INTRODUCTION TO TELEPHONY & VOIP

Advanced Internet Services (COMS 6181 – Spring 2015)

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Overview

- The Public Switched Telephone System (PSTN)
- VoIP as black phone replacement → interactive communications enabler
- Presence as a service enabler
- Peer-to-peer VoIP

Name confusion

- Commonly used interchangeably:
  - ***Voice-over-IP (VoIP)*** – but includes video
  - Internet telephony – but may not run over Internet
  - IP telephony (IPtel)
- Also: VoP (any of ATM, IP, MPLS)
- Some reserve Internet telephony for transmission across the (public) Internet
- Transmission of telephone services over IP-based packet switched networks
- Also includes video and other media, not just voice

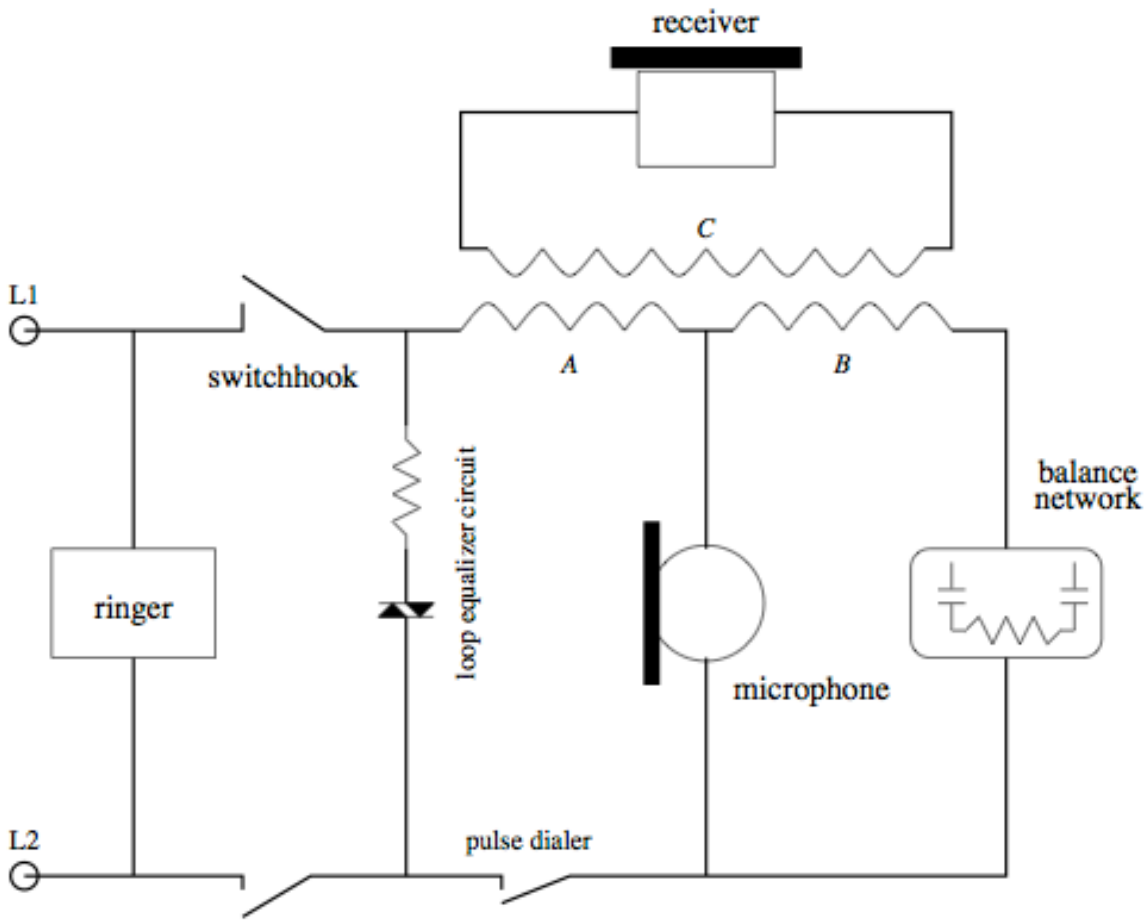
A bit of history

- 1876 invention of telephone
- 1915 first transcontinental telephone (NY–SF)
- 1920’s first automatic switches
- 1956 TAT-1 transatlantic cable (35 lines)
- 1962 digital transmission (T1)
- 1965 1ESS analog switch
- 1974 Internet packet voice (2.4 kb/s LPC)
- 1977 4ESS digital switch
- 1980s Signaling System #7 (out-of-band)
- 1990s Advanced Intelligent Network (AIN)
- 1992 Mbone packet audio (RTP)
- 1996 early commercial VoIP implementations (Vocaltec); PC-to-PC calling

Phone system

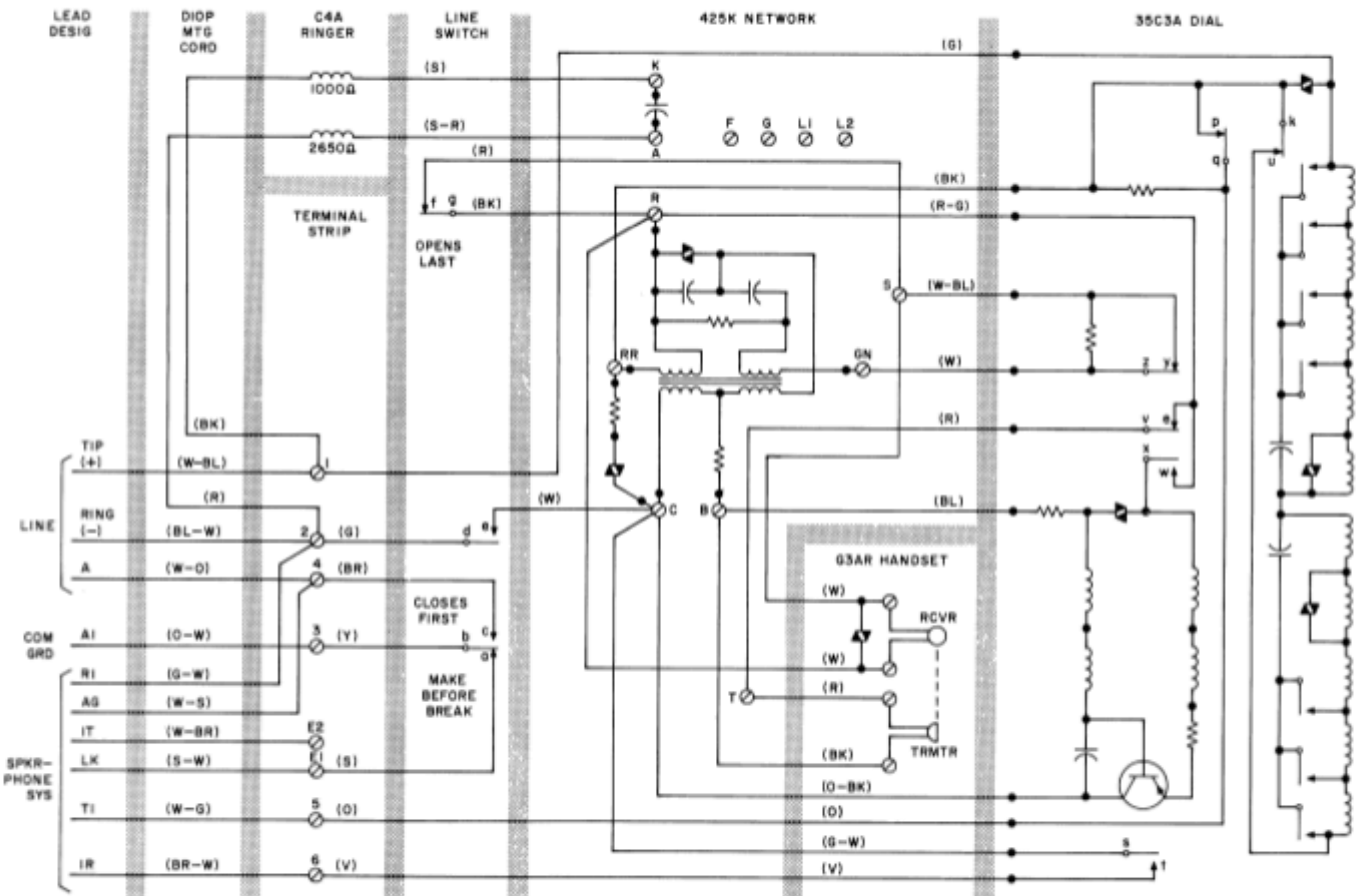
- analog narrowband circuits to “central office”
  - 48 Volts DC supply
- 64 kb/s continuous transmission, with compression across ocean
- µ-law: 12-bit linear range → 8-bit bytes
- everything clocked at a multiple of 125 µs
- clock synchronization → framing errors
- old AT&T: 136 “toll”switches in U.S.
  - interconnected by T1 and T3 digital circuits → SONET rings (AT&T: 50)
- call establishment “out-of-band” using packet-switched *signaling system (SS7)*

Circuit diagram



ringing: 25 Hz, 50 V AC

W 500 diagram



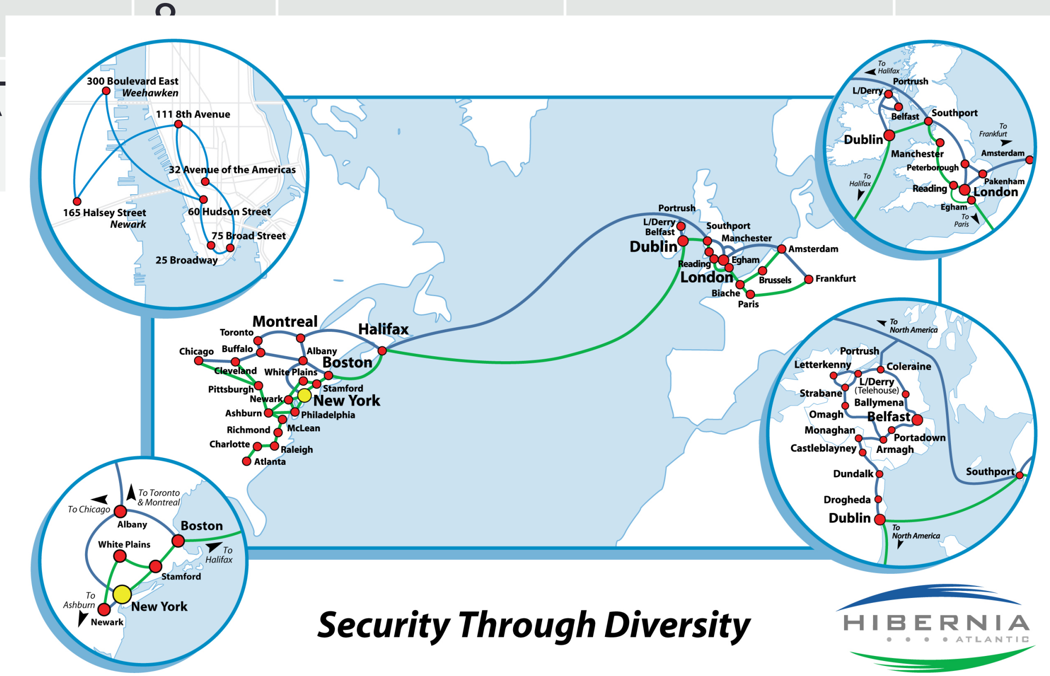
NOTE:
FURNISHED WITH RINGER CONNECTED FOR BRIDGED SERVICE. FOR RING PARTY MOVE RINGER (BK) LEAD FROM TERM. 1 TO TERM. 3. FOR TIP PARTY MOVE RINGER (R) LEAD FROM TERM. 2 TO TERM. 3.

Transatlantic cable systems

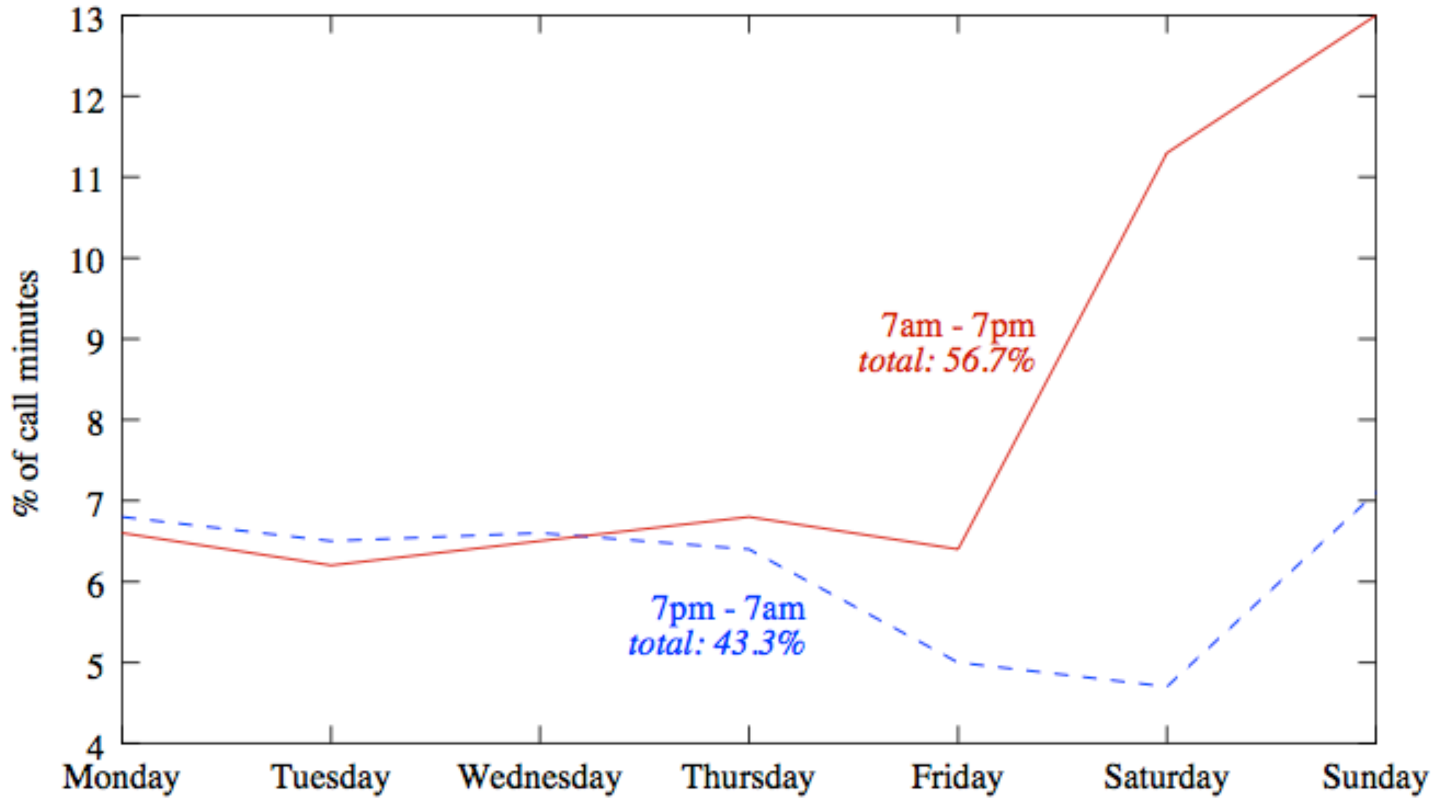
System	Year (use)	technology	cost ($M)	circuits	$/circuit	$/minute
TAT-1	1956-78	Coax + tubes	$49.6	40	213,996	2.443
TAT-2	1569	Coax	$42.7	44	167,308	1.910
TAT-3	1963	Coax	$50.6	79	111,027	1.267
TAT-4	1965	Coax	$50.4	62	140,238	1.601
TAT-5	1970	Coax	$70.4	648	18,773	0.214
TAT-6	1976-94	Coax	$197	3,200	10,638	0.121
TAT-7	1978-94	Coax	$180	3,821	8,139	0.093
TAT-8	1988-02	Fiber (20 Mb/s)	$360	6,048	10,285	0.117
TAT-9	1992-04	Fiber	$406	10,584	6,628	0.076
TAT-10	1992-03	Fiber (2x565 Mb/s)	$300	18,144	2,857	0.033
TAT-11	1993-04	Fiber (2x565 Mb/s)	$280	18,144	2,667	0.030
TAT-12	1996-08	Fiber ring (5 Gb/s)	$378	60,480	1,080	0.012
TAT-13	1996-08	Fiber (2x5 Gb/s)	$378	60,480	1,080	0.012

Transatlantic cable systems

System	Year	technology	cost ($M)	circuits	$/circuit	$/minute
TAT-13	199 6	Fiber	$378	60,480	1,080	0.012
Gemini	199 8	Fiber	$520	214,920	371	0.004
AC-1	199	120 Gb/s	$850	483,840	304	0.003
TAT				5M	<75	0.001



Call load over the week



Signaling System #7

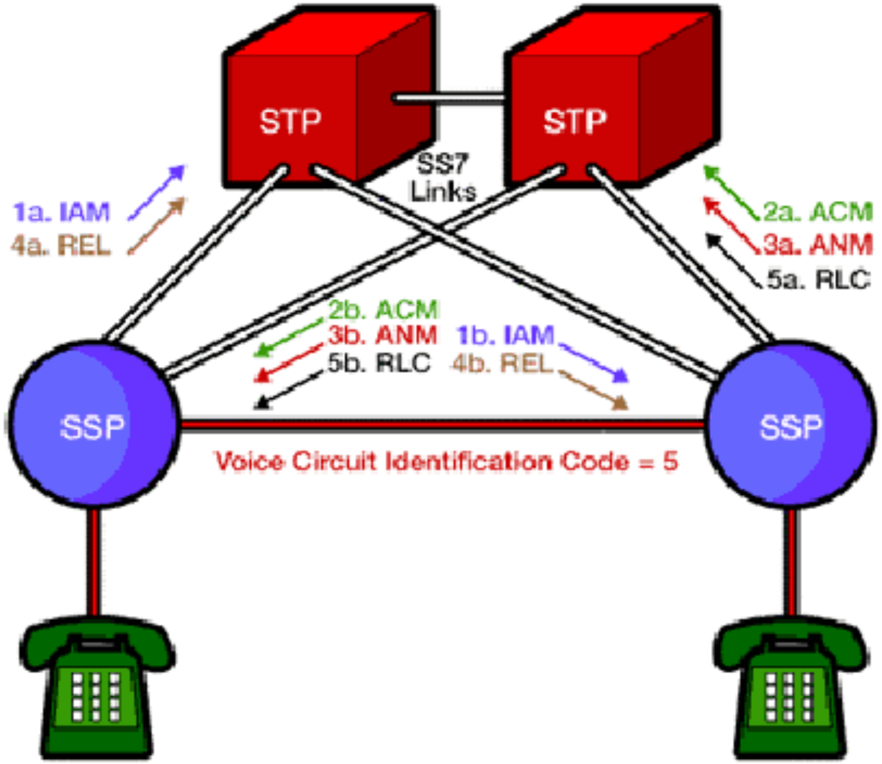
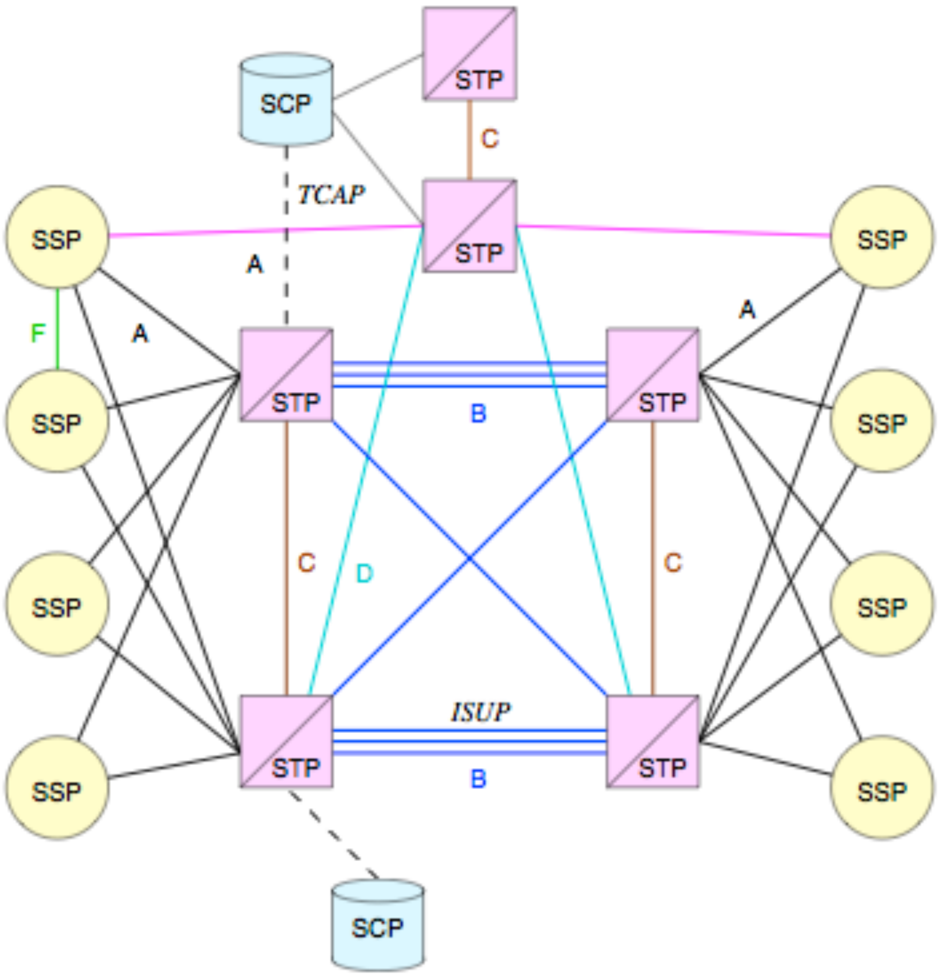
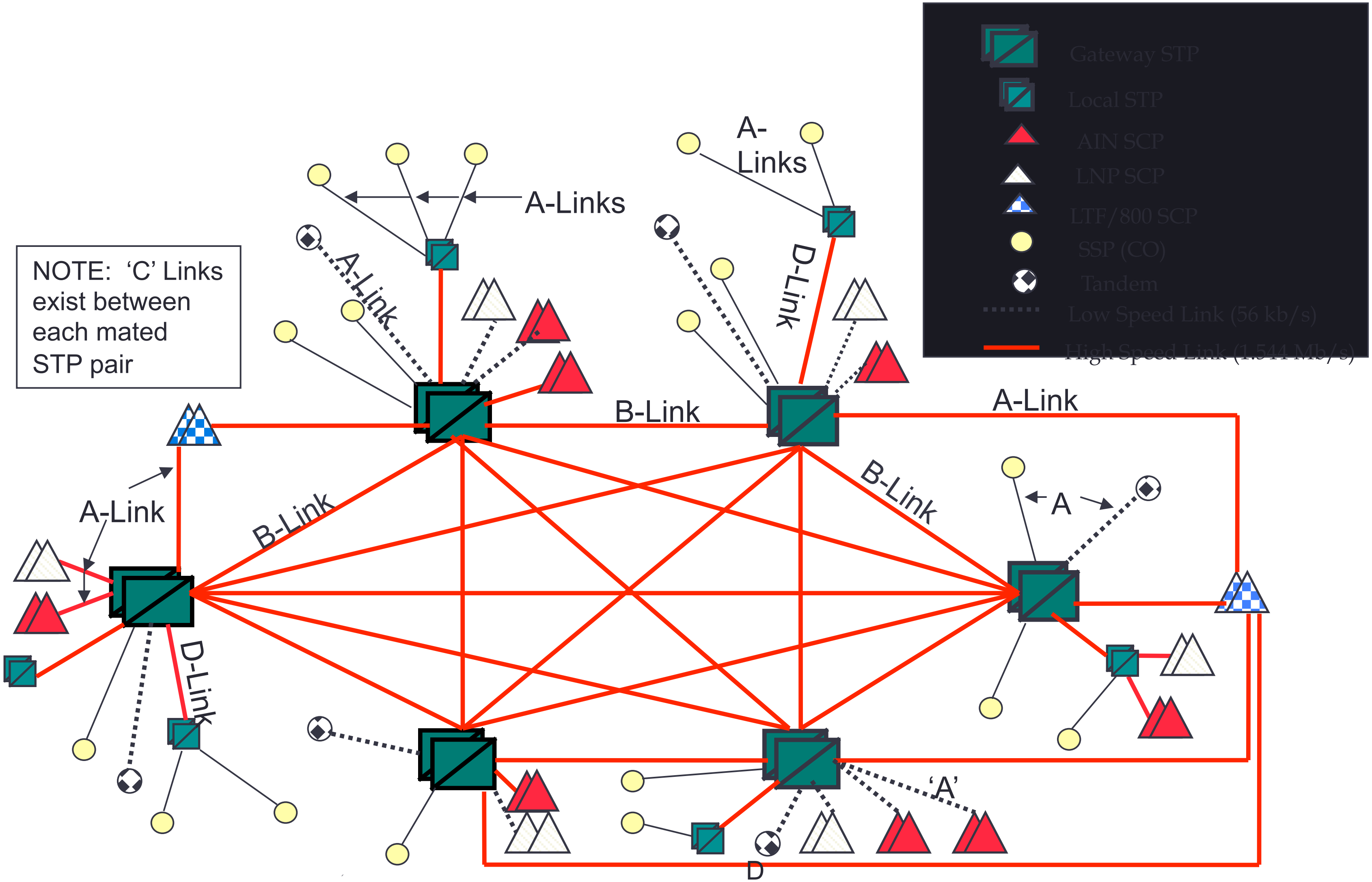


Figure 8. Basic ISUP Signaling

SS7 network


SCP
STP
C
TCAP
SSP
STP
SSP
A
F
A
A
STP
STP
SSP
SSP
B
C
D
C
SSP
SSP
ISUP
SSP
STP
STP
SSP
B
SCP

Typical signaling network



D. Finn (BellSouth 2006)

Types of switching entities

- Class 5 End Office (or C. O.)
  - Connects subscribers' telephone lines to the telecommunications network
  - Provides BORSCHT functionality (Battery, Overvoltage protection, Ringing, Supervision, Codec, Hybrid and Testing)
  - Provides line and trunk concentration
  - Serves as a "Host" for Remote Offices
  - Serves as an 'SSP' - Connects to SS7 for signaling and AIN functions
- Tandem Central Office
  - Serves as a 'hub' for connecting voice trunks from numerous Class 5 end offices
  - Provides voice trunk connections to Long Distance carriers and Wireless providers
  - Provides E9-1-1 Routing to PSAPs
  - Types include LATA/Access Tandem, Toll Tandem, E911 Tandem, TOPS Tandem

Types of switching entities: STP

- Signaling Transfer Points (STPs)
  - Provide efficient, fast call setup and teardown of telephone calls
  - Provide routing for database lookups (AIN, LNP, 800, etc.)
  - Are the primary switches used in a “packet-based” network as opposed to the circuit based network
  - Provide Gateway Screening for Customer Access (IXCs, ITCs, CLECs, Wireless)
  - Serve as the PSTN entry point into the VoIP Network

Example: BellSouth

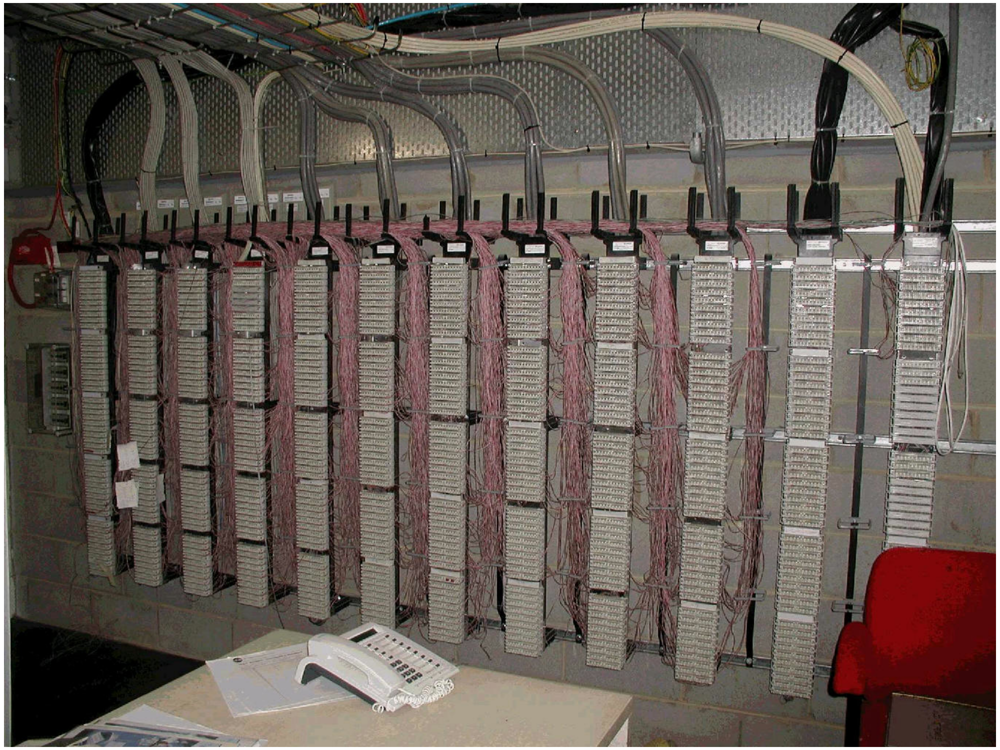
- 32 Analog 1AESS COs (SSPs)
- 856 Lucent 5ESS COs
  - 355 5ESS “Hosts’ and 501 Remotes
- 581 Nortel DMS COs
  - 285 DMS “Hosts” and 283 Remotes and 10 DMS-10
- 138 Siemens COs (includes 85 Remotes)
- 1607 Total COs with approx. 20.3 million NALs
  - hosts ~ 24,000 lines
  - remotes ~ 3,500 lines
- 109 tandems

CO picture

copper wires: home → cable vault → distribution frame

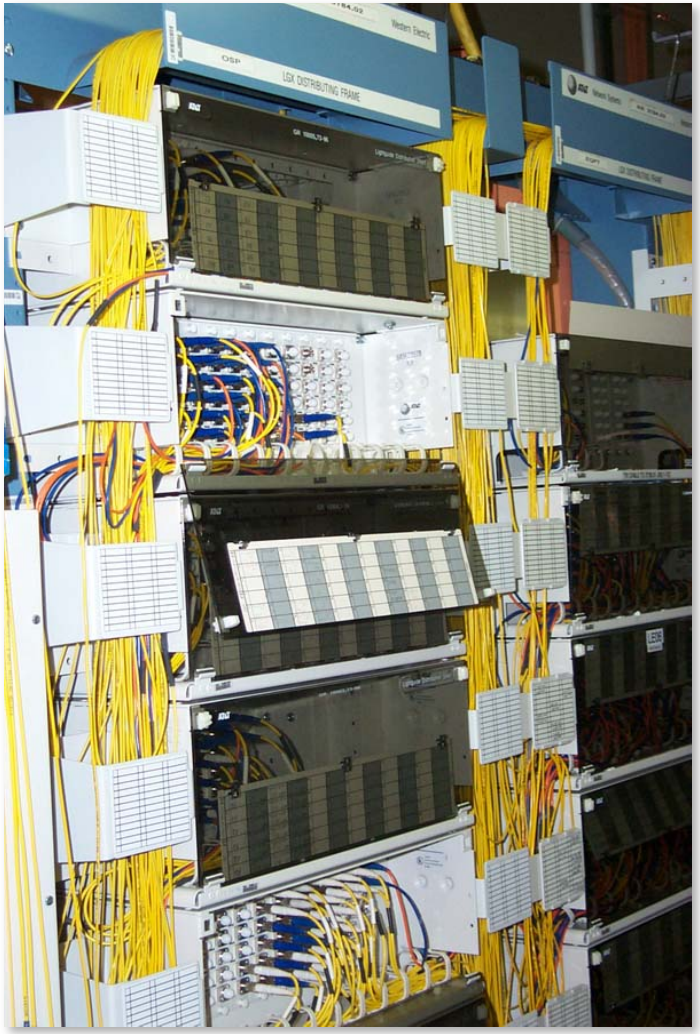
D. Finn (BellSouth 2006)

CO picture



distribution frame

CO pictures



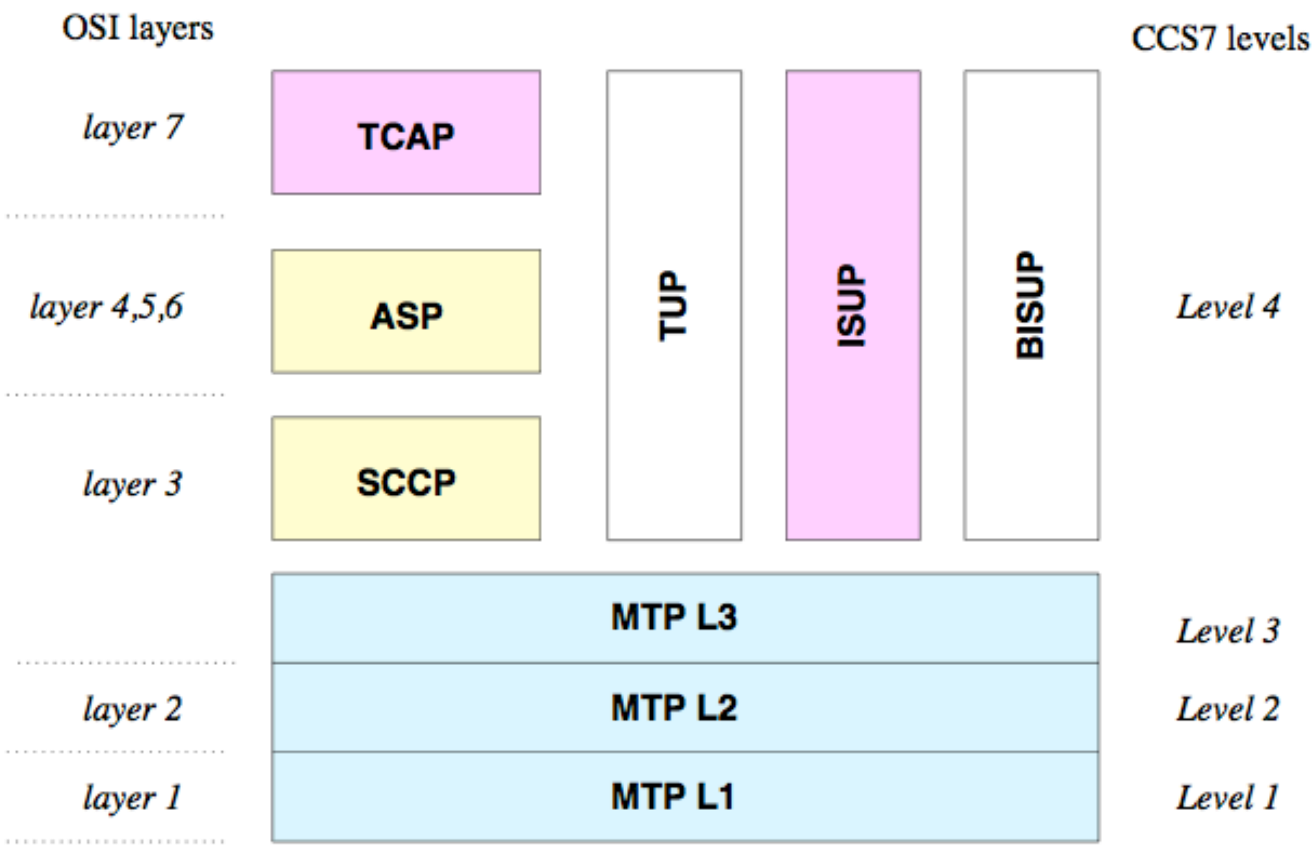
fiber cross connect point: fiber leaves CO

D. Finn (BellSouth 2006)

SS7

- SSP: service switching point = voice switch + adjunct
- STP: signal transfer point router
- SCP: service control point = interface to databases
  - call management service database
  - line information database
  - home location register (cellular)
  - visitor location register (cellular)
- traditionally, connected by 64 kb/s & T1 leased lines
  - future: IP (→ IETF Sigtran WG)

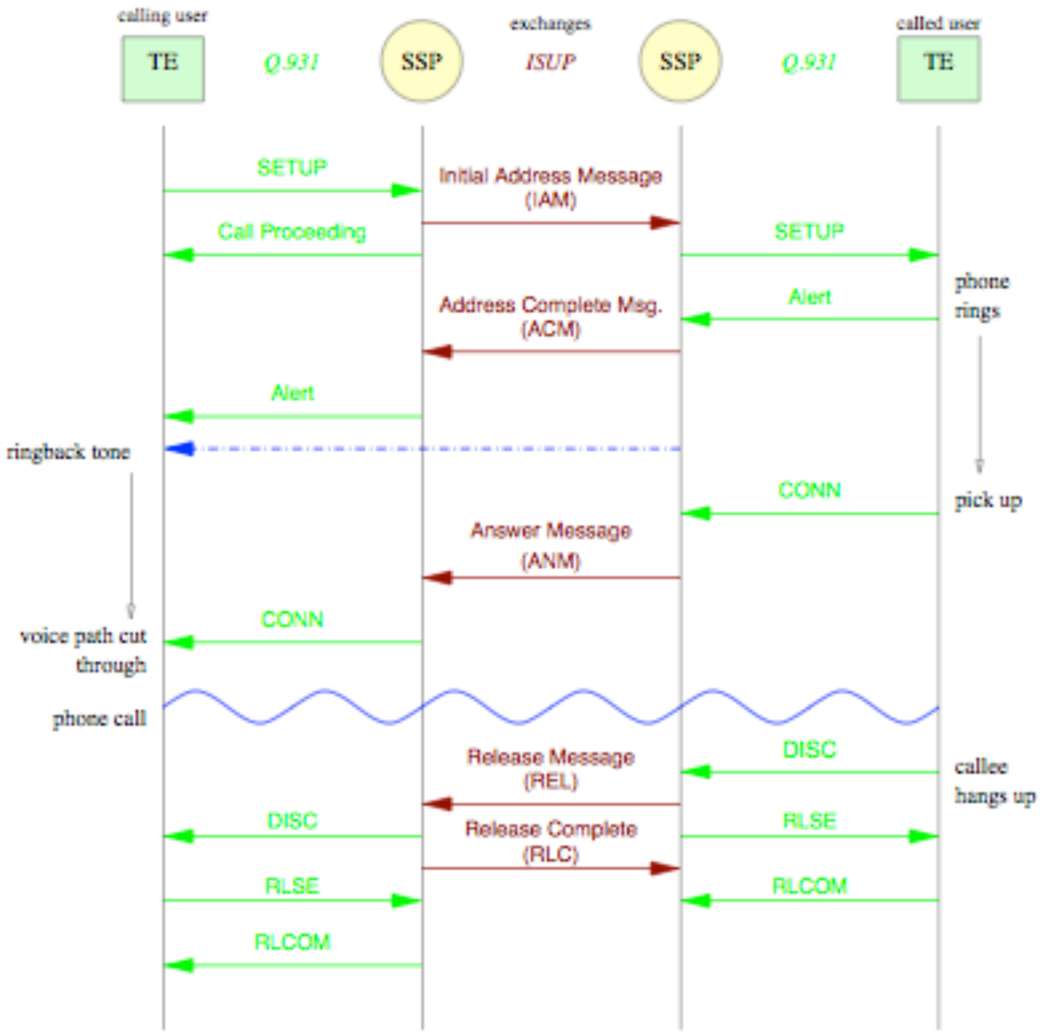
SS7 protocol stack



SS7 protocol stack

- **Level 1** (physical)
  - DS0A = 56/64 kb/s in DS1 facility
- **Level 2** (data link)
  - error detection/correction, link-by-link
- **Level 3** (network)
  - routing message discrimination ➠ "point codes" distribution
- **Level 4** (user parts)
  - basic signaling (ISUP)
  - Transaction Capabilities Application (TCAP)
  - Operations, Maintenance, Administration (OMAP)
  - Mobile Application Part (MAP)

SS7 call



calling user
TE
Q.931
SSP
exchanges
ISUP
SSP
Q.931
called user
TE
SETUP
Initial Address Message
(IAM)
Call Proceeding
SETUP
phone
rings
Alert
Address Complete Msg.
(ACM)
Alert
ringback tone
CONN
pick up
Answer Message
(ANM)
CONN
voice path cut
through
phone call
DISC
Release Message
(REL)
callee
hangs up
DISC
Release Complete
(RLC)
RLSE
RLSE
RLCOM
RLCOM

Reliability

#9's	reliability	outage/year	example
1	90%	36.5 days	
2	99%	3.65 days	
3	99.9%	8.8 hours	good ISP
4	99.99%	53 minutes	
5	99.999%	5 minutes	phone system
6	99.9999%	32 seconds	

Reliability

- FCC incidents: ≥ 90,000 customers, > 30 minutes (972 between 1992 and 1997)
- FCC ARMIS (Automated Reporting Management Information System)
- ANSI T1A1: logarithmic outage index = f(duration, # affected, time, functions, ...)
- call defects per million (e.g., AT&T 173 ppm)

Service Quality:

http://fjallfoss.fcc.gov/eafs7/PresetMenu.cfm

Info Average Installation Intervals in Days

Info Percent Local Installation Commitments Met

Info Out of Service Repair Intervals in Hours

Info Repeat Out-of-Service Trouble Reports as a Percentage of Initial Out-of-Service Trouble Reports

Info State Complaints per 1,000,000 Lines

Info Total Trouble Reports per Month per 100 Lines

Outages

- median outage lasts 2.9 hours
  - (natural disasters: 13.4 hours)
  - causes:
    - facilities (45%)
    - local switches (18%), CCS (13%), CO power (7.3%)
  - facility failures:
    - dig-ups ("back-hoe fade", 58%)
    - cable electronics (8%)
- ARMIS example:
  - Bell Atlantic 1998: 180 switches, combined downtime of 628 minutes, or $6.6 \cdot 10^{-6}$

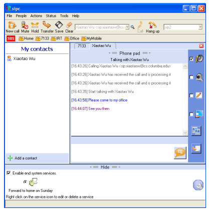
The phone works – why bother with VoIP

user perspective	carrier perspective
variable compression: tin can to broadcast quality → no need for dedicated lines	better codecs + silence suppression – packet header overhead = maybe reduced bandwidth
security through encryption	shared facilities simplify management, redundancy
caller & talker identification	advanced services
better user interface (more than 12 keys, visual feedback, semantic rather than stimulus)	cheaper bit switching
no local access fees (but dropping to 1c/min for PSTN)	fax as data rather than voiceband data (14.4 kb/s)
adding video, application sharing is easy	

Old vs. new

	old reality	new idea	new reality
service provider	**ILEC**, *CLEC*	email-like, run by enterprise, homes	E.164-driven; MSOs, some ILECs, Skype, European SIP providers, Vonage, ~~SunRocket~~
media	4 kHz audio	wideband audio, video, IM, shared apps, …	4 kHz audio
services	CLASS (CLID, call forwarding, 3-way calling, ...)	user-created services (web model) presence	still CLASS
user IDs	E.164	email-like	E.164 IM handles

Evolution of VoIP



"Can it really replace the phone system?"

long-distance calling, ca. 1930

"does it do call transfer?"

"How can I make it stop ringing?"

replacing the global phone system

going beyond the black phone

"amazing – the phone rings"

catching up with the digital PBX

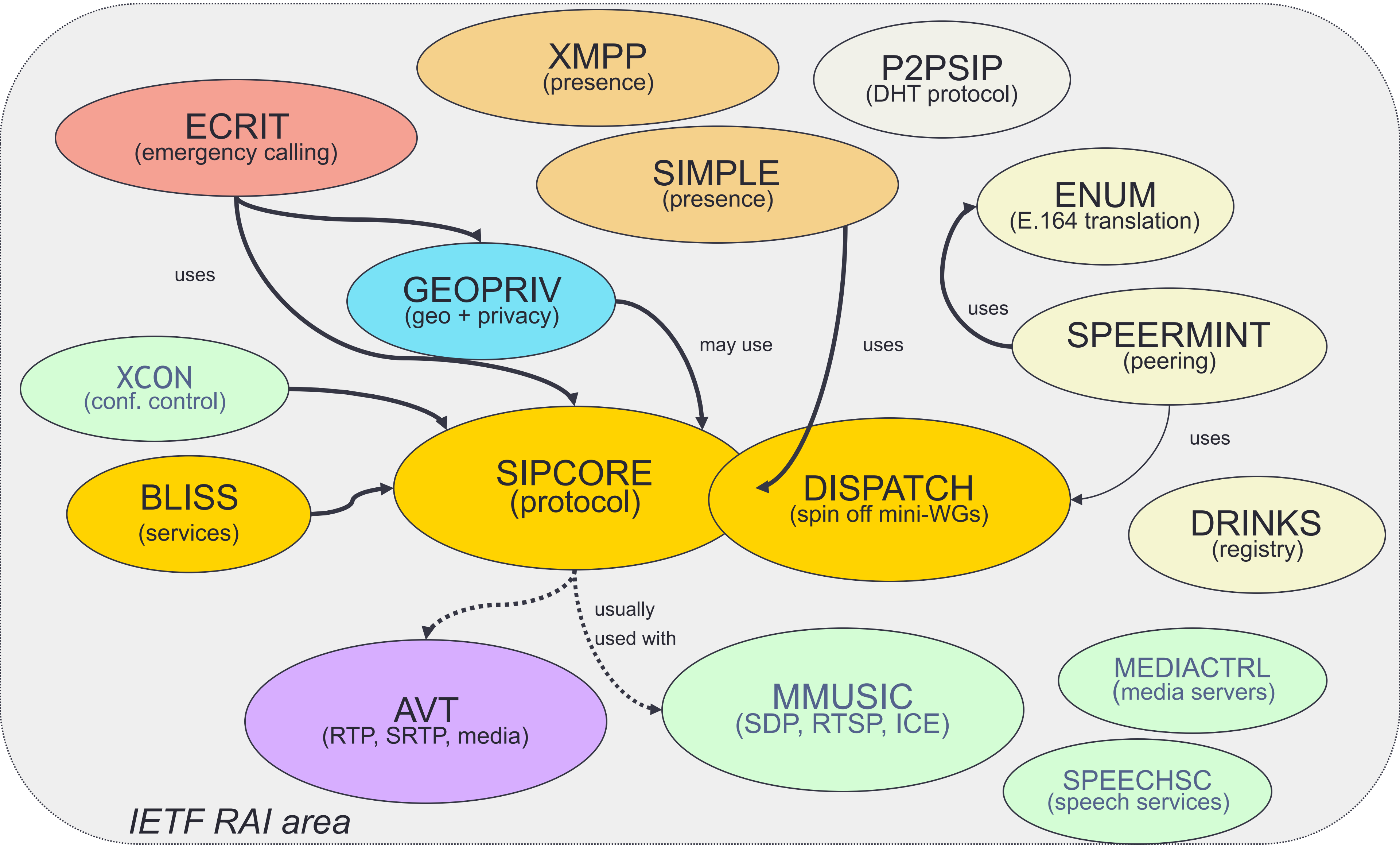
1996-2000 2000-2003 2004-2005 2006-

VoIP Signaling Protocols

- H.323
  - ITU standard, ISDN-based, distributed topology
  - early on, used to be 90%+ of all Service Provider VoIP networks
  - video conferencing (Microsoft NetMeeting, room units [Polycom, Tandberg, …])
- Skinny
  - Centralized call control architecture
  - CallManager controls all features
  - over 1 mio. IP Phones deployed – probably most popular corporate IP-PBX
- MGCP
  - IETF RFC 2705
  - *Centralized* call control architecture
  - Call-Agents (MGC) & Gateways (MG)
- SIP
  - IETF RFC 2543 and RFC 3261
  - *Distributed* call control
  - Used for more than VoIP…SIMPLE: Instant Messaging / Presence

Brian Gracely, Cisco, 2001 (mod.)

IETF VoIP & presence efforts



PBX features

centrex-style features

call waiting/multiple calls	RFC 3261
hold	RFC 3264
transfer	RFC 3515/Replaces
conference	RFC 3261/callee caps
message waiting	message summary package
call forward	RFC 3261
call park	RFC 3515/Replaces
call pickup	Replaces
do not disturb	RFC 3261
call coverage	RFC 3261

boss/admin features

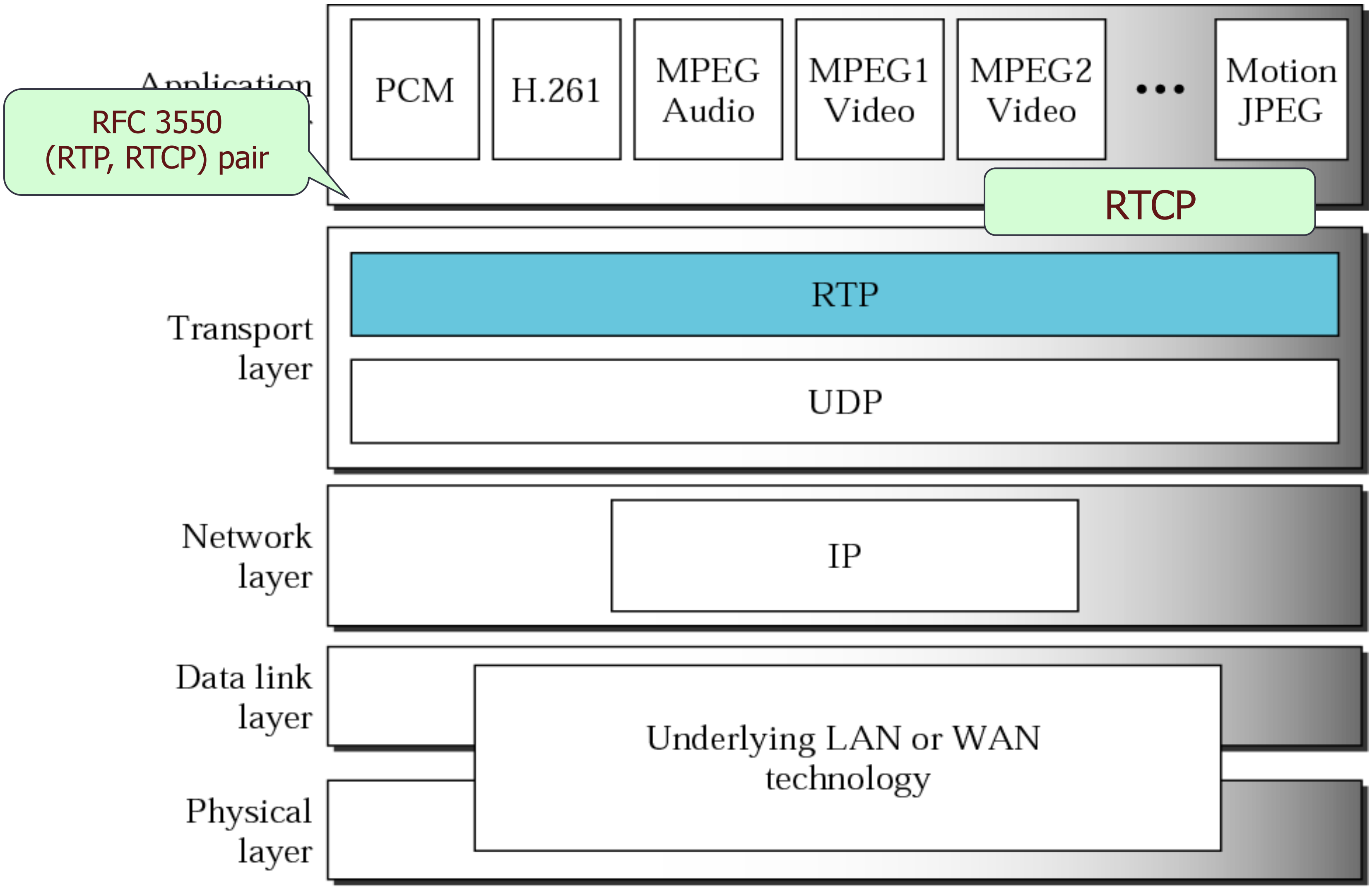
simultaneous ringing	RFC 3261
basic shared lines	dialog/reg. package
barge-in	Join
"Take"	Replaces
Shared-line "privacy"	dialog package
divert to admin	RFC 3261
intercom	URI convention
auto attendant	RFC 3261/2833
attendant console	dialog package
night service	RFC 3261

attendant features

from Rohan Mahy's VON Fall 2003 talk

RTP

RTP stack



Application
RFC 3550
(RTP, RTCP) pair
PCM
H.261
MPEG
Audio
MPEG1
Video
MPEG2
Video
• • •
Motion
JPEG
RTCP
Transport
layer
RTP
UDP
Network
layer
IP
Data link
layer
Physical
layer
Underlying LAN or WAN
technology

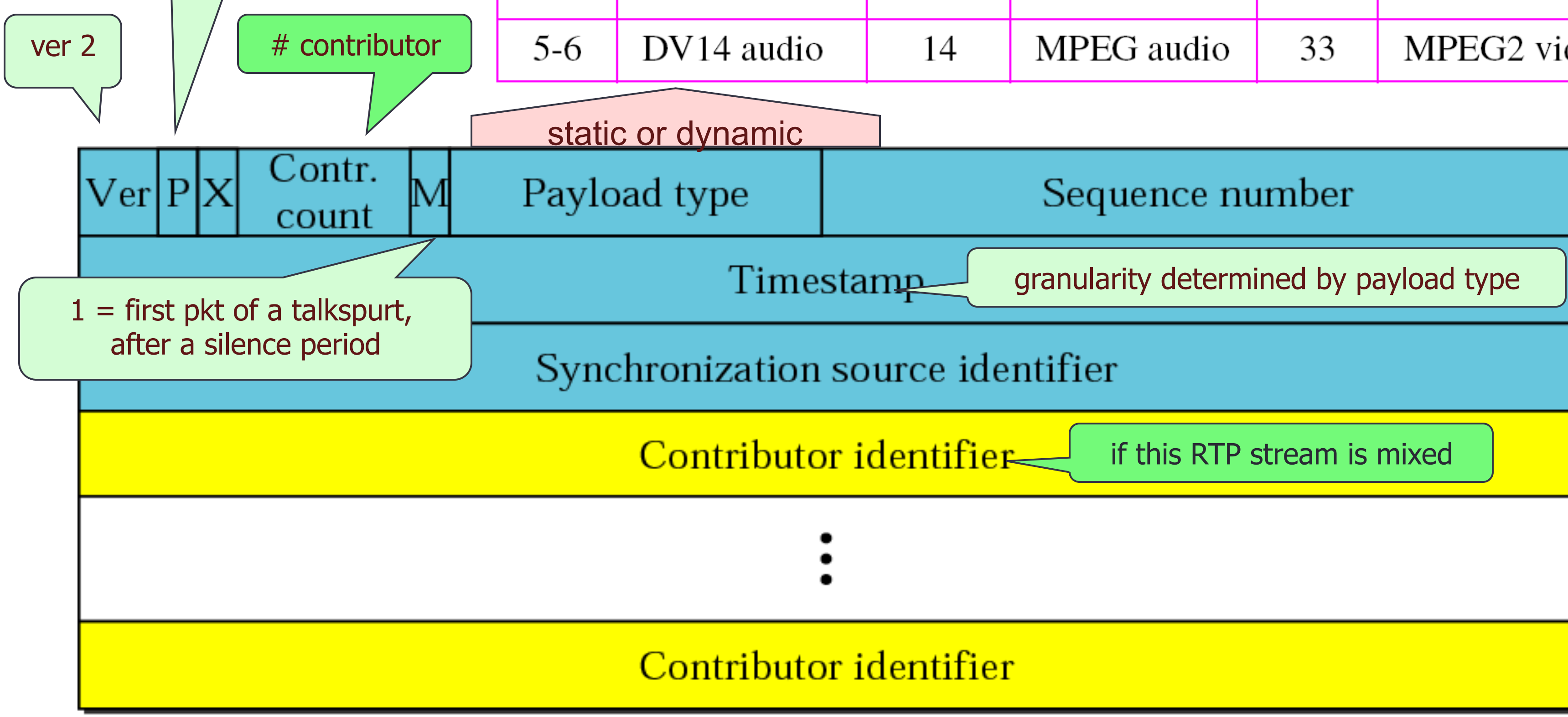
RTP

- Real-Time Transport Protocol (RTP) = data + control
  - data (media):
    - timing
    - loss detection
    - content labeling
    - talkspurts & video frames
    - encryption
  - control (RTCP):
    - ➠ periodic with T ~ population
    - QoS feedback
    - membership estimation in multicast
    - loop detection

RTP Packet Header

RFC 3551
audio-video profile

Type	Application	Type	Application	Type	Application
0	PCMμ Audio	7	LPC audio	15	G728 audio
1	1016	8	PCMA audio	26	Motion JPEG
2	G721 audio	9	G722 audio	31	H.261
3	GSM audio	10-11	L16 audio	32	MPEG1 video
5-6	DV14 audio	14	MPEG audio	33	MPEG2 video



RTP timestamp

- +1 per sample (e.g., 160 for 20 ms packets @ 8000 Hz)
- random starting value
- time per packet may vary
- different fixed rate for each audio PT
  - typically, 20 – 100 ms / packet
- 90 kHz for video
  - several video frames may have same timestamp
- ➠ gaps ≡ silence
  - split video frame (carefully. . .) across packets