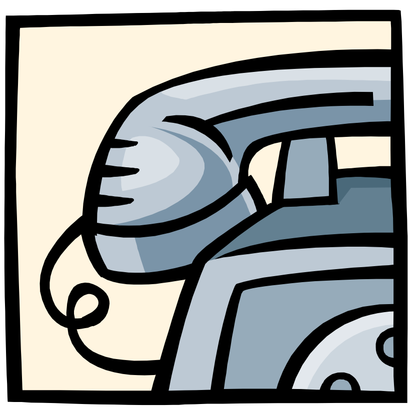
Voice Over IP: Architectures, Applications and Challenges



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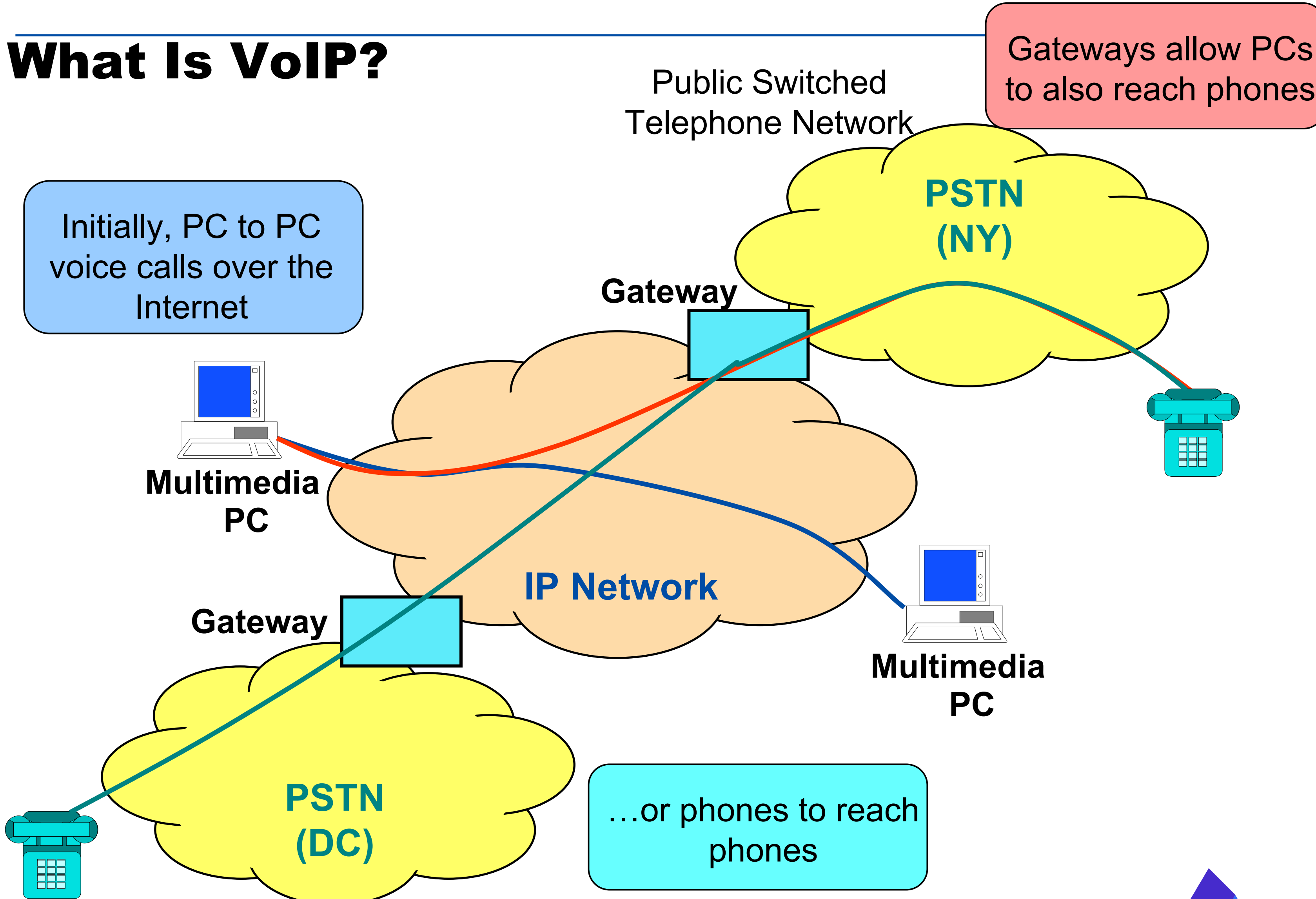
What Is VoIP?



Public Switched
Telephone Network
Gateways allow PCs to also reach phones
Initially, PC to PC voice calls over the Internet
PSTN (NY)
Gateway
Multimedia PC
IP Network
Gateway
Multimedia PC
PSTN (DC)
…or phones to reach phones

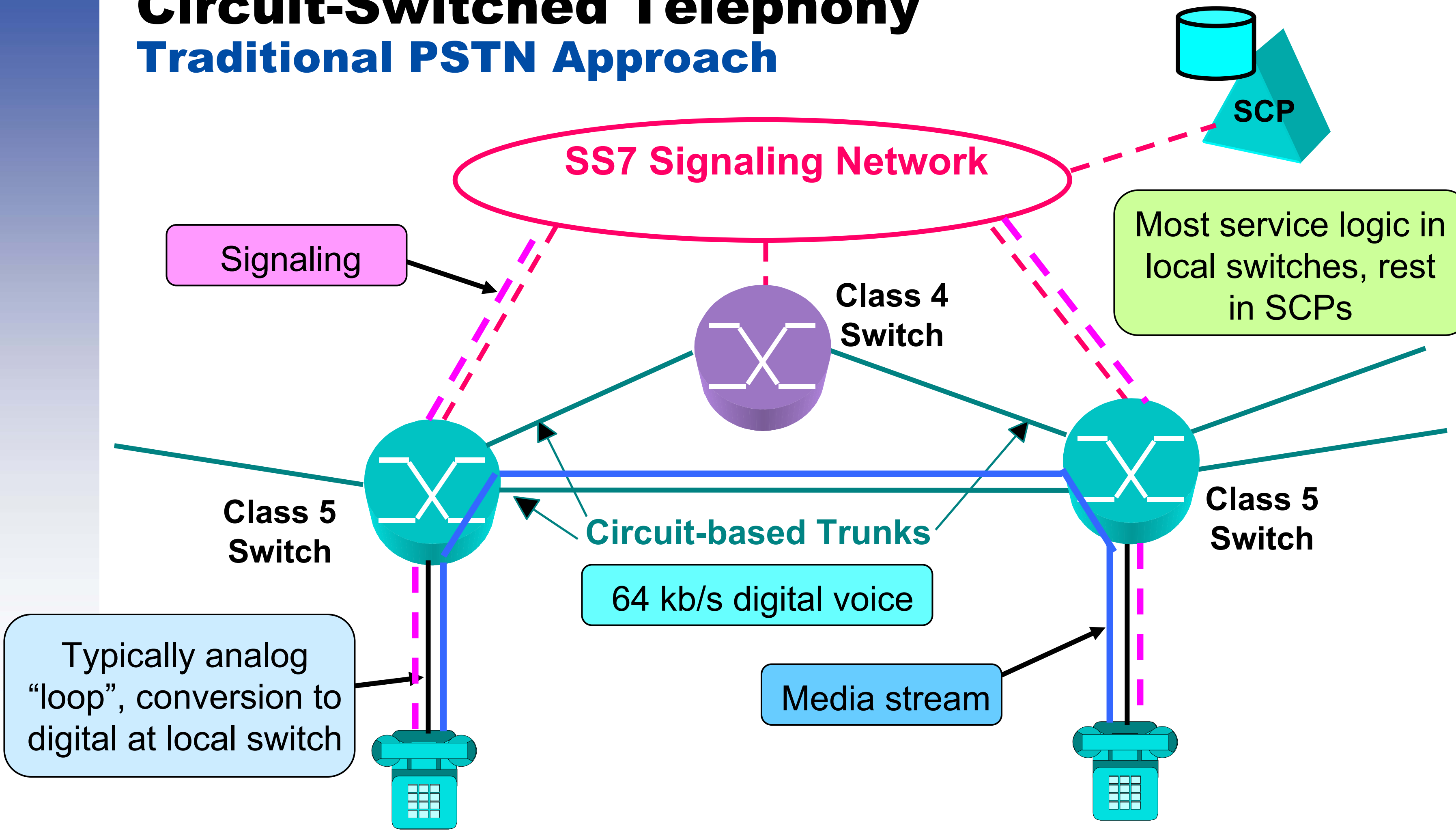
Outline

- 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

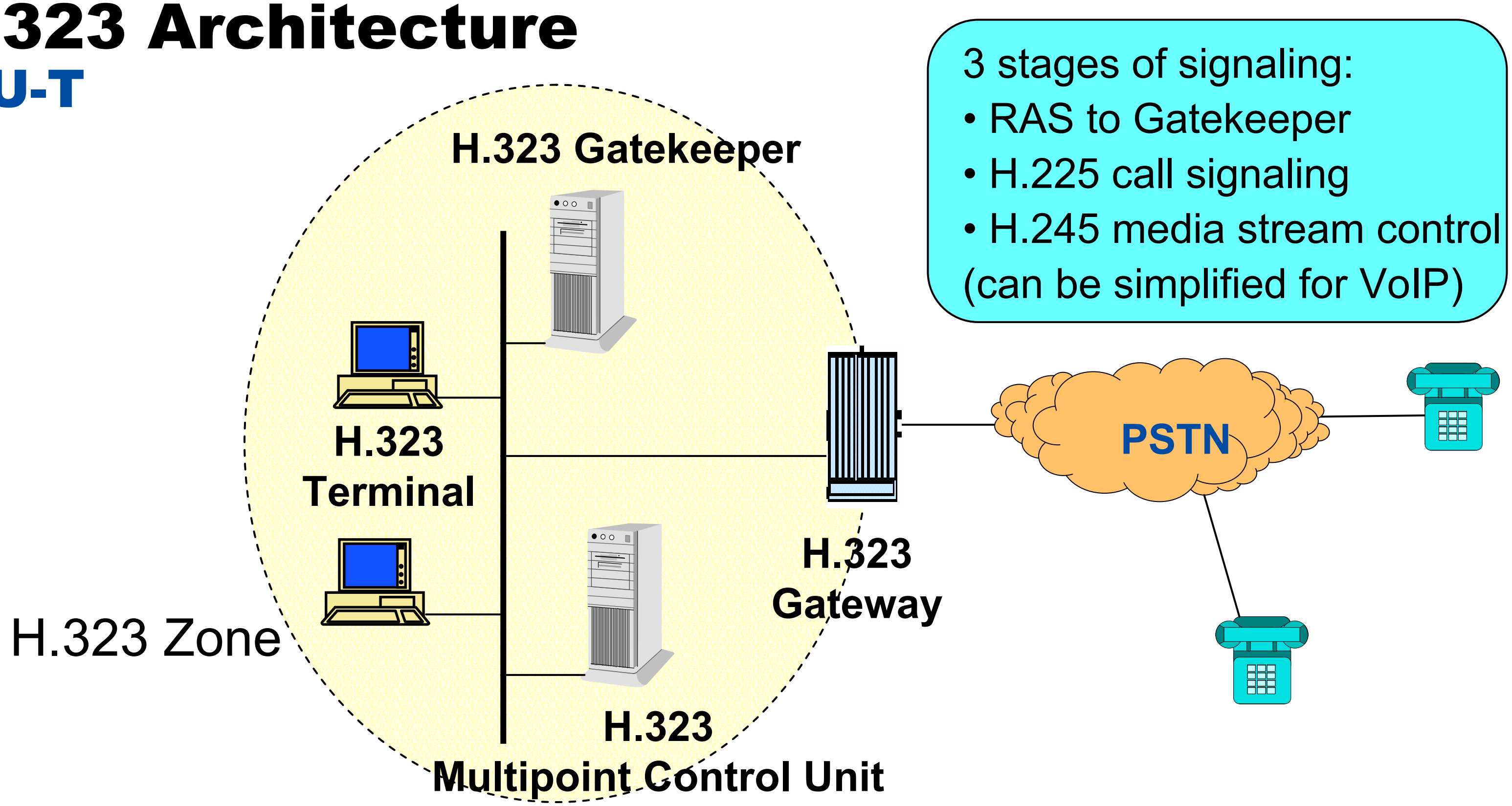
- Consolidation of voice, data on a single network
  - Simplify infrastructure, operations; provide bundled services
- Support for intelligent terminals as well as phones
- Increased flexibility
  - Multiple bit rates, multiple media types, richer signaling
  - Distinguish calls from connections (add/modify 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

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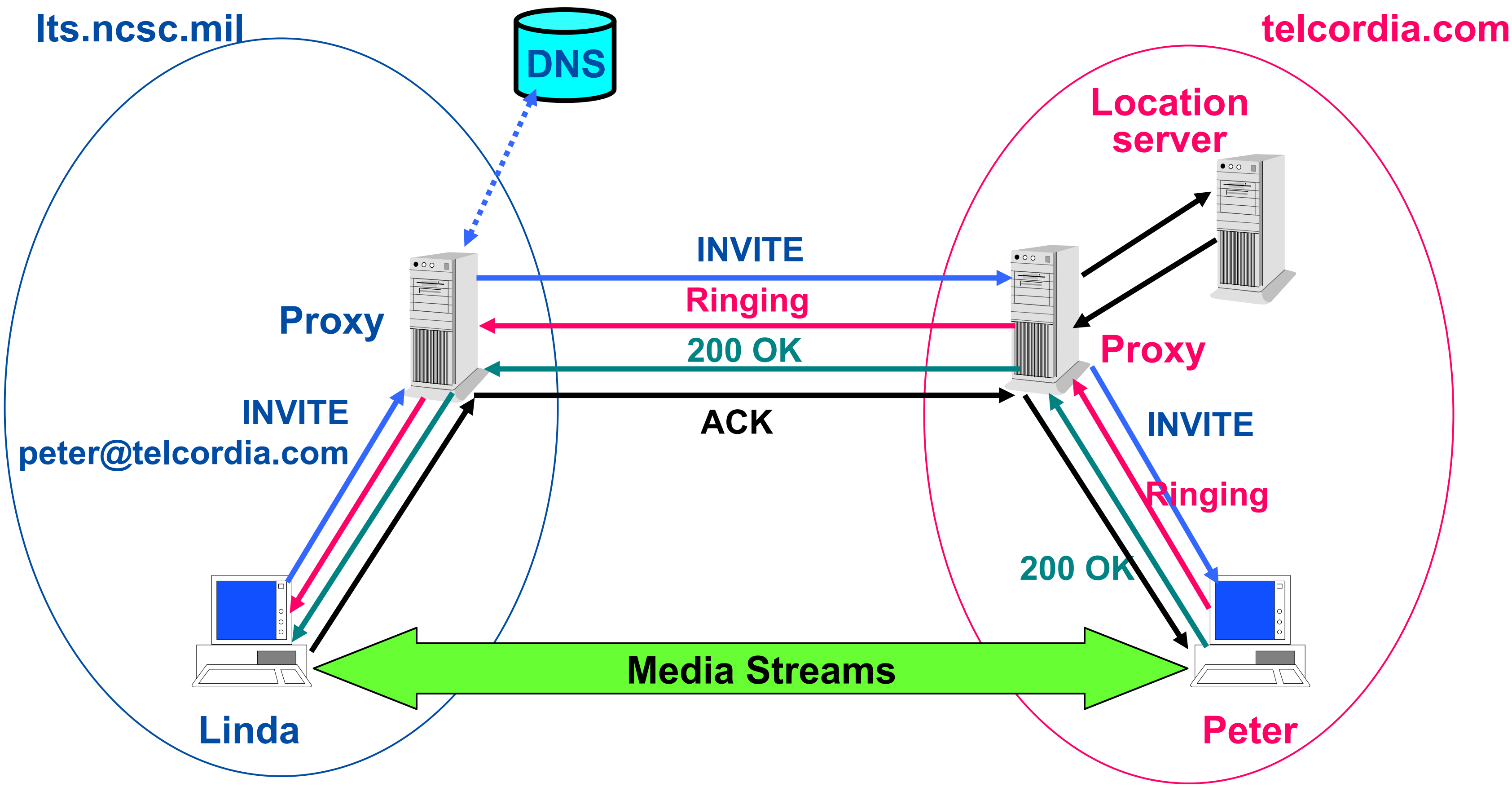
SIP (Session Initiation Protocol)
IETF Multimedia Architecture

- 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

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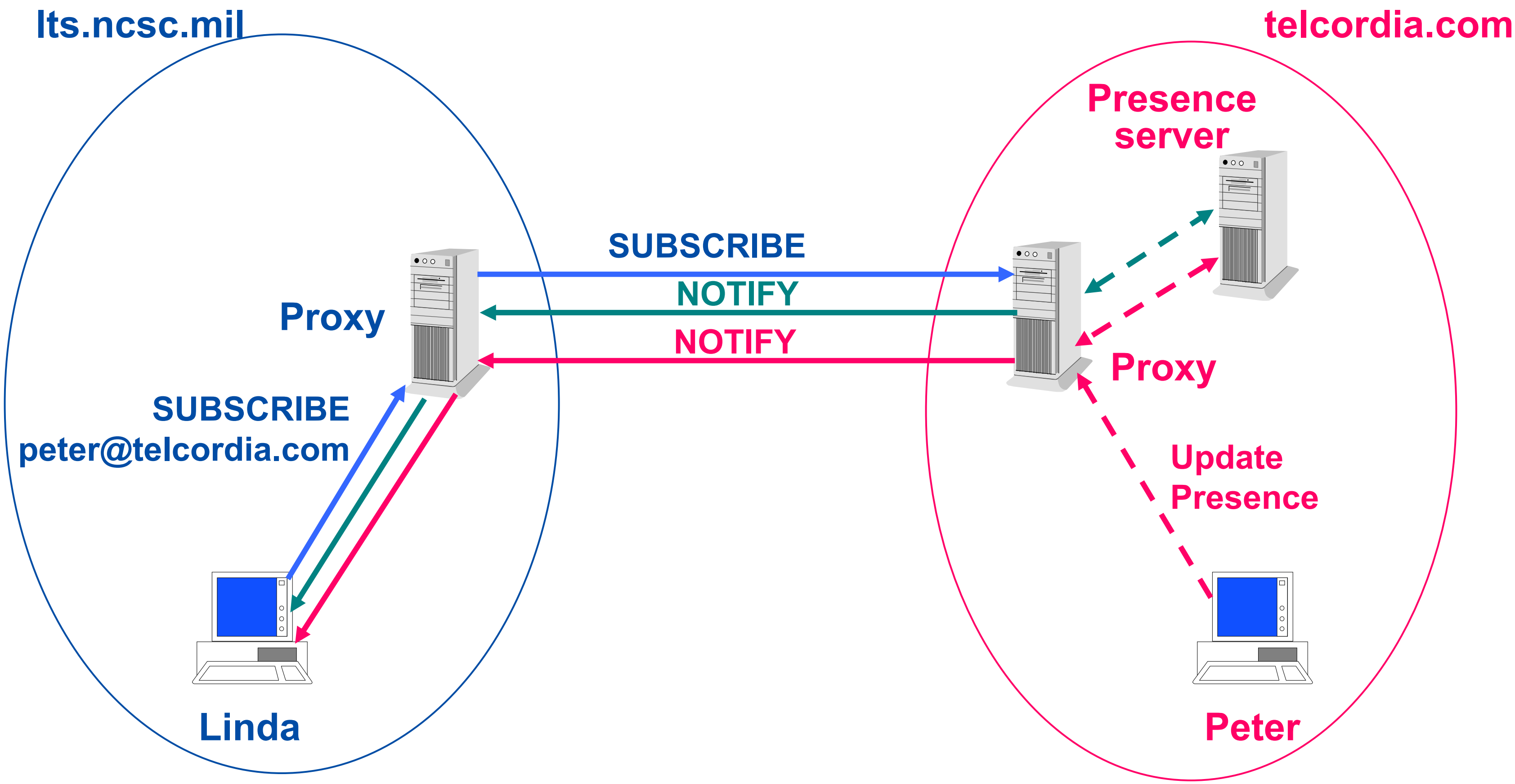
Where Do Services Live?

- 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



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

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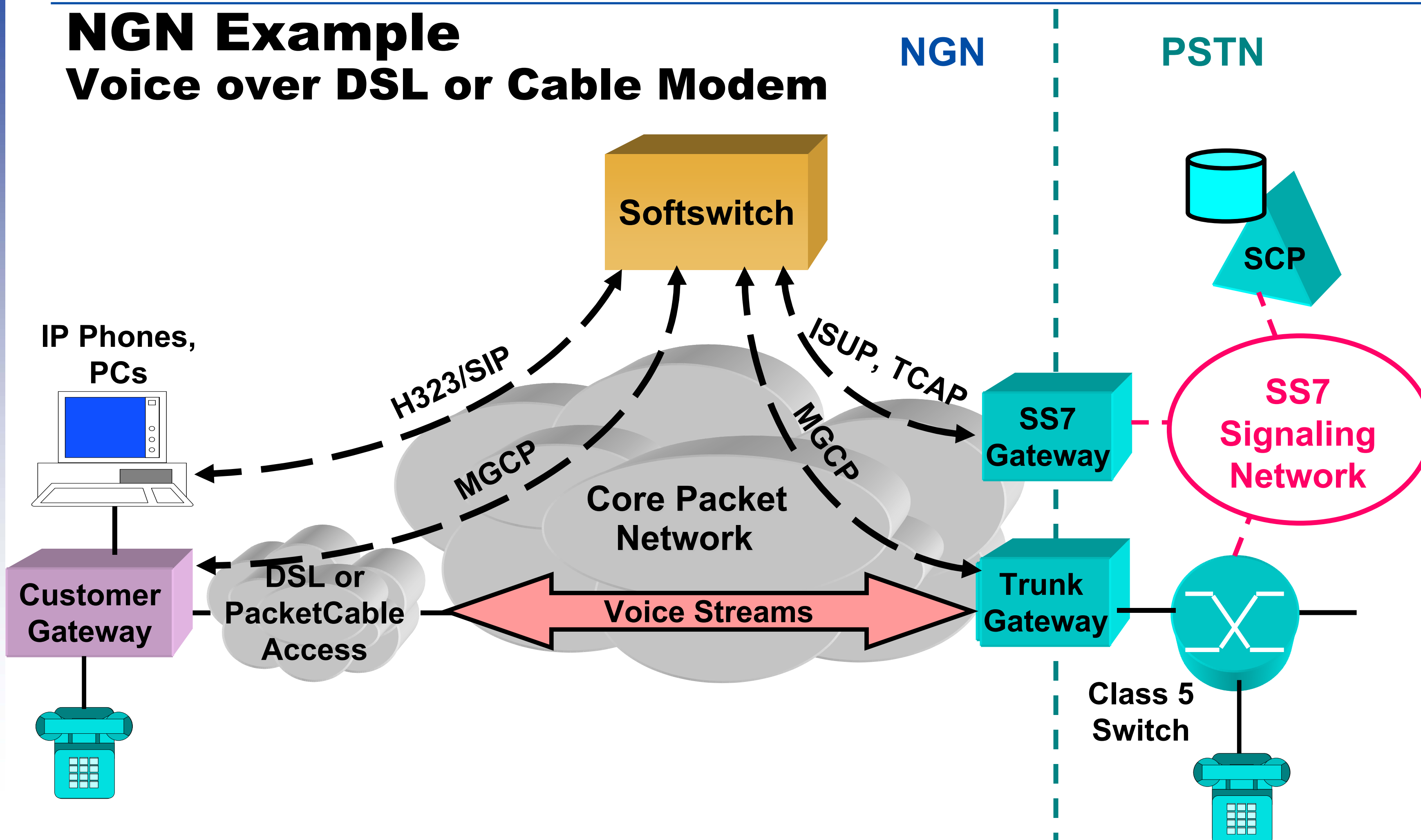
NGN Architecture
Next-Generation Network

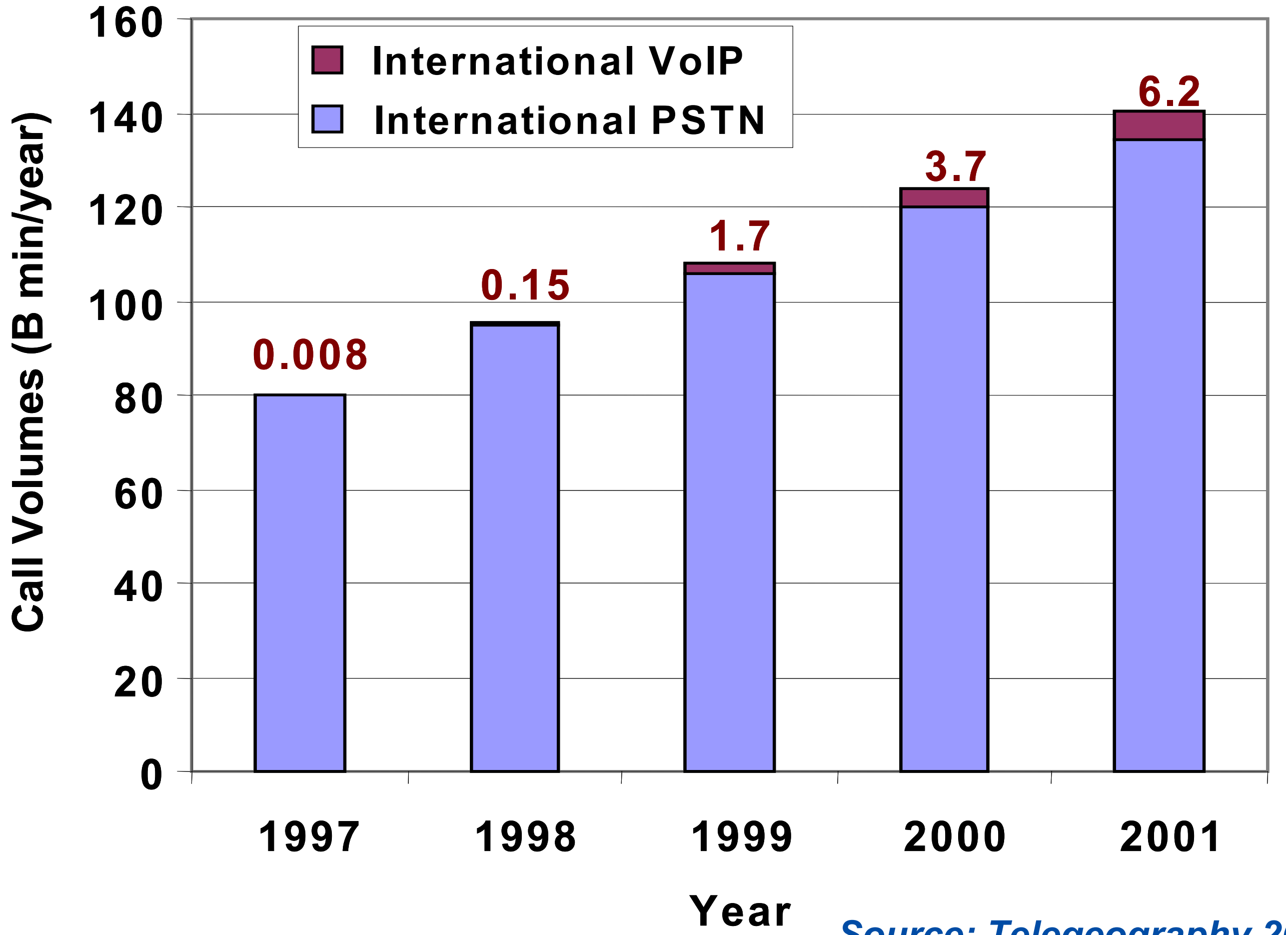
- Oriented towards application of VoIP (or VoATM) to large-scale public networks
- Focus on scalability, network control, support for traditional phones, sophisticated gateway (GW) to the PSTN and its services
- Media GW interfaces voice stream to PSTN trunk or phone line
- Signaling GW allows signaling 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

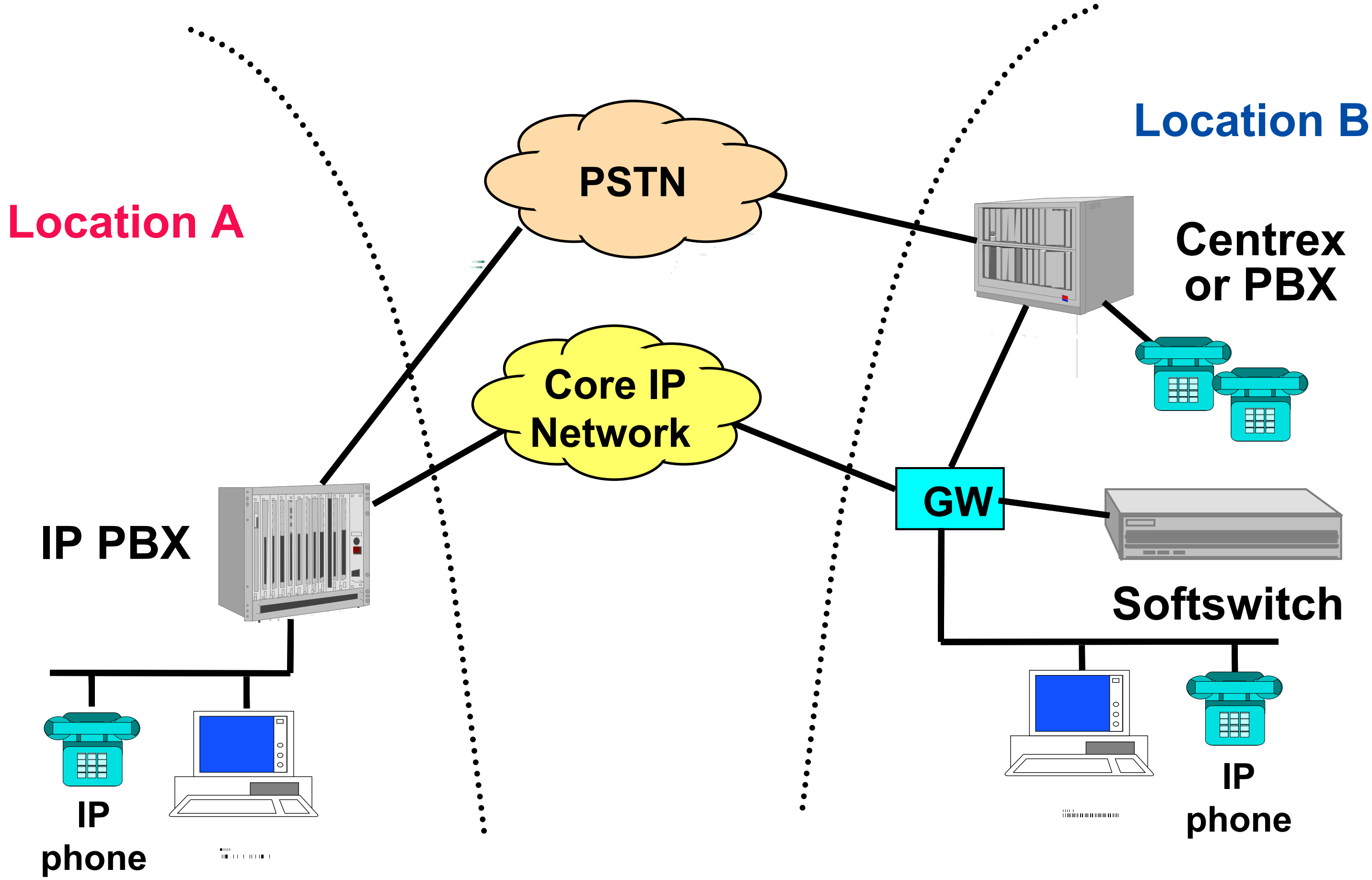


Source: Telegeography 2001
(2001 figures were projections)

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)

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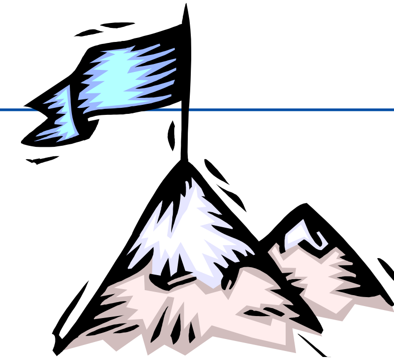
Performance from Experience

LTS

Continuing Challenges

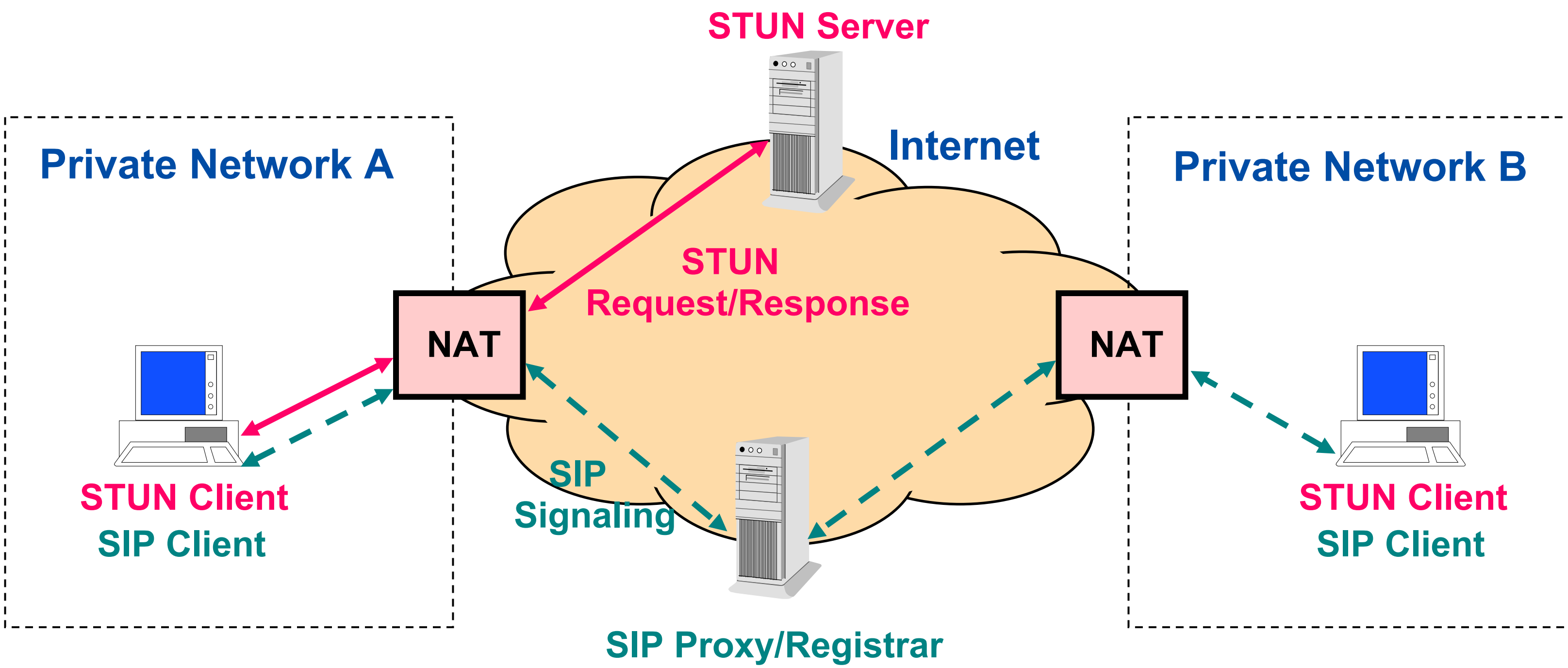


- Quality of Service
  - Diffserv, MPLS, traffic engineering, bandwidth brokers, call admission…
  - What is really needed for consolidated voice and data networks?
- Security, reliability
- Extending SIP to provide conference control
- Operations (configuration of IP phones, version control and upgrading 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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