely unpredictable behavior of IP with respect to bandwidth and latencies!
  - Engineers in 1970s have suggested to solve it by replacing the IP with ST.
  - What present generation of engineers will do?

THANK YOU