# Voice/Video over Internet Protocol (VoIP)

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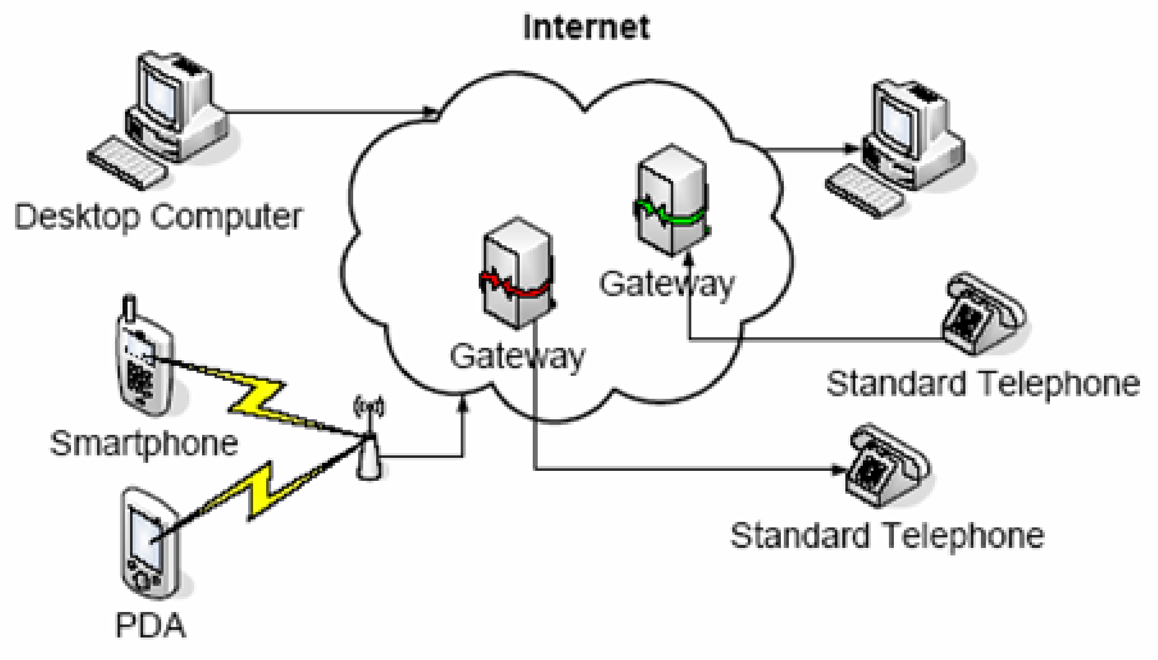
# Outline

- Introduction
- Benefits compared traditional telephony network (PSTN)
- Signaling protocol
  - H.323
  - SIP (Session Initiate Protocol)
- Application
  - Pre-paid phone card
  - Skype
- Conclusion

# Introduction of VoIP

- Transmit voice traffic using Internet Protocol (IP) instead of conventional telephony network (PSTN)
- Packet switch instead of circuit switch



# Why VoIP?

- Integration of voice, data, and video
  - New functions (ex. video phone)
- Universal presence of Internet
  - Lower equipment cost
  - Lower operation cost
- Potentially lower bandwidth requirement
  - PSTN: 8K x 8bit =64Kbps (G.711 standard)

# Challenge of VoIP

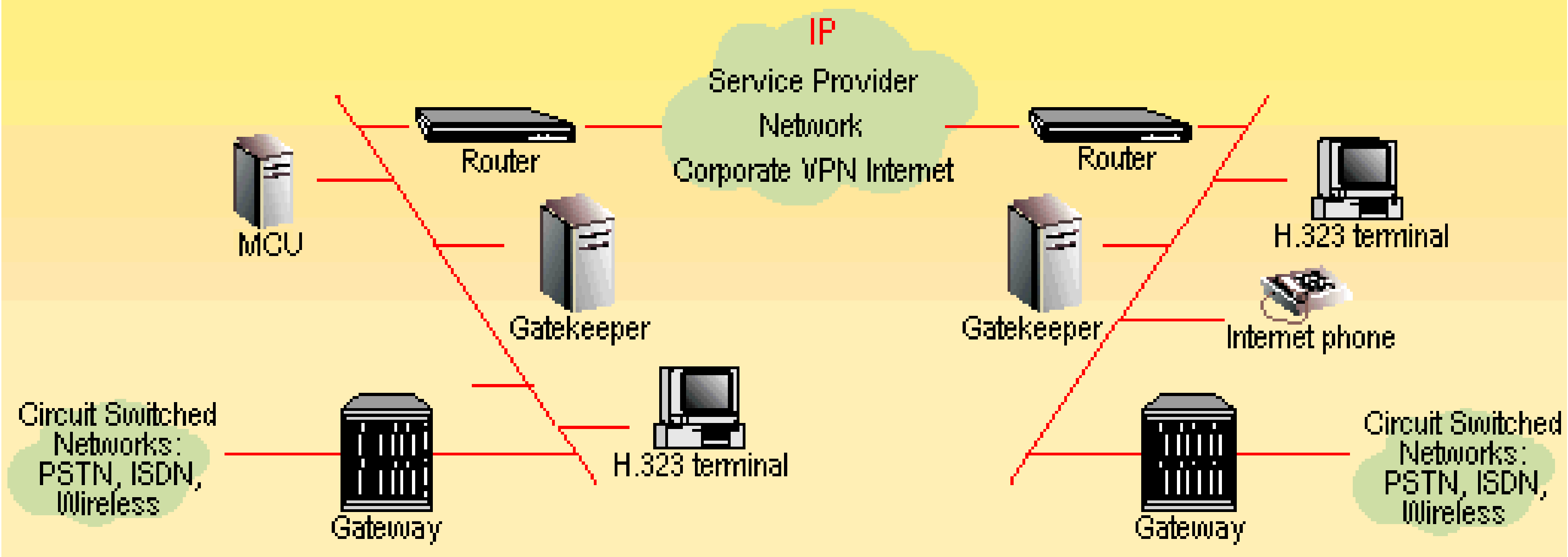
- Data
  - Asynchronous
  - Error sensitive
- Voice
  - Synchronous
  - More tolerant for errors
- VoIP must
  - Meet all requirements of traditional telephony network ex. quality
  - Offer new function and attractive capabilities with lower cost
  - Easy to use

# H.323

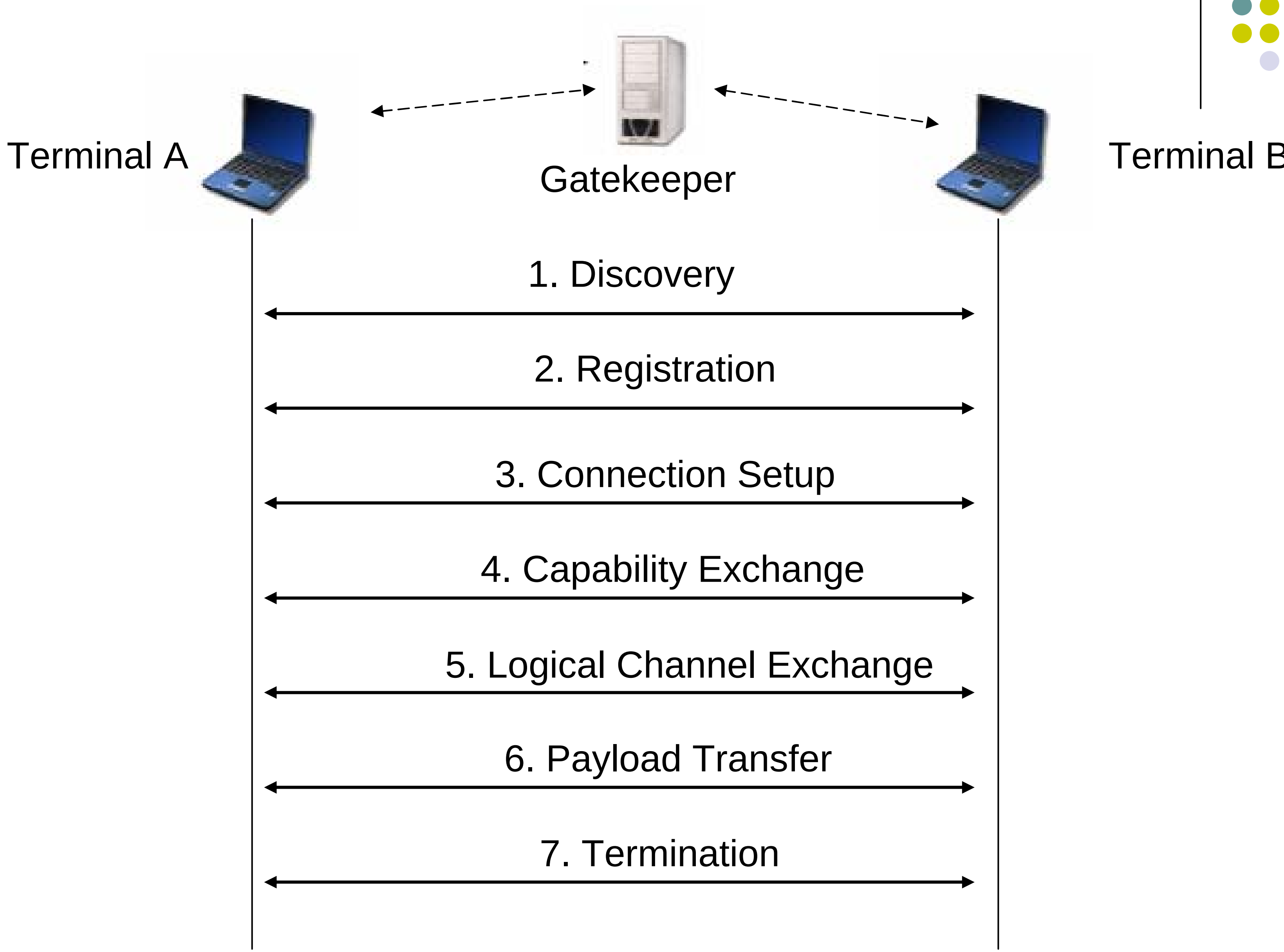
- Enable the exchange of media stream between H.323 endpoints
- Developed by International Telecommunications Union (ITU)
- Advantages
  - Prevalent
  - Powerful
- Drawbacks
  - Complex ➔ expensive

# H.323 Architecture

- **Gateway**
  - an H.323 endpoint that provides translation service between the H.323 network and another type of network
- **Gatekeeper**
  - control a number of H.323 terminals, gateways, and multipoint controllers (MCs)
- **Multipoint controller**
  - An H.323 endpoint that manages multipoint conferences between three or more terminals



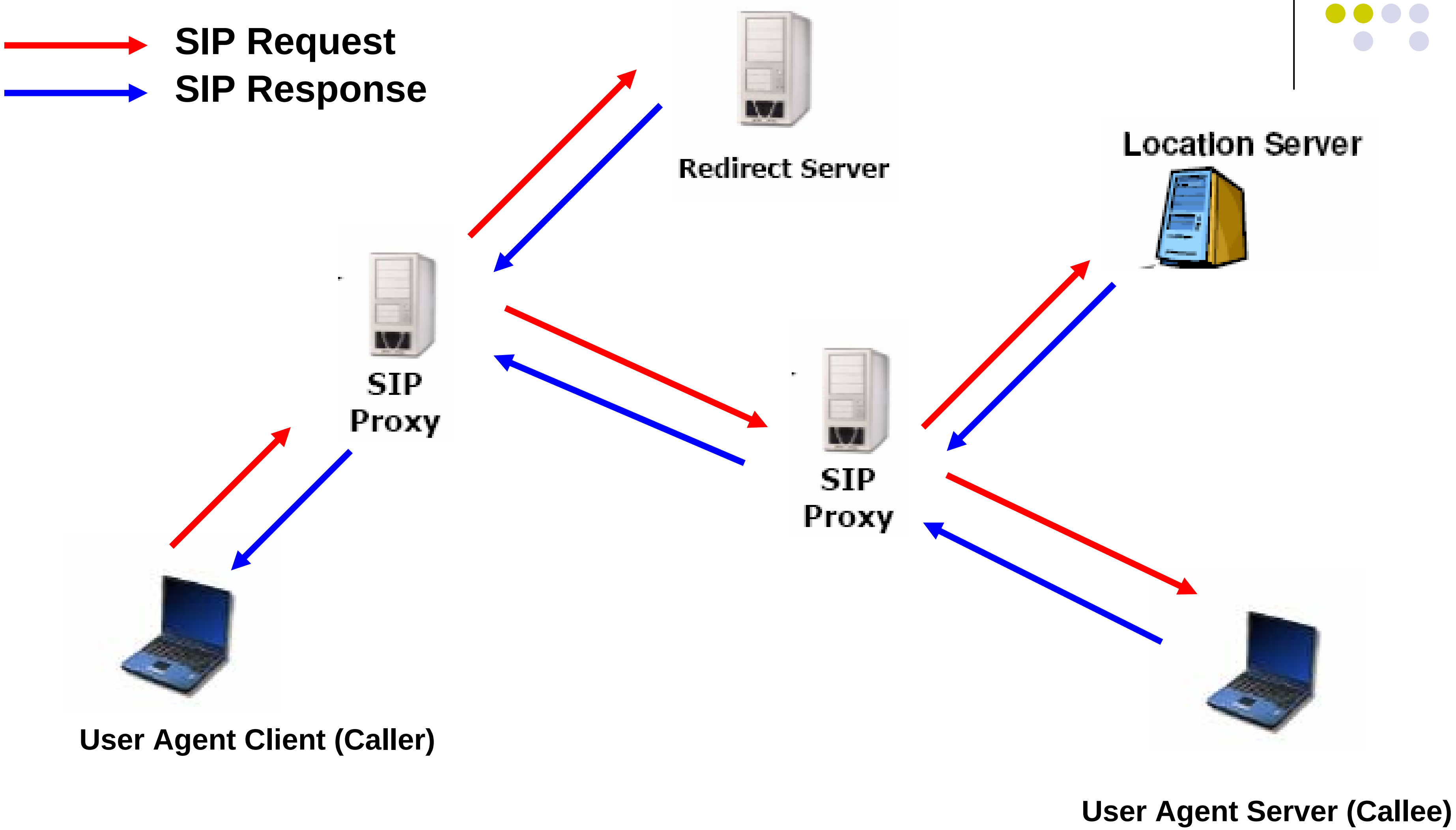
# H.323 Operation



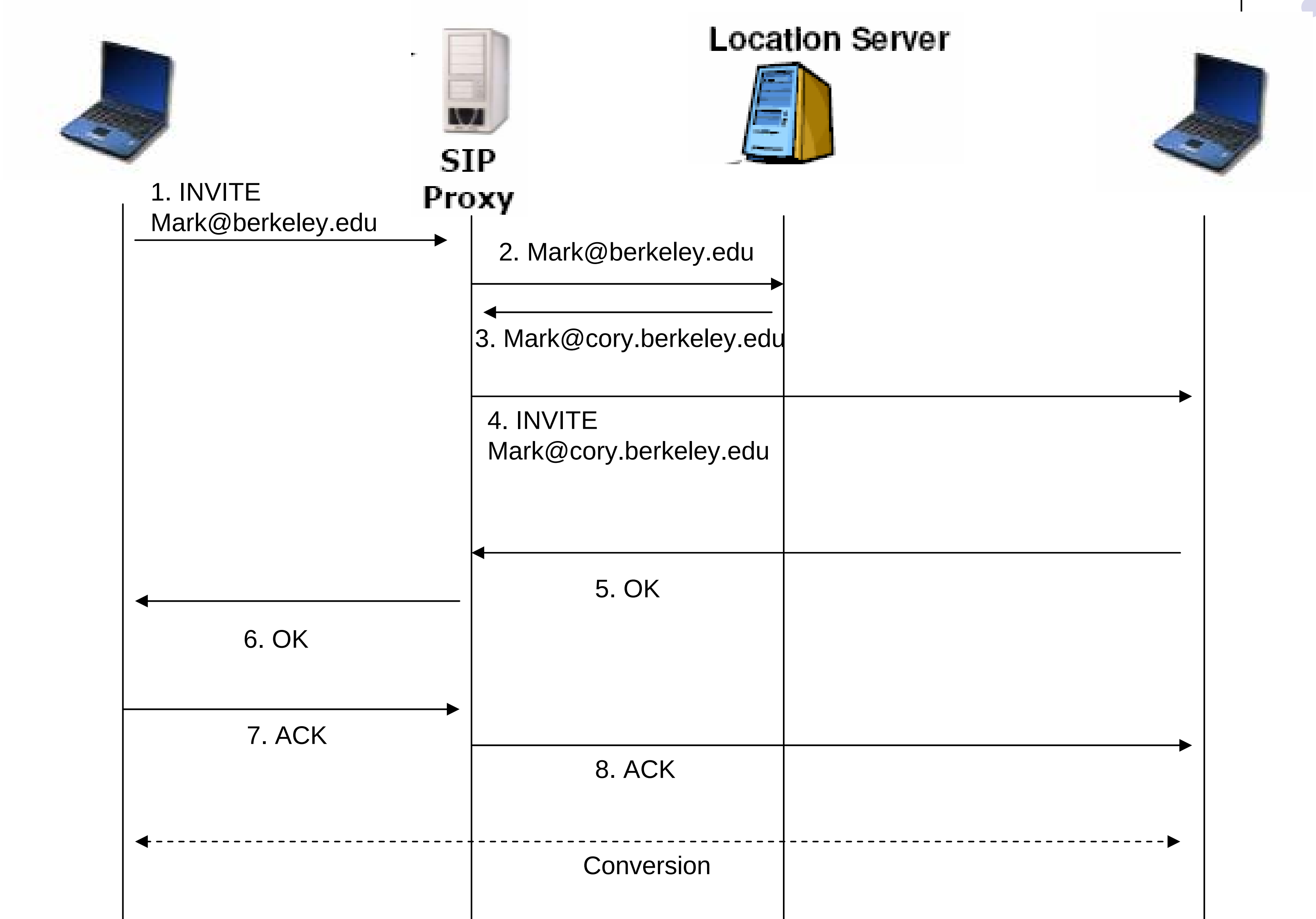
# SIP

- A call processing system for voice/video/data
- Developed by Internet Engineering Task Force (IETF)
- Function
  - Negotiate capabilities
  - sets up and clears calls
  - Finds called parties in the Internet
- Advantages
  - Simpler and more flexible signaling protocol
  - Various pieces of information can be included within messages

# Architecture of SIP



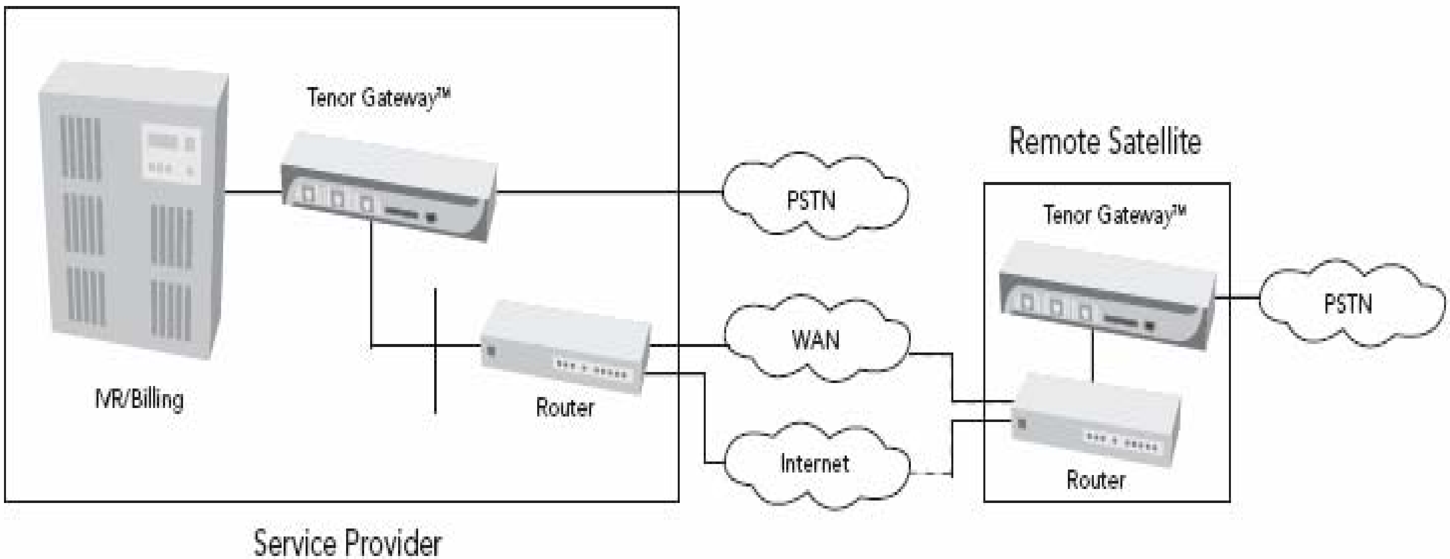
# Operation of SIP



# Comparison between H.323 and SIP

| SIP | H.323 |
|---|---|
| Newer developed protocol → simple | Earlier protocol → complex |
| IETF | ITU |
| Many vendors developing products | The majority of existing IP telephony products rely on H.323 |
| Leave issues of reliability to underlying network | Assume the fallibility of network |
| SIP messages are formatted as text | Binary format doesn't sit well with the internet |

# Prepaid Phone Card



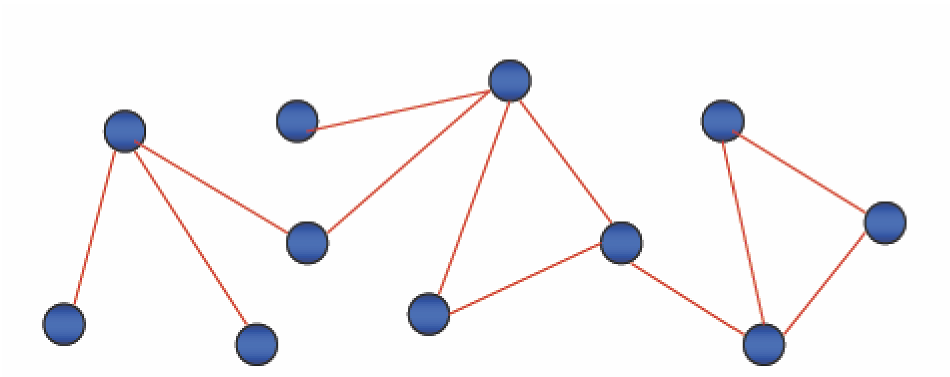

**<u>Advantage:</u>**

**--More services available (e.g. PTT)**

**--Low cost**

# Skype

- P2P Overlay Netowork + Proprietary Protocol
- Distributed System
- Directly sharing the computer resources



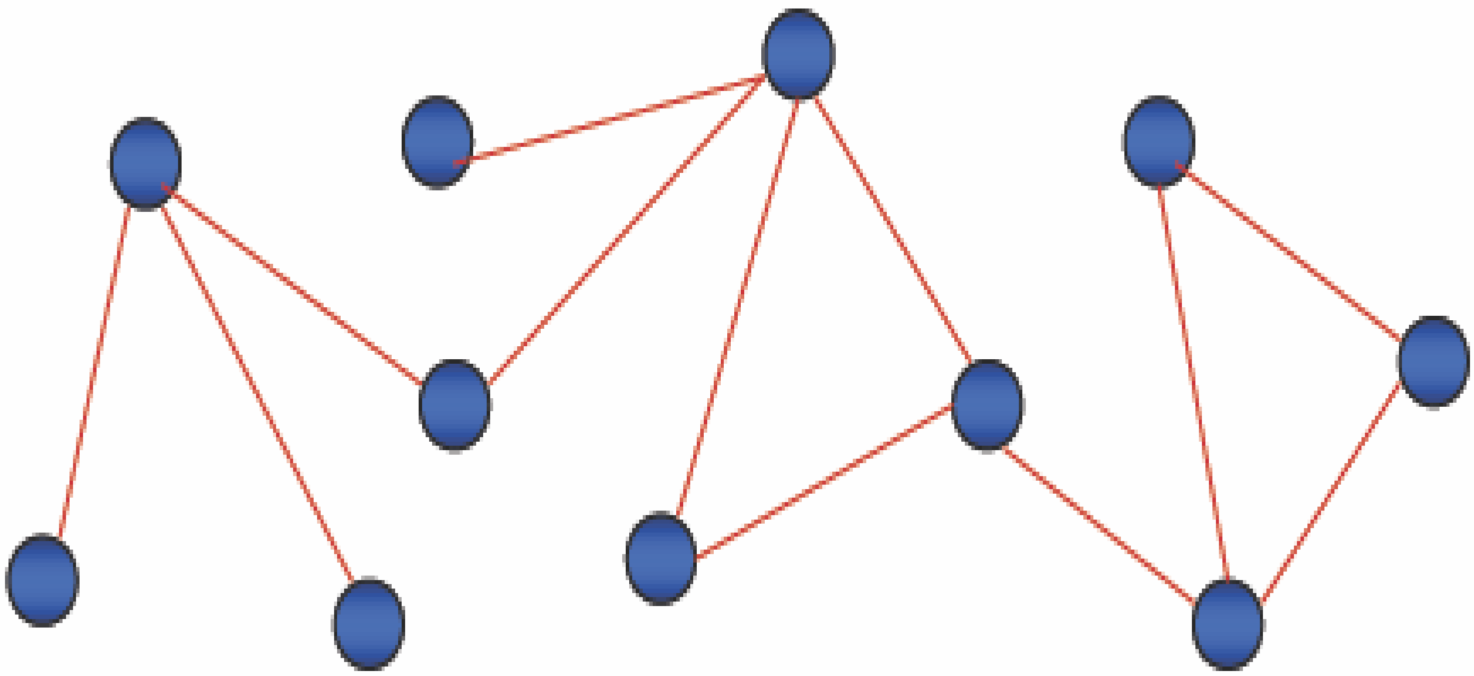
# What is Overlay Network

- The operation of any peer-to-peer system relies on a network of peer computers (nodes), and connections (edges) between them.
- This network is formed on top of –and independently from—the underlying physical computer (typically IP) network and is thus referred to as an “overlay” network.

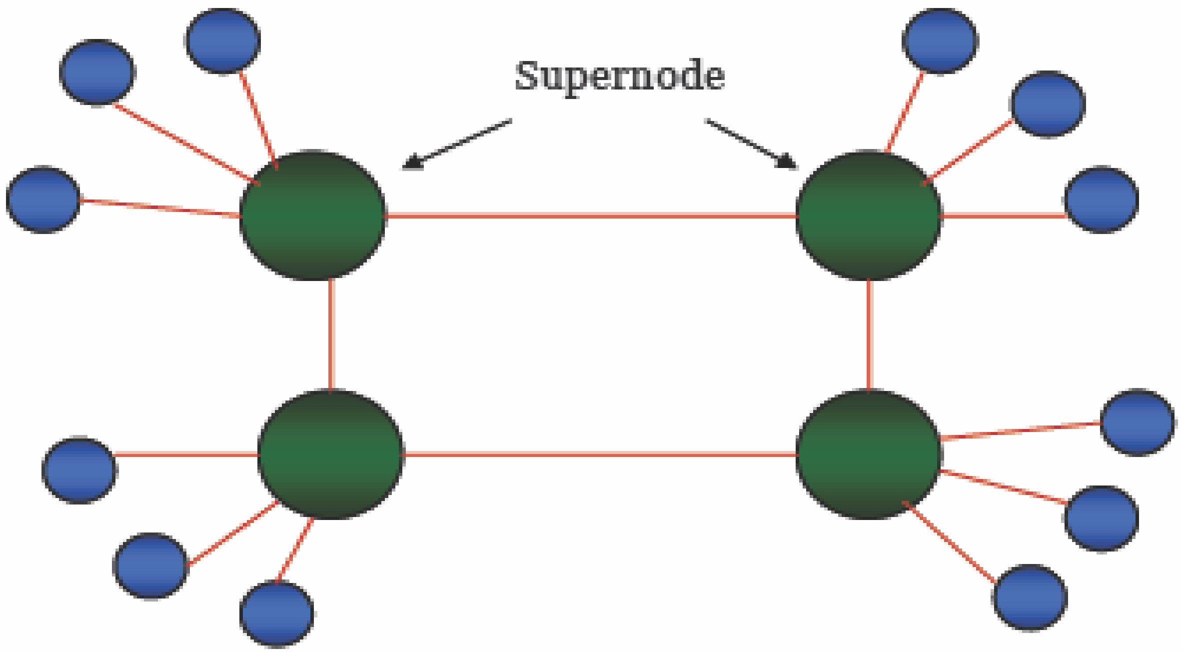
# Classification of P2P

- Purely Decentralized Network
  - Gnutella
- Partially Centralized Network
  - KaZaA
- Hybrid Decentralized Network
  - Napster

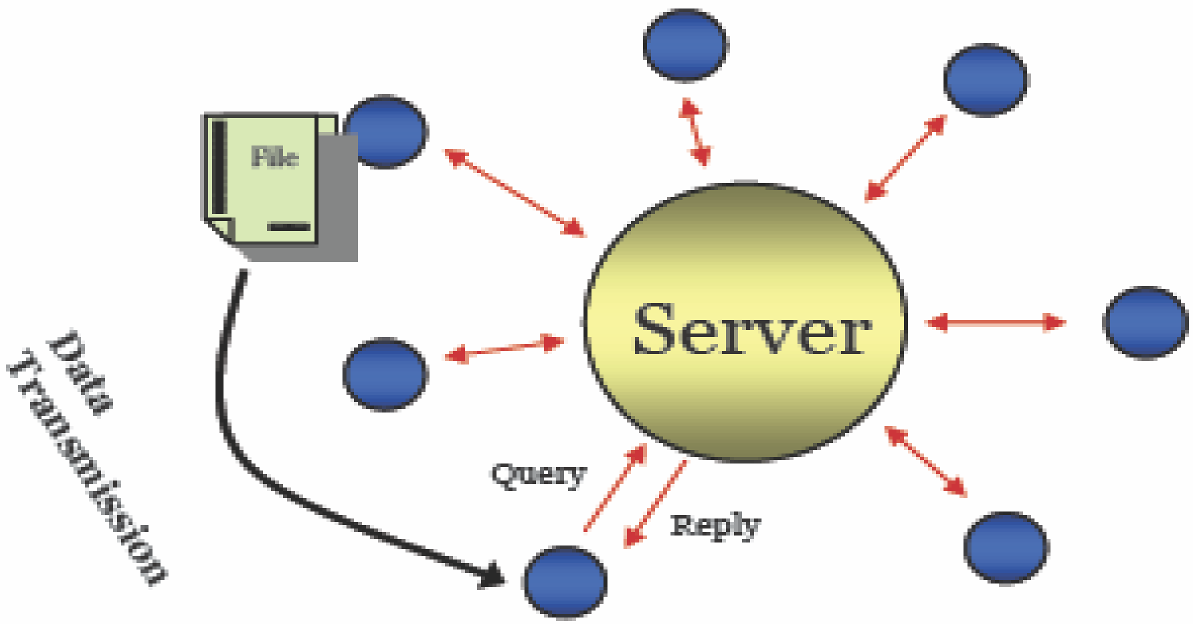
# Purely Decentralized



# Partially Centralized

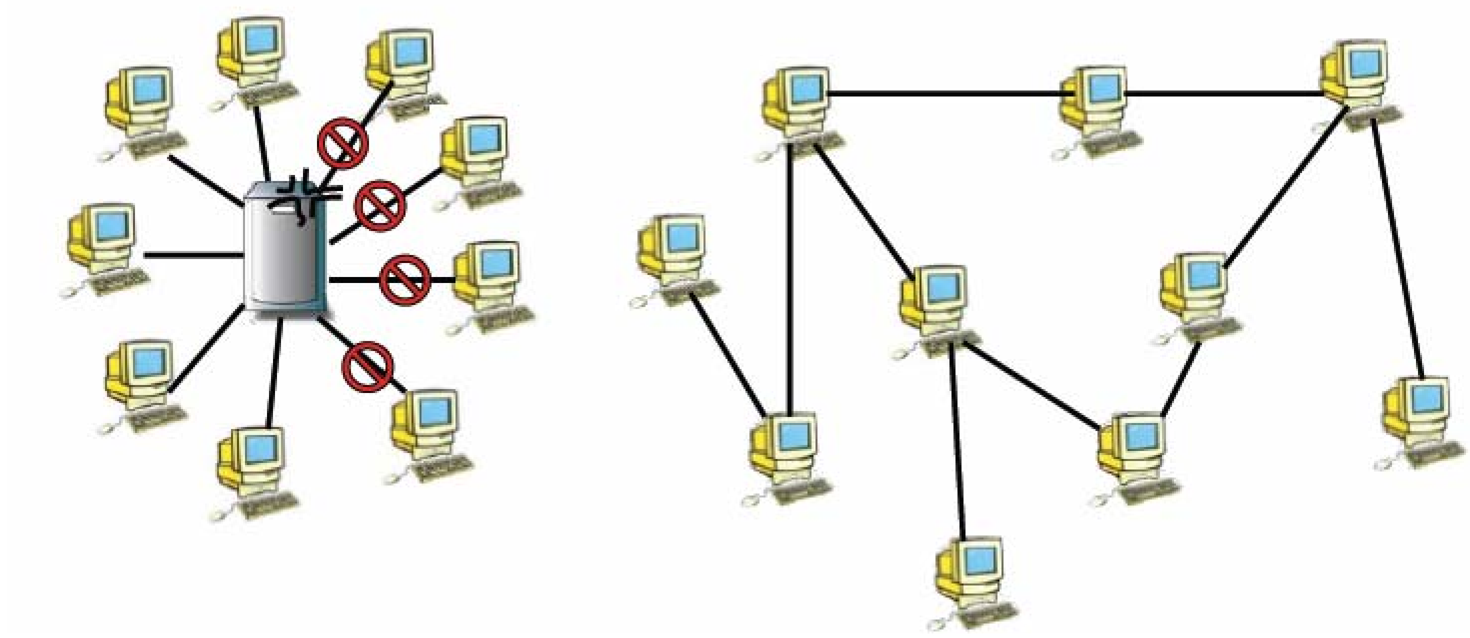


# Hybrid Decentralized



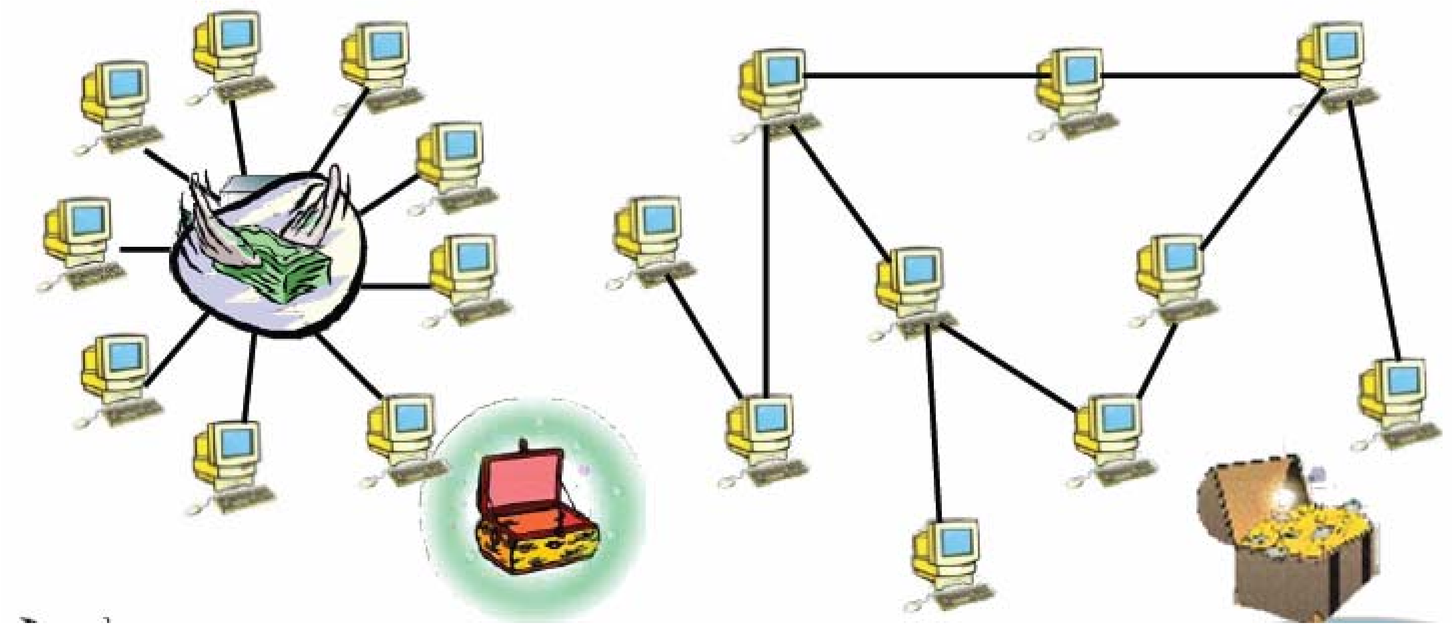
# Advantage of P2P

- **Scalability**
  - A dramatic increase in the number of nodes or documents will have minimal effect on performance and availability.



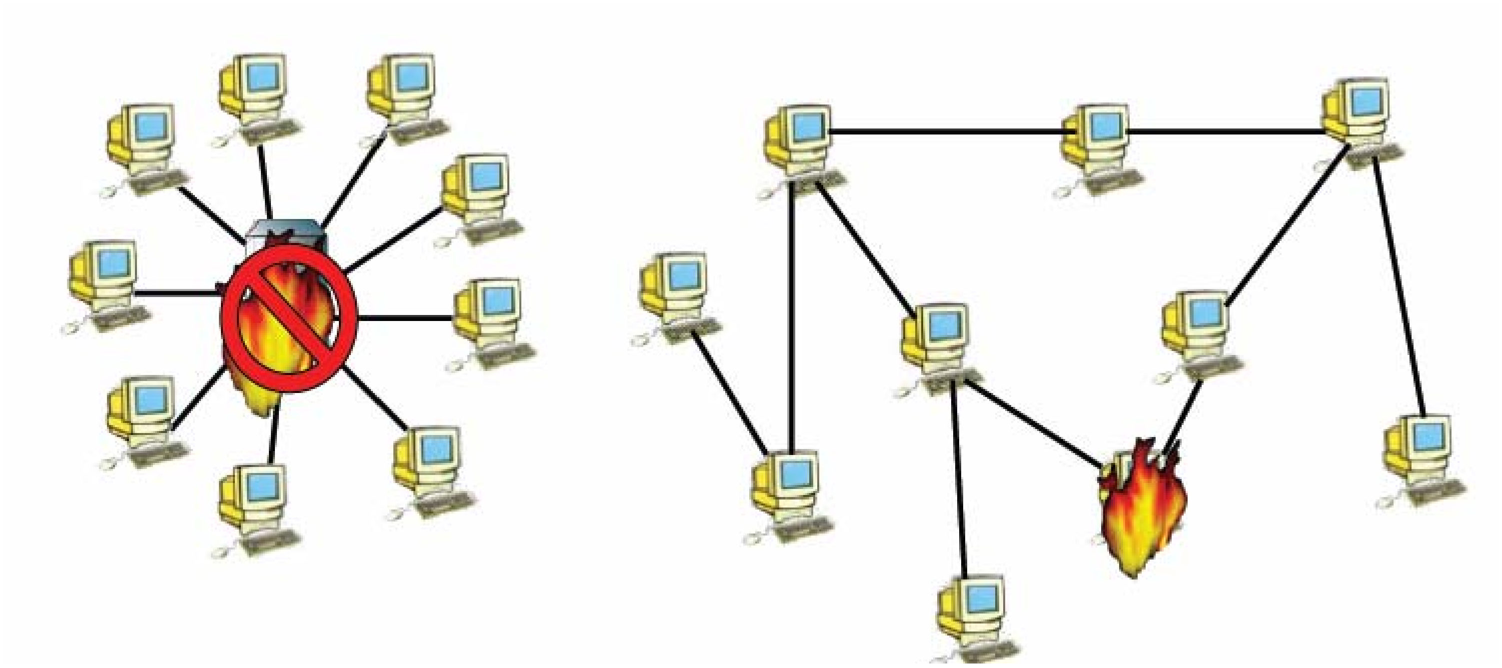
# Advantage of P2P

- Cost—there is no need to buy expensive machines for servers



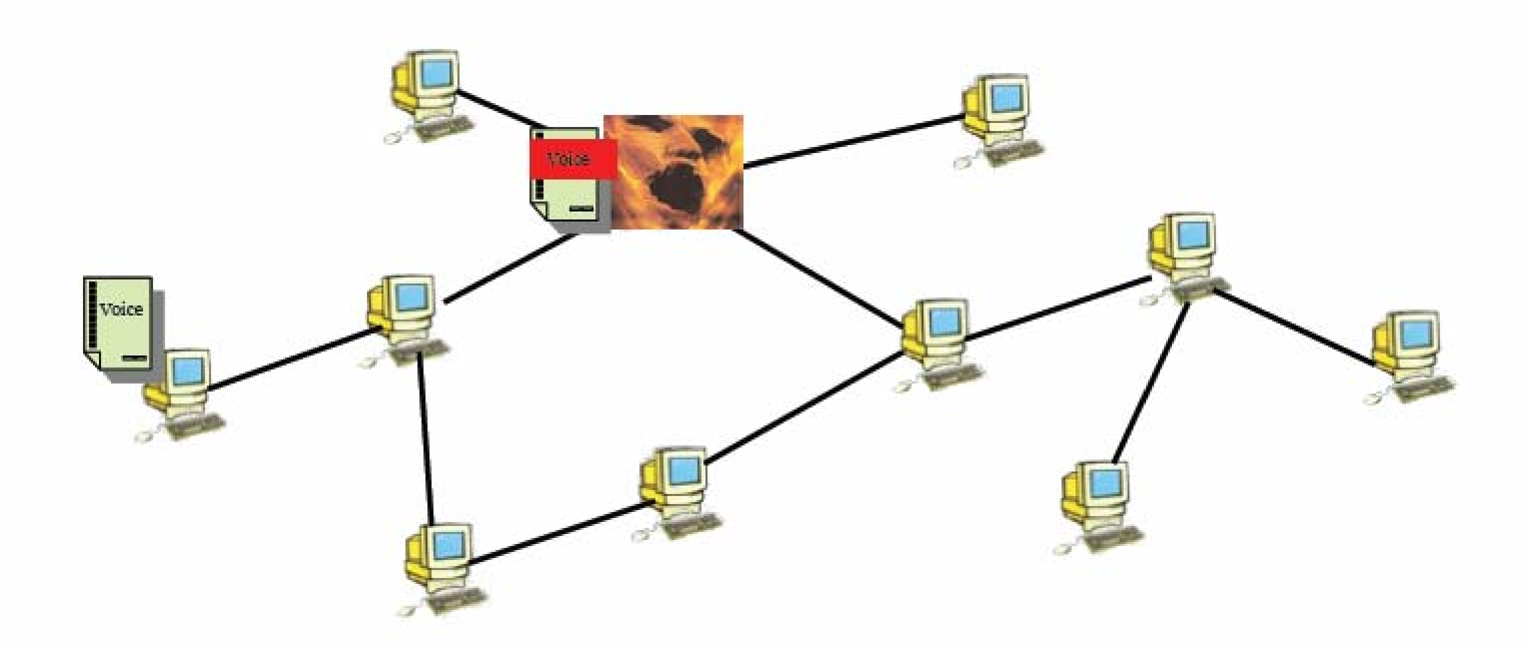
# Advantage of P2P

- Robustness and Reliability



# Issues of P2P

- Security and Authenticity

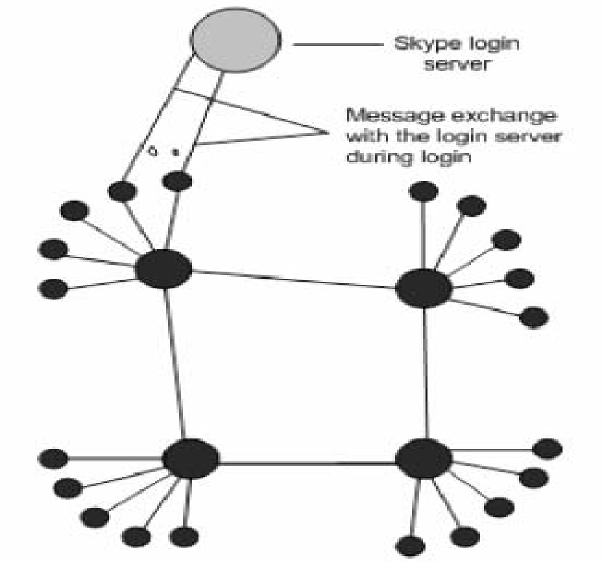


# Issues of P2P

- **Performance**
  - The time required for performing the operations allowed by the system, typically routing, searching, and retrieval of documents.
- **Fairness**
  - Ensuring that users offer and consume resources in a fair and balanced manner.
- **Resource Management Capabilities**

# Skype Protocol

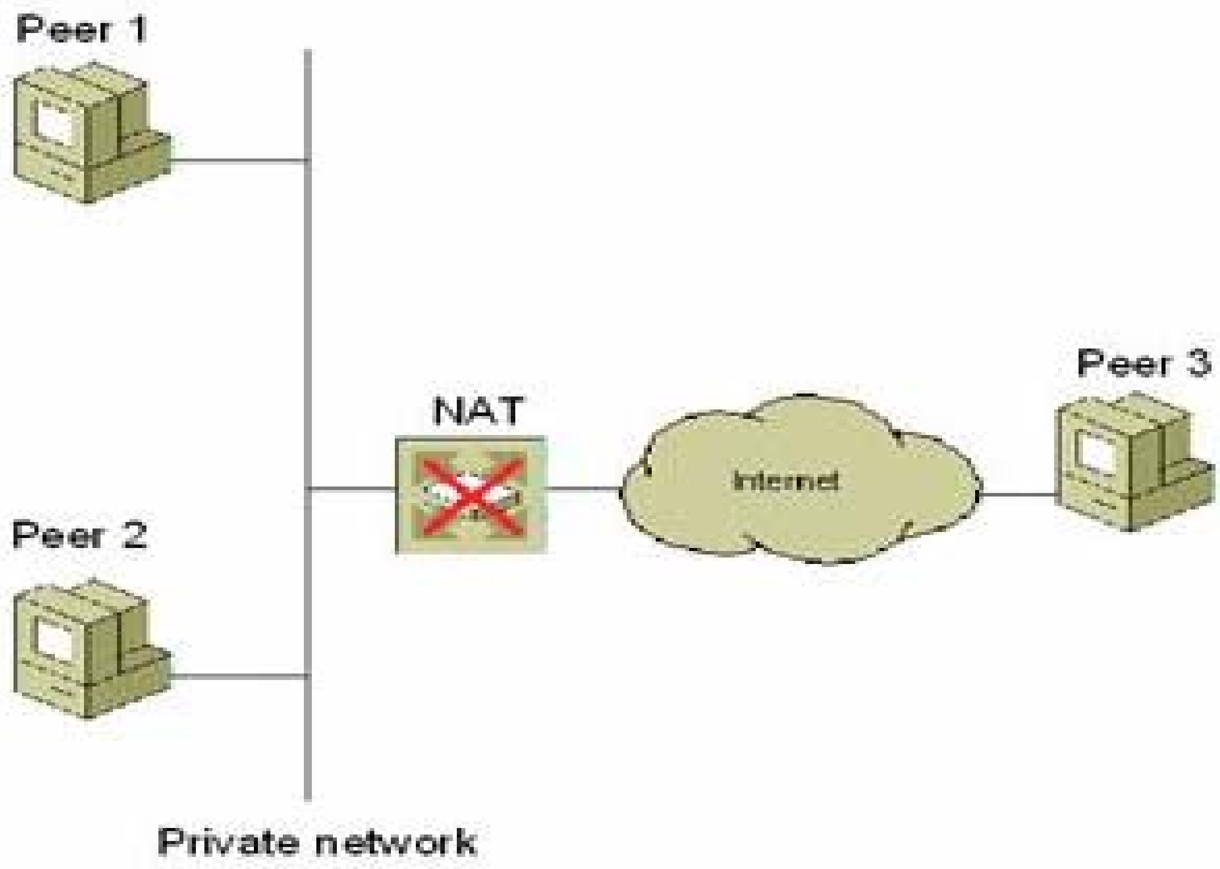
- Any SkypeClient (SC) with a **public IP address**having **sufficient CPU**, **memory**, and **network bandwidth**isa candidate to become a super node (SN)



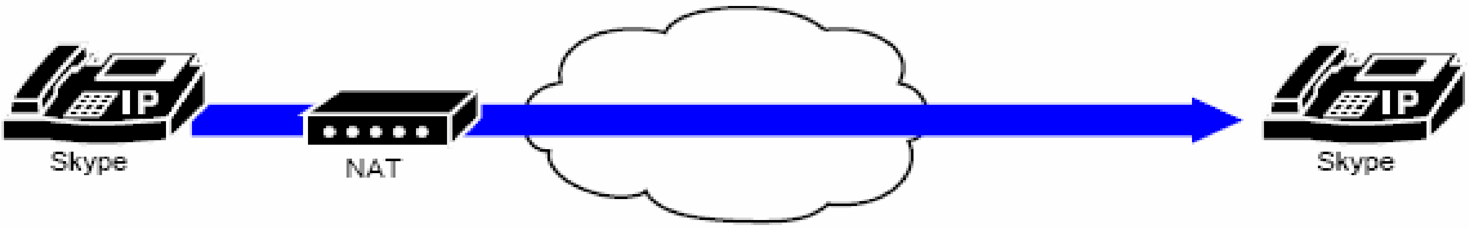
# Login

- **Login is perhaps the most critical function to the Skype operation**
- **During this process, a Skype Client (SC)**
  - Authenticates its user name and password with the login server
  - Advertises its presence to other peers and its buddies
  - Determines the type of NAT and firewall it is behind
  - Discovers online Skype nodes with public IP addresses

# Network Address Translation

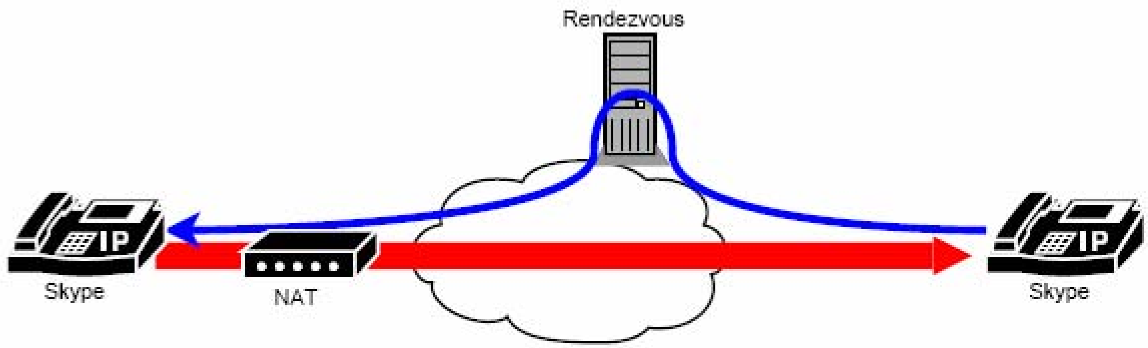


# NAT




| Level 0: Initiator NAT'ed |
| --- |
| **Solution:** Don't embed IP address in payload<br>**Apps:** Most old software, almost all new software |

# NAT

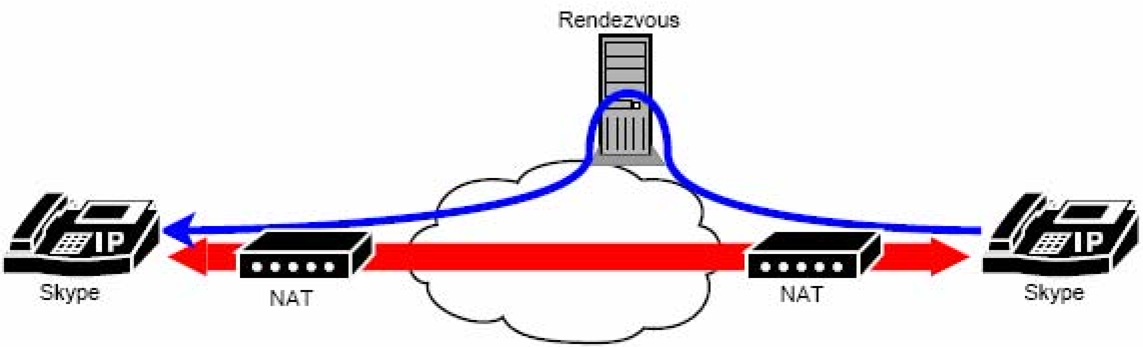



Level 1: Recipient NAT'ed

**Solution:** Use Rendezvous Service

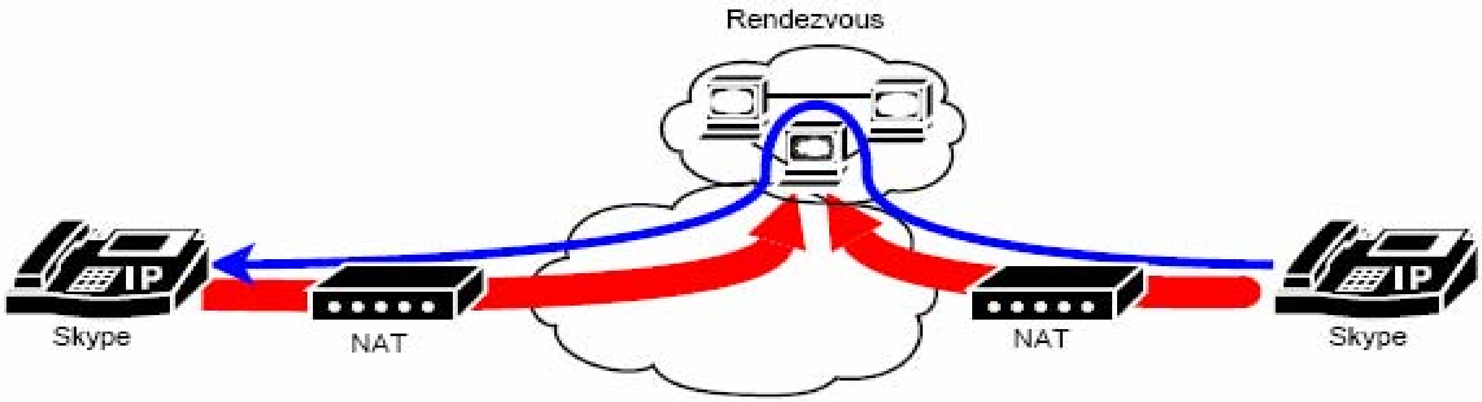
**Apps:** Bittorrent, MSN, Yahoo, Skype, . . .

# NAT




| Level 2: Both NAT'ed (well-behaved NATs) |
|---|
| **Solution:** Use STUN (UDP) or STUNT (TCP)<br>**Apps:** MSN, Yahoo, Skype, . . . |

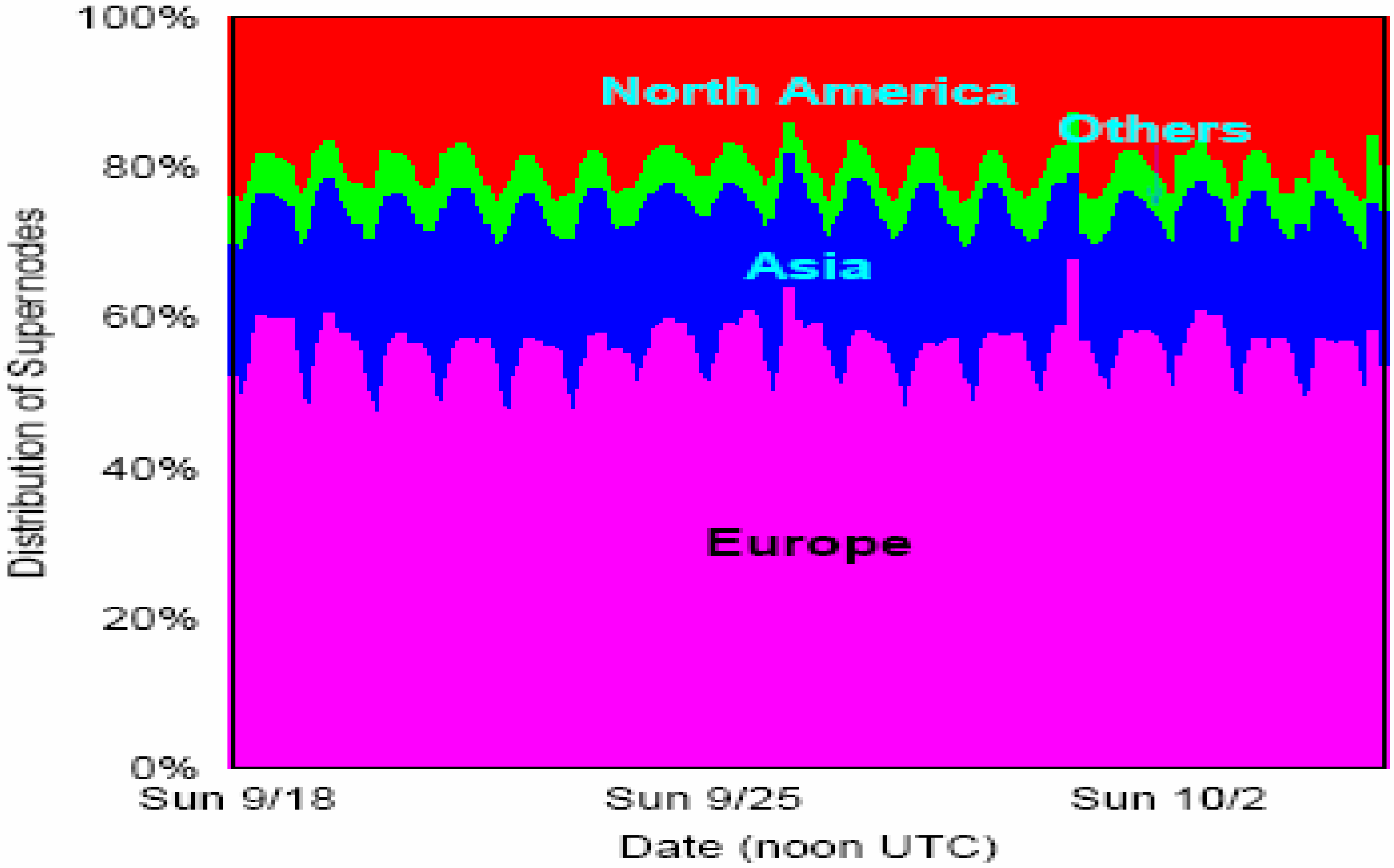
# NAT




Level 3: Both NAT'ed (broken NATs)

**Solution:** Use TURN + P2P
**Apps:** Skype

# Super Node Density



# Call Placement Case 1

- **The call signaling is always carried over TCP**
- **Both users were on public IP address**
- The caller SC established a TCP connection with the callee SC

# Call Placement Case2

- **The caller was behind port-restricted NAT and callee was on public IP address**
- The caller sends signaling information over TCP to an online Skype node which forwarded it to callee over TCP
- The online node also routed voice packets from caller to callee over UDP and vice versa

# Call Placement Case 3

- **Both users were behind port-restricted NAT and UDP-restricted firewall**
- Caller SC sent media over TCP to an online node, which forwarded it to callee SC over TCP and vice versa
- **Advantages of having a node route the voice packets from caller and callee**
  - It provides a mechanism for users behind NAT and firewall to talk to each other
  - If other users want to participate in a conference, this node serves as a mixer
- **Call tear-down**

# Skype Functions

- **Codec Frequency Range**
- The min. and max. audible frequency Skype codecs allow to pass through are 50 Hz and 8000 Hz
- **Congestion Control**
- Uplink and downlink bandwidth of **2k**bytes/s each was necessary for reasonable call quality
- The voice was almost unintelligible at an uplink and downlink bandwidth of **1.5k**bytes/s

# Conclusion

- VoIP can bring us a lot more conveniences which can’t be provided by traditional PSTN network. However, there remains some issues to be resolved
  - VoIP over wireless network, such as admission control, congestion detection…etc.
- Security Issue