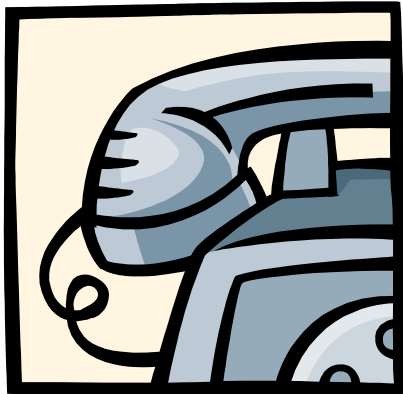




Voice Over IP: Architectures, Applications and Challenges



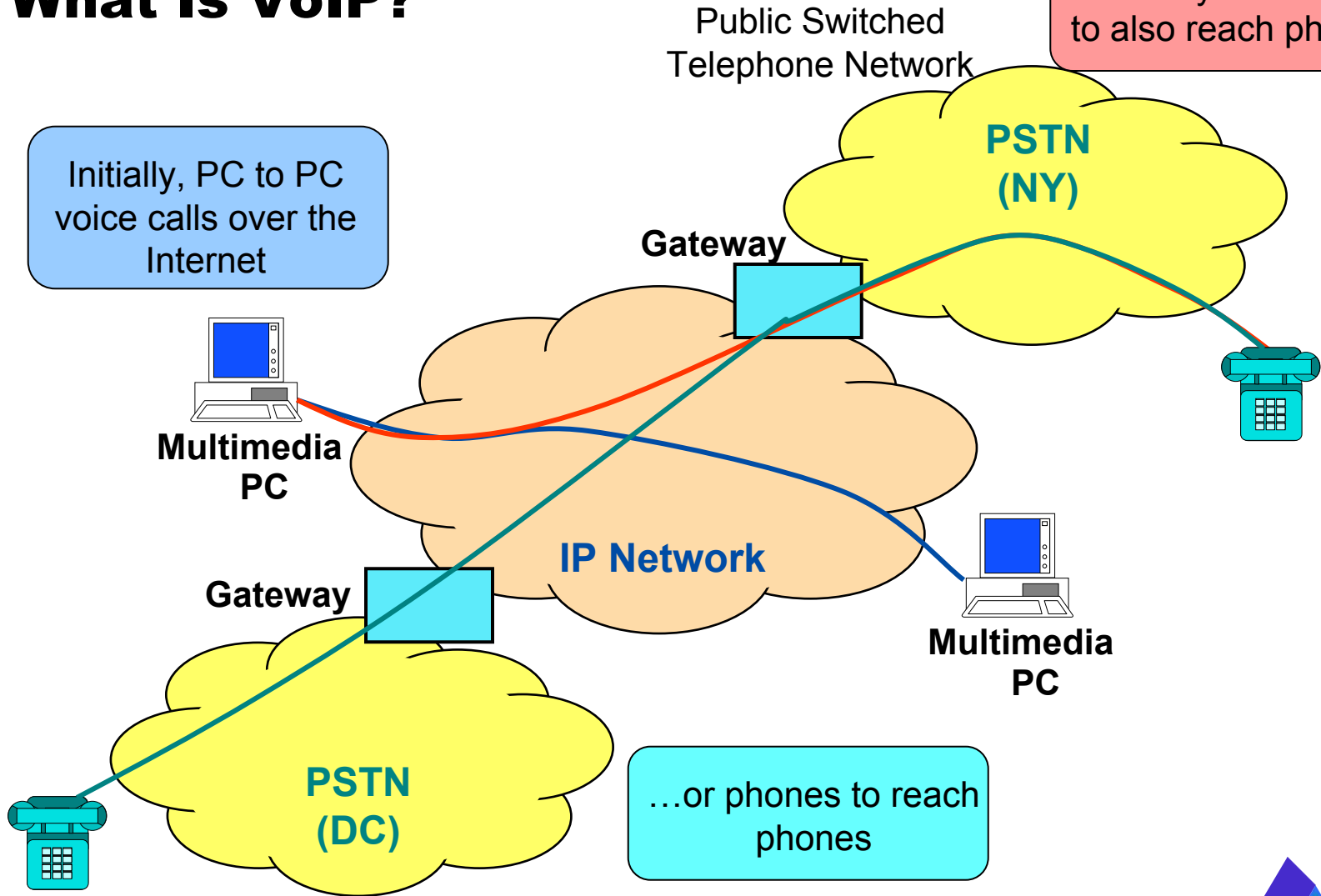
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April 15, 2002

What Is VoIP?

Gateways allow PCs to also reach phones

Initially, PC to PC voice calls over the Internet



...or phones to reach phones



Outline

- Why Voice over IP?
- Packet voice transport
- Signaling and control architectures
- Network applications
- Outlook and challenges



Origins of VoIP

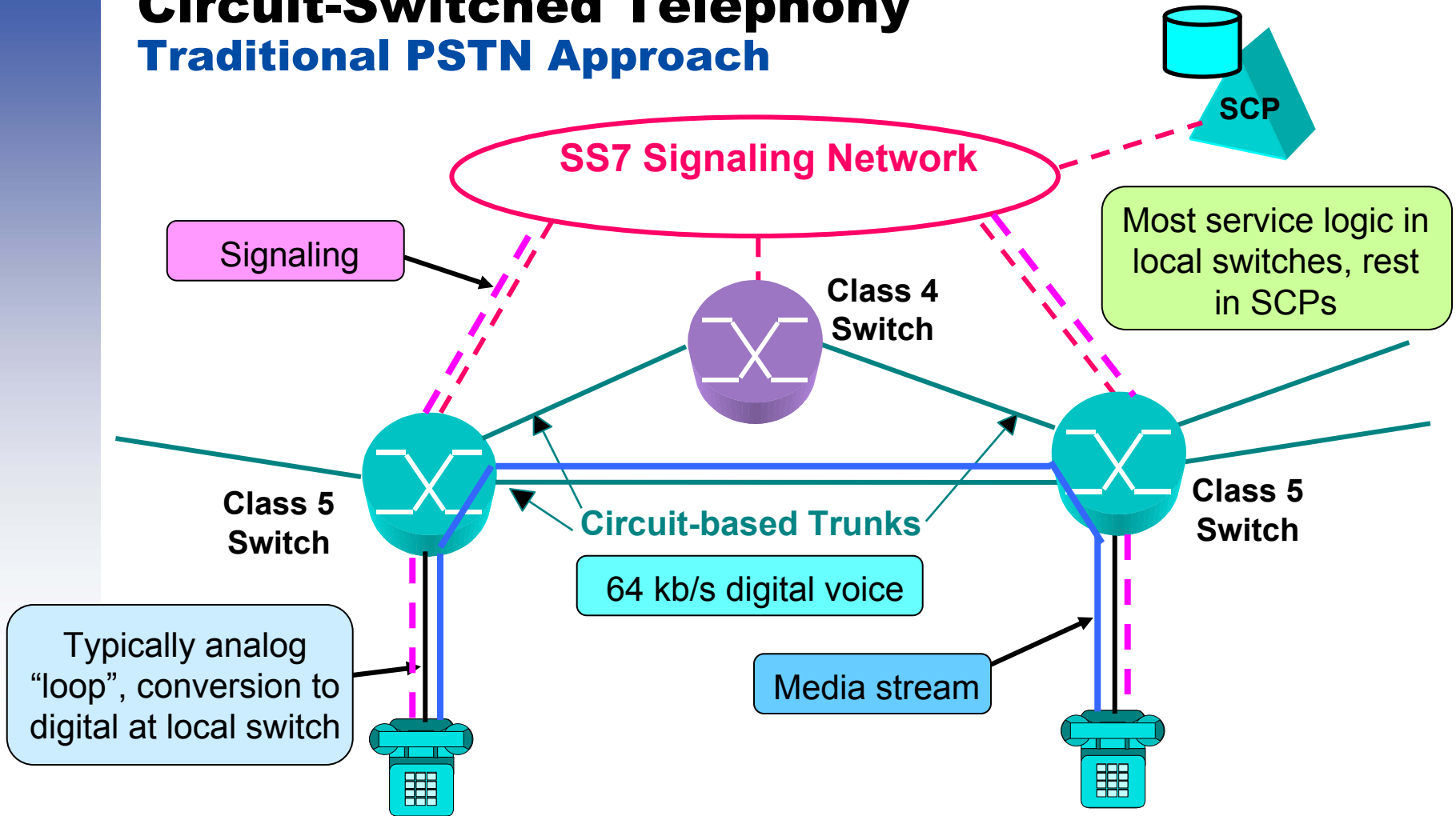
Lessons learned

- Internet Telephony software for multimedia PCs (1995)
 - Users frustrated by poor QoS, difficulty of use, lack of interoperability
- Standards are critical for success
 - Coding/decoding (codec) between analog voice and digital packets
 - Locating the party you want to call
 - Signaling to set up, modify, tear down the voice call
 - Access to vertical services (call forwarding, 3-way calling, ...)
 - Gateways to PSTN
- Media routing, quality of service (QoS) left to other IP mechanisms (not VoIP-specific)



Circuit-Switched Telephony

Traditional PSTN Approach



- Data travels over a parallel (but separate) network

VoIP

Goals and Potential Benefits

- Consolidation of voice, data on a single network
 - Simplify infrastructure, operations; provide bundled services
- Support for intelligent terminals as well as phones
- Increased flexibility
 - Multiple bit rates, multiple media types, richer signaling
 - Distinguish calls from connections (add/modify streams during call)
- Separation of service control from switching/routing
 - Accelerate new service development, increase end-user control, evolve from VoIP towards advanced services
- Expansion of competition



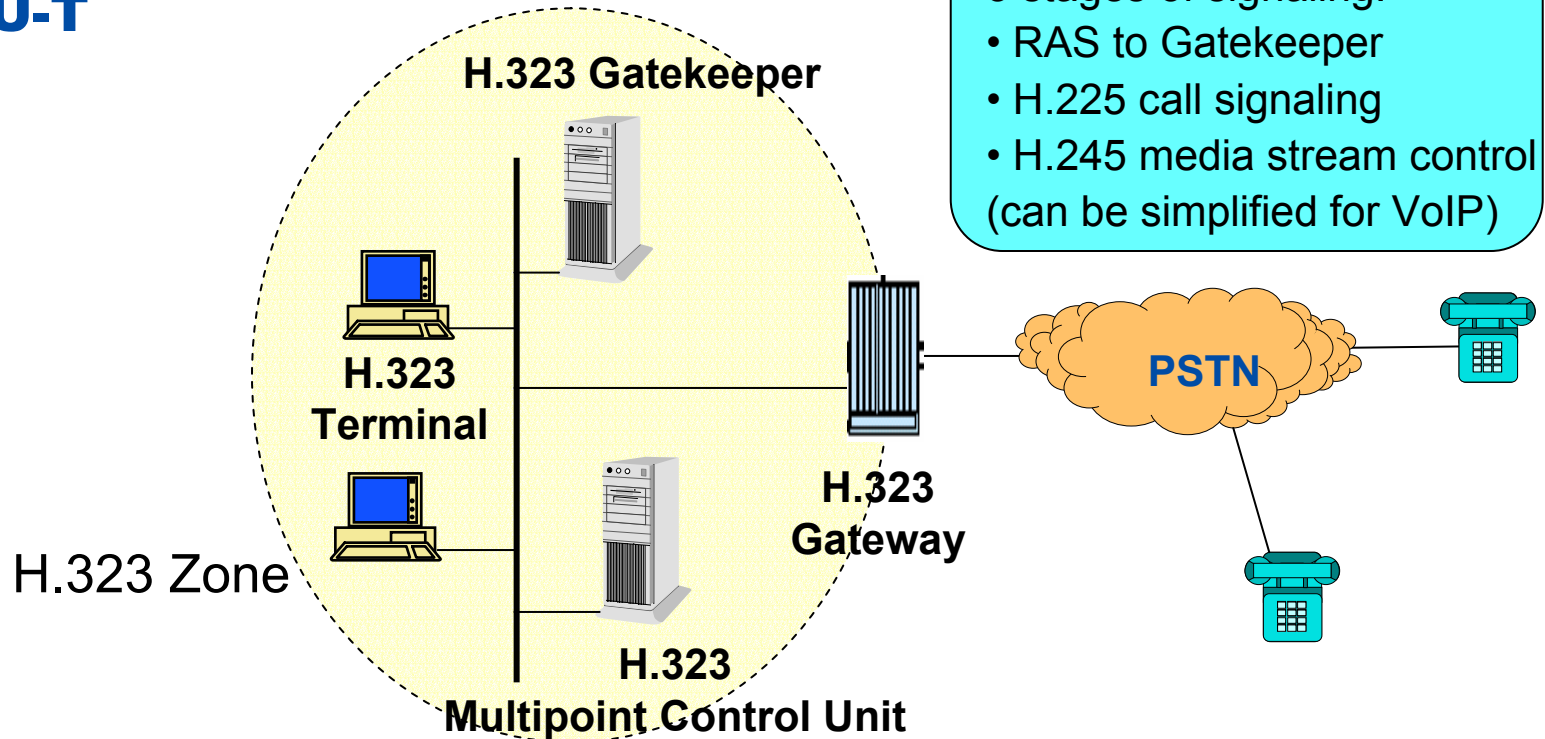
Packet Voice Transport

- Key targets for voice call service quality:
 - Average packet loss: < 2%
 - Consecutive packet loss: < 200 ms burst
 - End-to-end (lip-to-ear) delay: < 150 ms for comfortable conversation
- Packet loss cannot be corrected by retransmission (TCP), because the packets arrive too late to be useful
- Use RTP (Real-time Transport Protocol) over UDP (User Datagram Protocol) for voice or video transport
 - Payload ID, sequence numbers, timestamps, monitoring via RTCP
- Packet and buffer lengths limited by constraint on end-to-end delay
- Typical codecs: G.711 (64 kb/s), G.729 (8 kb/s) G.723 (~ 6 kb/s)
 - Transmitted bit rates depend on overheads, optional silence suppression



H.323 Architecture

ITU-T



- Telco-centric multimedia, multiparty conferencing (initially for LANs)
- Gatekeeper for network control, heavy-weight protocols
- Widely deployed in first wave of VoIP standardization

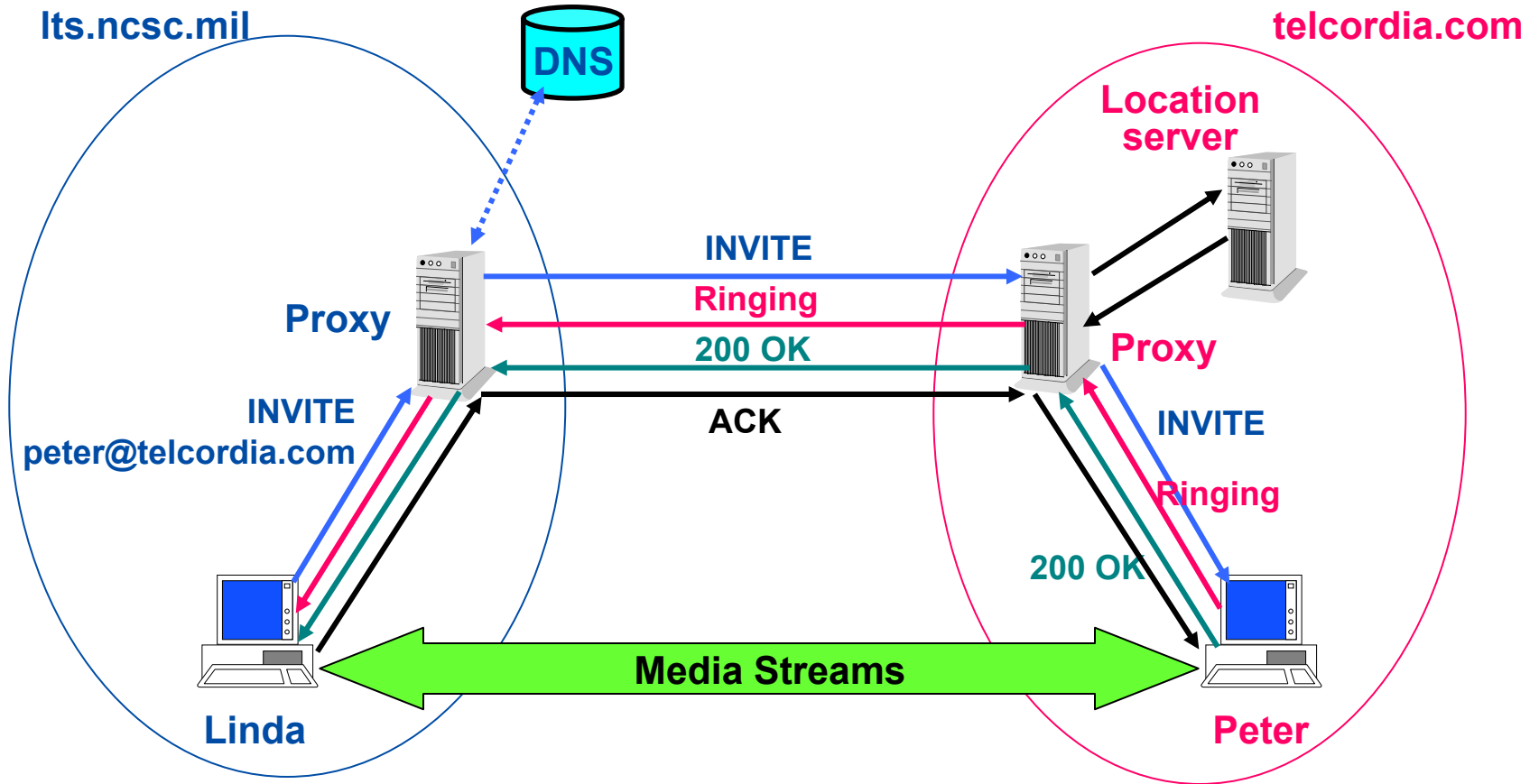
SIP (Session Initiation Protocol)

IETF Multimedia Architecture

- Internet-centric alternative, initially for large multicast conferences
 - SIP for call signaling, SDP (Session Description Protocol) for media
- Initially very simple, light-weight, loosely-coupled sessions; oriented towards direct signaling between endpoints
- Network servers for additional capabilities:
 - **Registrar** for terminal registration, aliases
 - **Redirect** returns contact address directly to end user
 - **Proxy** forwards signaling (requests, responses)
- Evolution towards greater use of **proxy/registrar** for locating users, vertical services, call tracking, network control
- Strong, rapidly growing support (e.g., Microsoft XP, 3GPP)



SIP Call Setup Simplified View



INVITE SDP proposes media type(s), IP & ports to send to
200 OK SDP accepts/rejects media, gives IP & ports to send to

Where Do Services Live?

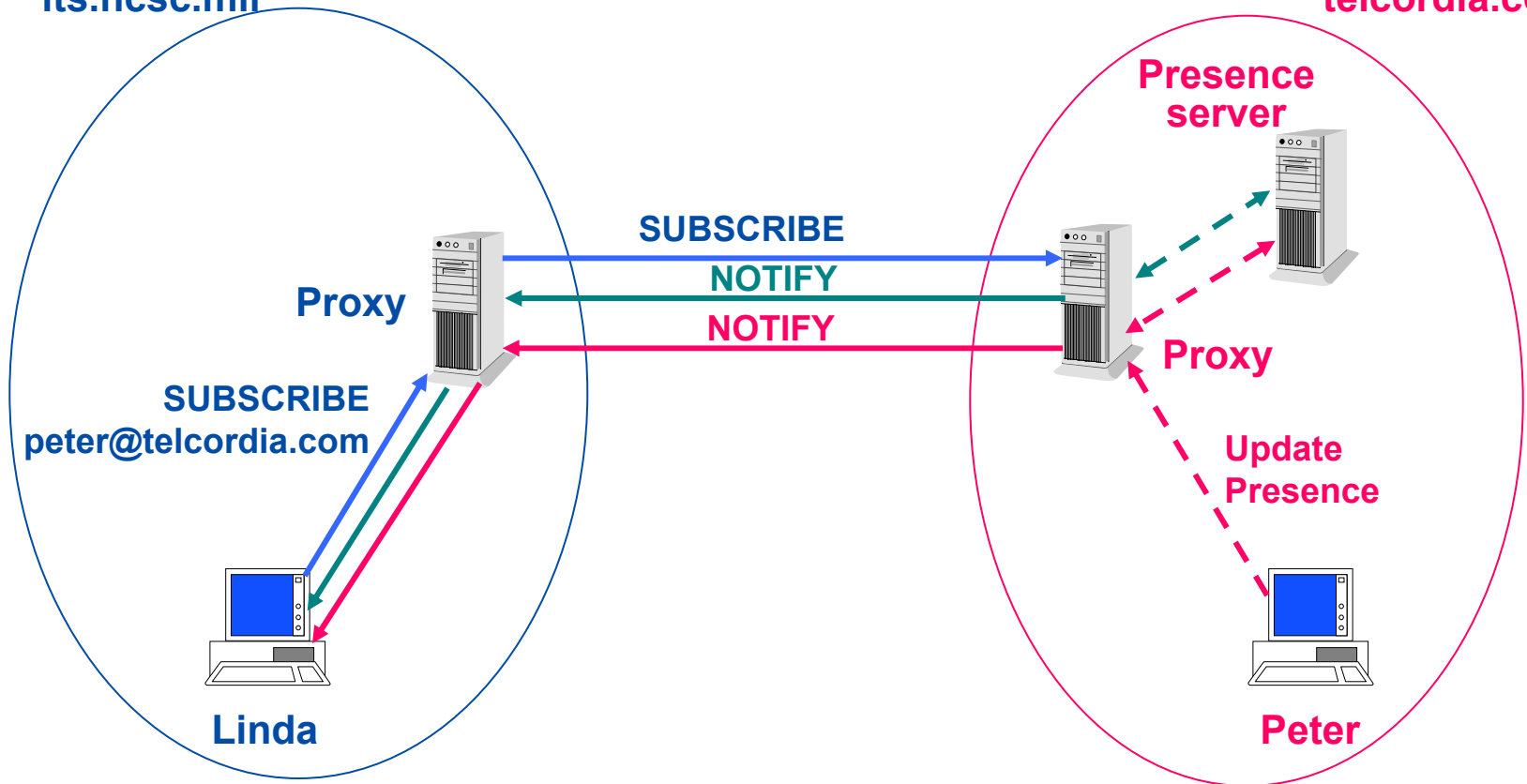
- Some implemented at the endpoints
 - Last-number redial, call hold...
- Others may be better supported from the network
 - Avoid need for PC or IP phone to be turned on (call forwarding)
 - More complex services, such as conferencing
 - Integration with web-based services (unified messaging)
- Example: SIP Proxy runs a script for each incoming call for Peter
 - Parallel forking: forward INVITE to multiple endpoints simultaneously
 - Sequential forking: try his office PC first, then lab, then cell phone, ...



SIMPLE (SIP for IM and Presence) Simplified Example

Its.ncsc.mil

telcordia.com



Linda subscribes to notifications of changes in Peter's status:
Off-line, on-line, busy, away, available, ...

NGN Architecture

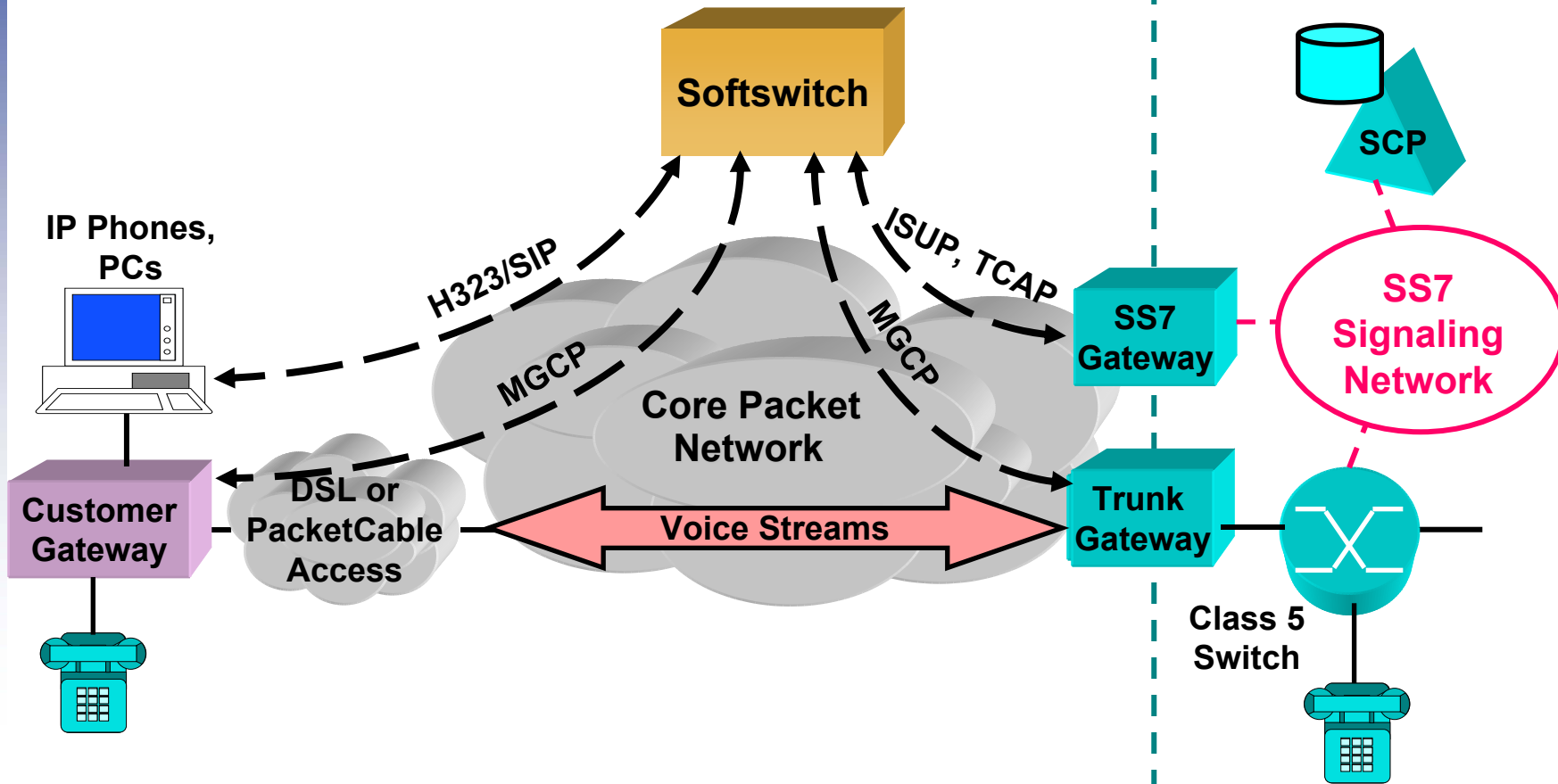
Next-Generation Network

- Oriented towards application of VoIP (or VoATM) to large-scale public networks
- Focus on scalability, network control, support for traditional phones, sophisticated gateway (GW) to the PSTN and its services
- Media GW interfaces voice stream to PSTN trunk or phone line
- Signaling GW allows signaling directly to SS7 network
- Softswitch controls Media GWs and does call processing
 - Allows smaller, cheaper Media GWs (e.g., for individual homes)
 - Control via MGCP (Media Gateway Control Protocol) or H.248



NGN Example

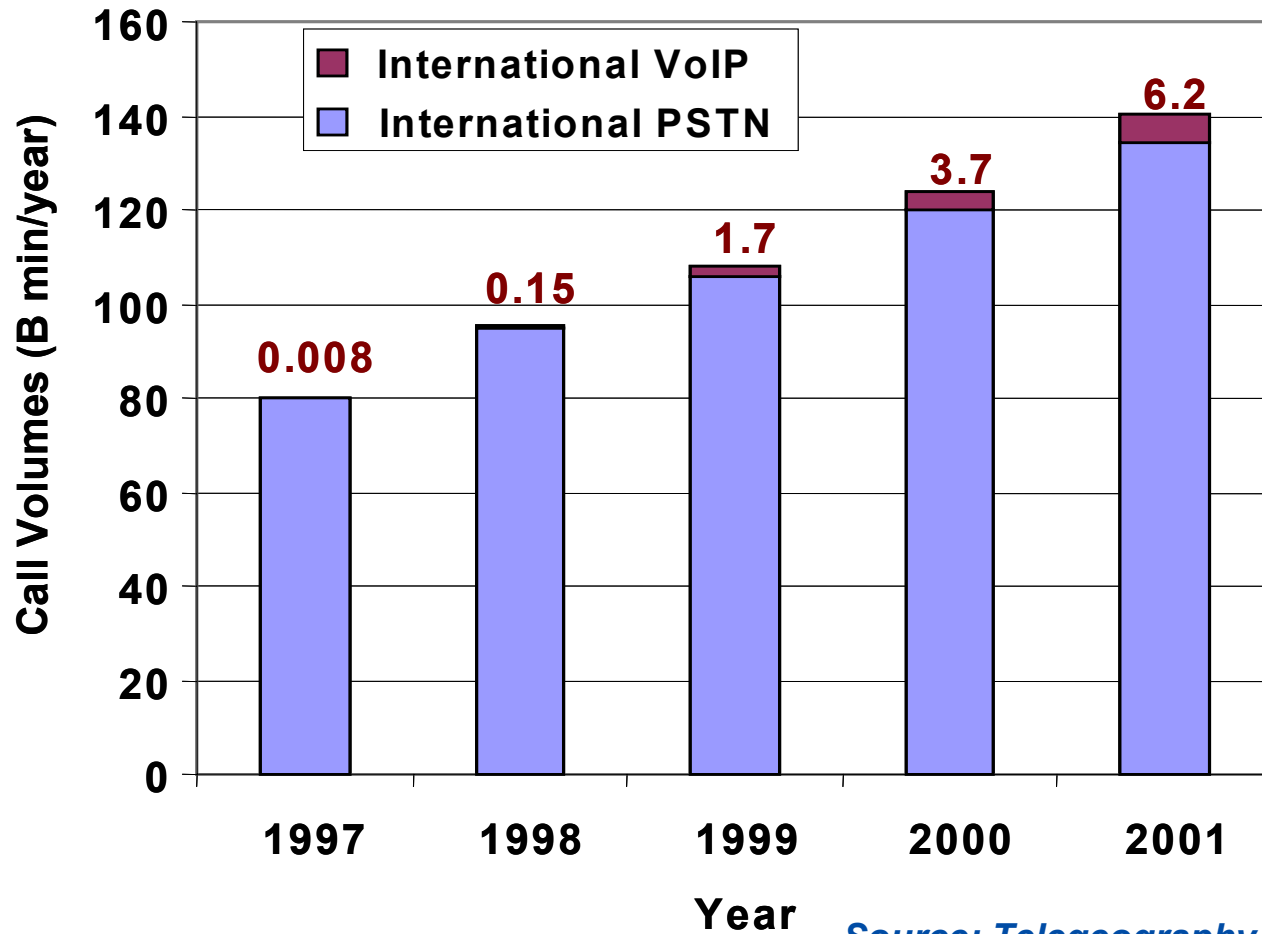
Voice over DSL or Cable Modem



Can also use to interconnect PSTN clouds (long-distance), or PSTN switches (interoffice backbone)

International Voice Market

Calls Terminated on PSTN



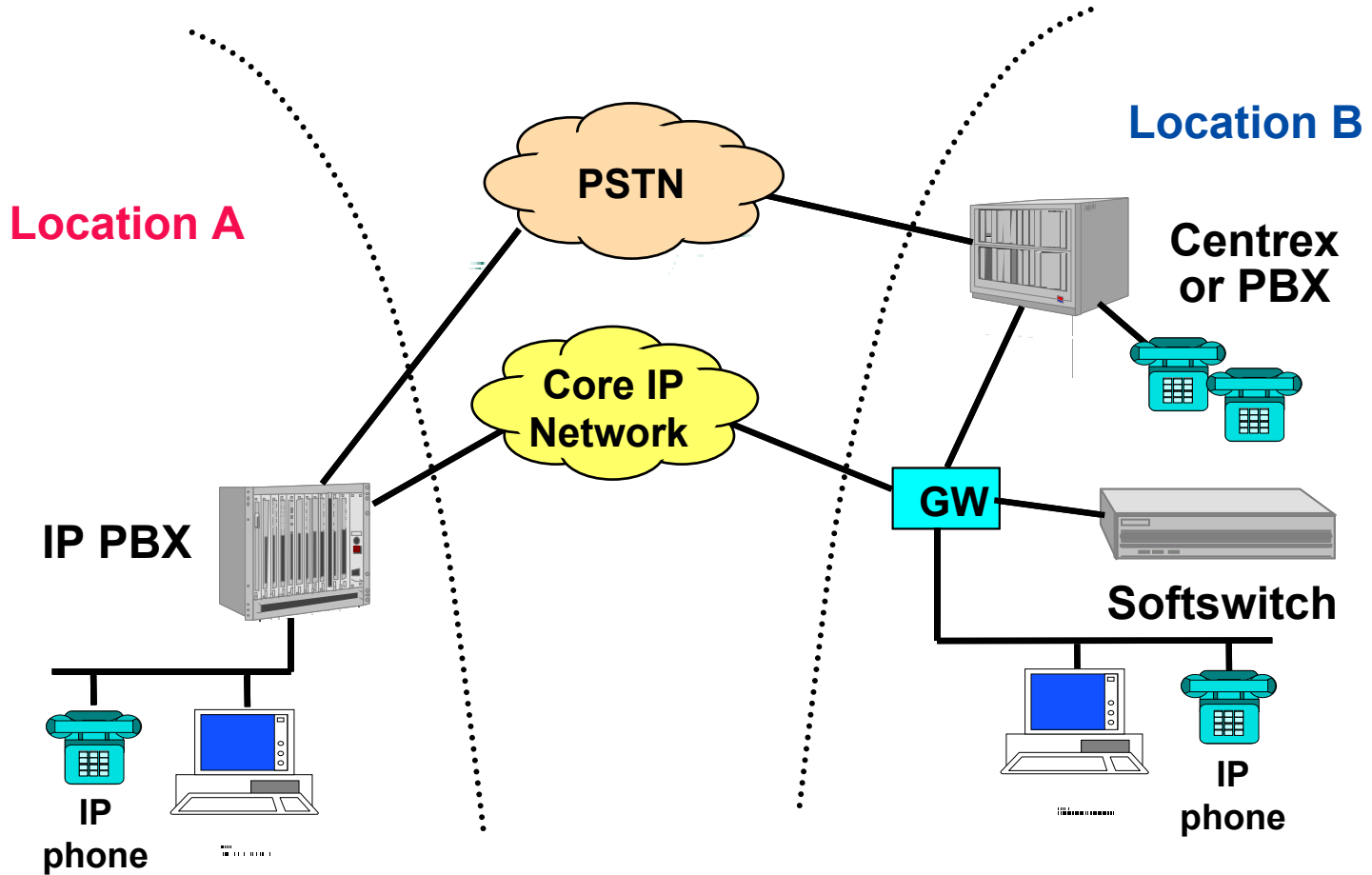
Source: Telegeography 2001
(2001 figures were projections)

Carrier Applications of VoIP

- First major inroads for VoIP have been in long-distance
 - Avoid regulation, high international PSTN tariffs
 - VoIP invisible to end user, doesn't rely on him to do anything
 - Installed base dominantly H.323, movement now towards NGN
- Local-carrier interest for interoffice connections
 - Consolidate voice and data networks (typically ATM)
 - Use NGN, or packet-enable existing switches
- Many trials of VoIP to residences, but deployments few
 - Cable TV has laid groundwork for NGN approach (DOCSIS 1.1)
 - Decline of CLECs likely to slow multi-line VoDSL



Enterprise VoIP



Many possible combinations of VoIP and circuit-switched telephony

Enterprise Applications of VoIP

- Leverage spare data-network capacity, minimize phone bills, create platform for multimedia conferencing
- H.323 and SIP both being deployed, softswitches and IP-PBX options emerging, unclear which will prevail
- Examples: Telcordia/SAIC (H.323), Telia (SIP)
- Carrier-managed VPN networks last year from AT&T (H.323) and Worldcom (SIP)
- VoIP adoption slower than expected, partly due to plunging PSTN long-distance prices, QoS concerns



Peer-to-Peer VoIP

PC-to-PC

- Internet Telephony revisited, often facilitated by software or network servers from new types of voice service providers
 - Microsoft, Net2Phone, Dialpad, AOL, Yahoo!
 - Mass market alternative to telcos, requiring limited network infrastructure, capital costs, operating expenses
- What's the business case for “free” VoIP?
 - Sell advertising, software, or enhanced services
 - Charge for PC-to-phone, phone-to-phone
 - Give away as a competitive differentiator
- Mostly H.323 today, likely to move towards SIP
- Could be key industry driver, even if penetration were limited



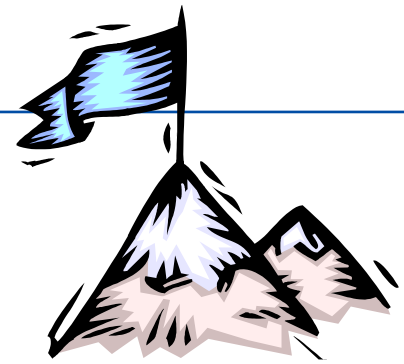
Outlook for VoIP

Current Status and Trends

- VoIP is not monolithic – many applications, with different drivers, will maintain a heterogeneous mix of technologies
- H.323 is most widely implemented today, but trends are towards SIP for intelligent terminals, NGN for most carrier networks
- Most success thus far in long-distance networks, perhaps with local carrier backbones to follow in next few years
- Footholds made in enterprise and access markets, but VoIP has not taken off as fast as initially expected
- Adoption being slowed by economic conditions, plummeting long distance rates, declining advertising market (peer-to-peer)



Continuing Challenges



- Quality of Service
 - Diffserv, MPLS, traffic engineering, bandwidth brokers, call admission...
 - What is really needed for consolidated voice and data networks?
- Security, reliability
- Extending SIP to provide conference control
- Operations (configuration of IP phones, version control and upgrading of highly distributed software, accounting/billing,...)
- Packet-level interconnection of VoIP islands which use competing architectures and protocols
- Controlling feature interactions in a distributed-services environment
- *Traversal of NATs and firewalls*
- *Support for services beyond voice*



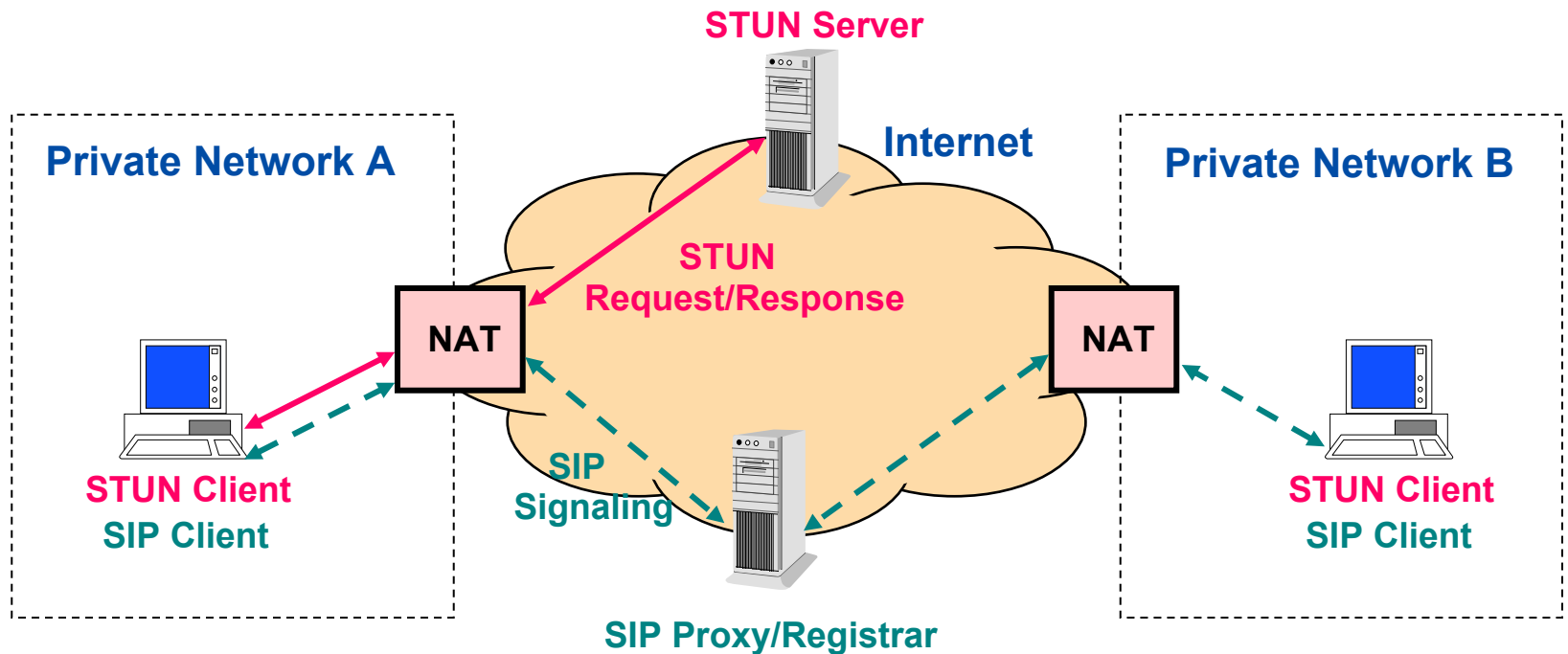
NAT Traversal

- Network Address Translators (NATs) map a private IP address space to externally visible (public) IP addresses
 - Conserve scarce public IP addresses
 - Shield internal hosts from outside world
- Useful for enterprises, cable modem networks, broadband access routers, internet cafes...
- NATs interfere with peer-to-peer protocols such as SIP
 - SIP clients must identify the IP address and ports they will use to receive media streams (in payload of their signaling messages)
 - But they don't know their externally visible addresses
- “One of the SIP community's biggest problems”



STUN – Simple Traversal of UDP Through NATs

draft-rosenberg-midcom-stun-01.txt



Source: P. Thermos, Telcordia

- STUN client contacts STUN server, discovers NAT, address translation
- SIP client uses “external” address in signaling for setup of media streams
- This approach being implemented and tested at Columbia and LTS

Advanced Services

- VoIP: natural platform for evolution to advanced services
 - Supports intelligent terminals and rich signaling
 - Separates calls from connections
 - Multimedia capabilities already in the protocols (SIP/H.323)
 - Removes bottleneck by separating call control from switching
- Thus far, focus is almost entirely on voice
 - For many players (but not all), voice *is* the killer app
 - Solve the simpler problem first
- This simplifies many network control issues, because of predictability of voice bandwidth, traffic patterns
 - But current solutions are likely to require significant extensions to accommodate more flexible advanced services



Moving Beyond Two-Party Voice

What's Different About Advanced Services?

- Flexibility in media streams, participants, “ownership”; service not pre-defined at call setup
 - Multiple media per call, differing (and very wide range of) bandwidths
 - Dynamic reconfigurability during call
 - Potential for multicast conferencing, streaming
- Implications
 - Call admission control becomes more complex
 - Much less aggregation, localization of flows than with NGN voice
 - Usage, traffic patterns may be highly variable and hard to predict
- New approaches to traffic engineering, resource allocation and network control will be needed to address even a modest penetration of these new services



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