An Analysis of the Skype Peer-to-Peer Internet Telephony Protocol

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What's Overlay Network

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What's P2P ?

What is P2P？

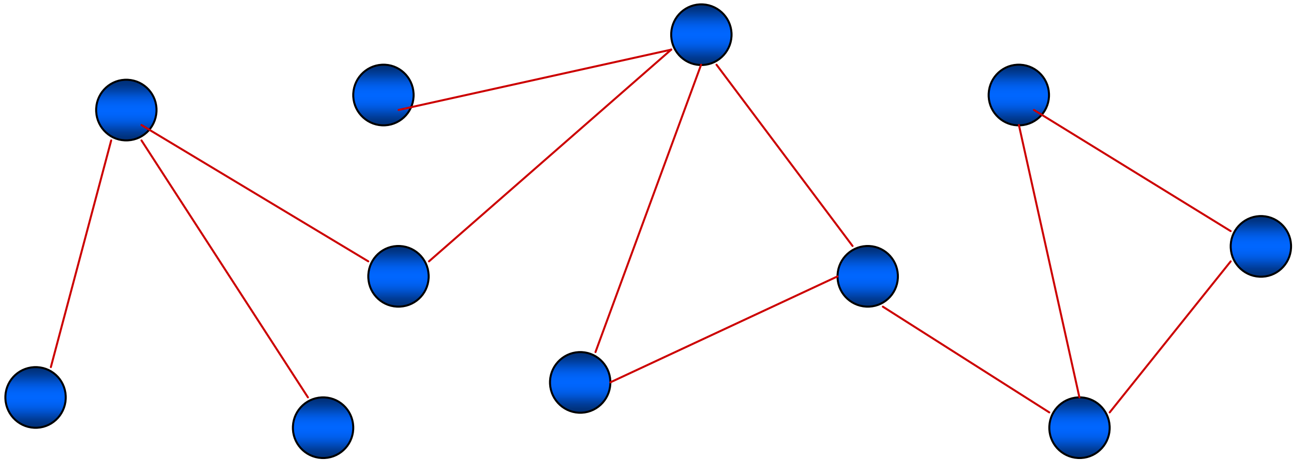
- **Distributed systems**
- **Direct sharing of computer resources**
- **Without requiring the intermediation or support of a global centralized server or authority.**

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.

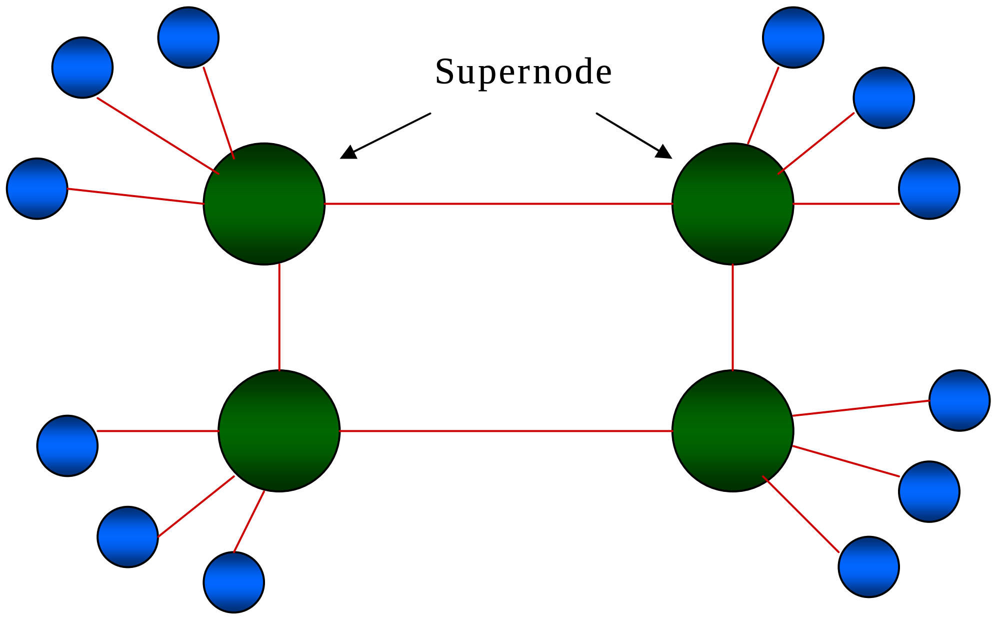
Overlay Network Architecture (1/3)

- **Purely Decentralized Architectures**
  - All nodes in the network perform exactly the same tasks, acting both as servers and clients, and there is no central coordination of their activities.



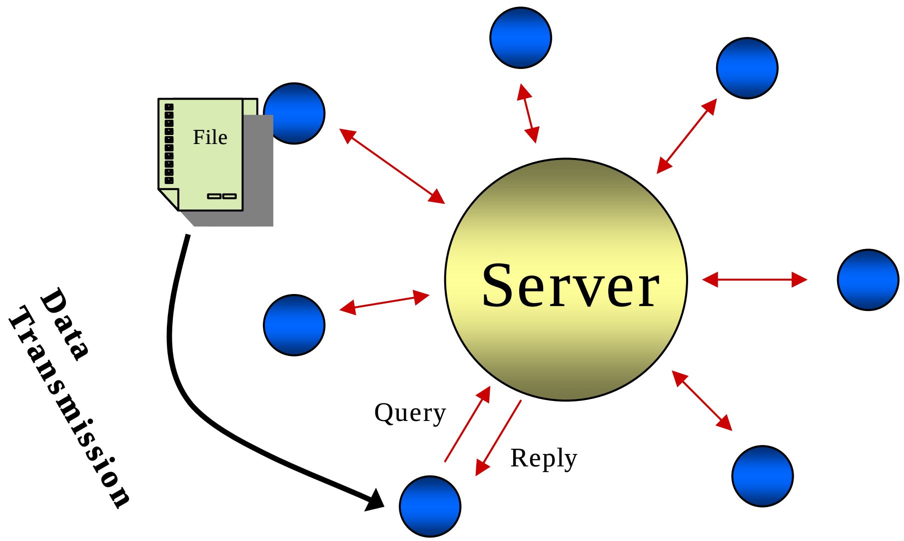
Overlay Network Architecture (2/3)

- **Partially Centralized Architectures**



Overlay Network Architecture (3/3)

- **Hybrid Decentralized Architectures**

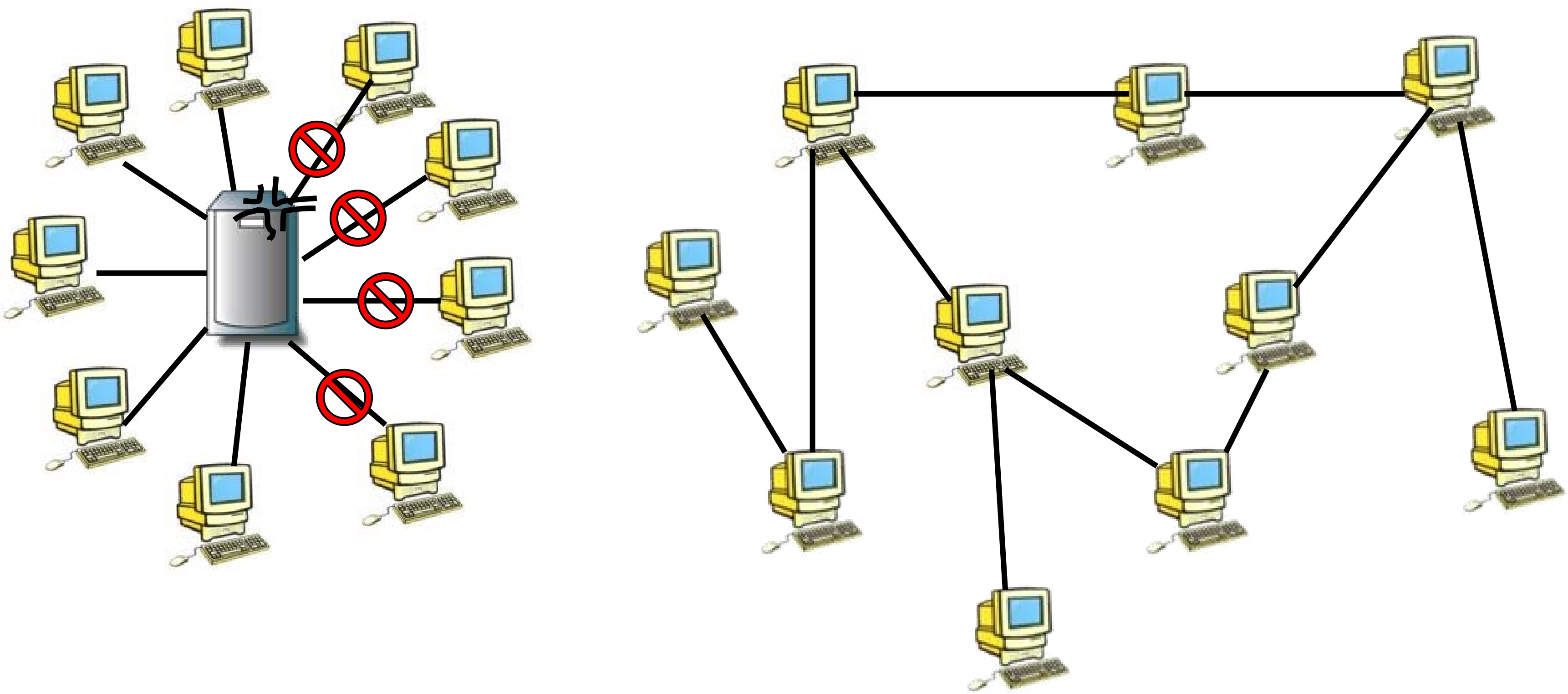


Classification of P2P Applications

- **Communication and Collaboration**
- **Distributed Computation**
- **Database Systems**
- **Content Distribution**
  - **Peer-to-Peer File Exchange Systems**
    - *Napster*：Hybrid decentralized.
    - *KaZaA*：Partially centralized.
    - *Gnutella*：Purely decentralized.

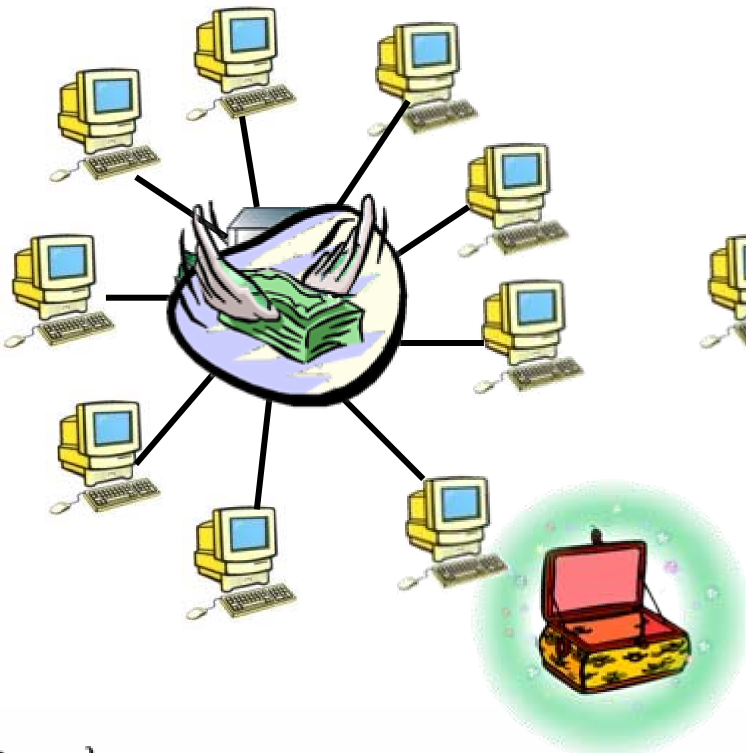
Advantages of P2P (1/3)

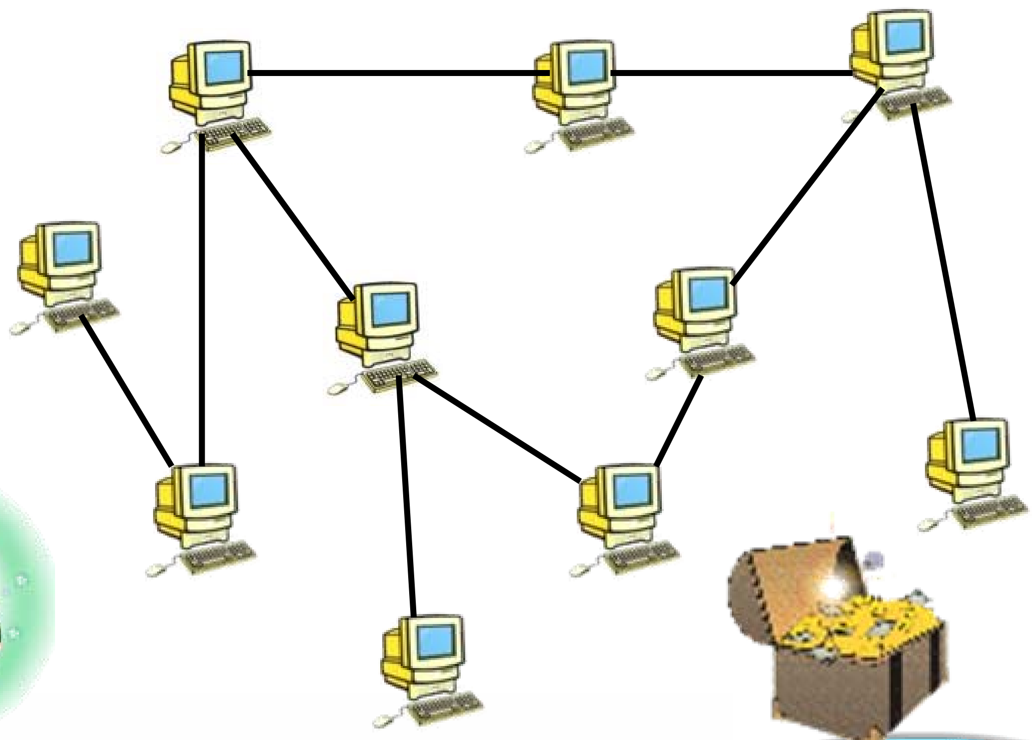
- **Scalability**
  - A dramatic increase in the number of nodes or documents will have minimal effect on performance and availability.



Advantages of P2P (2/3)

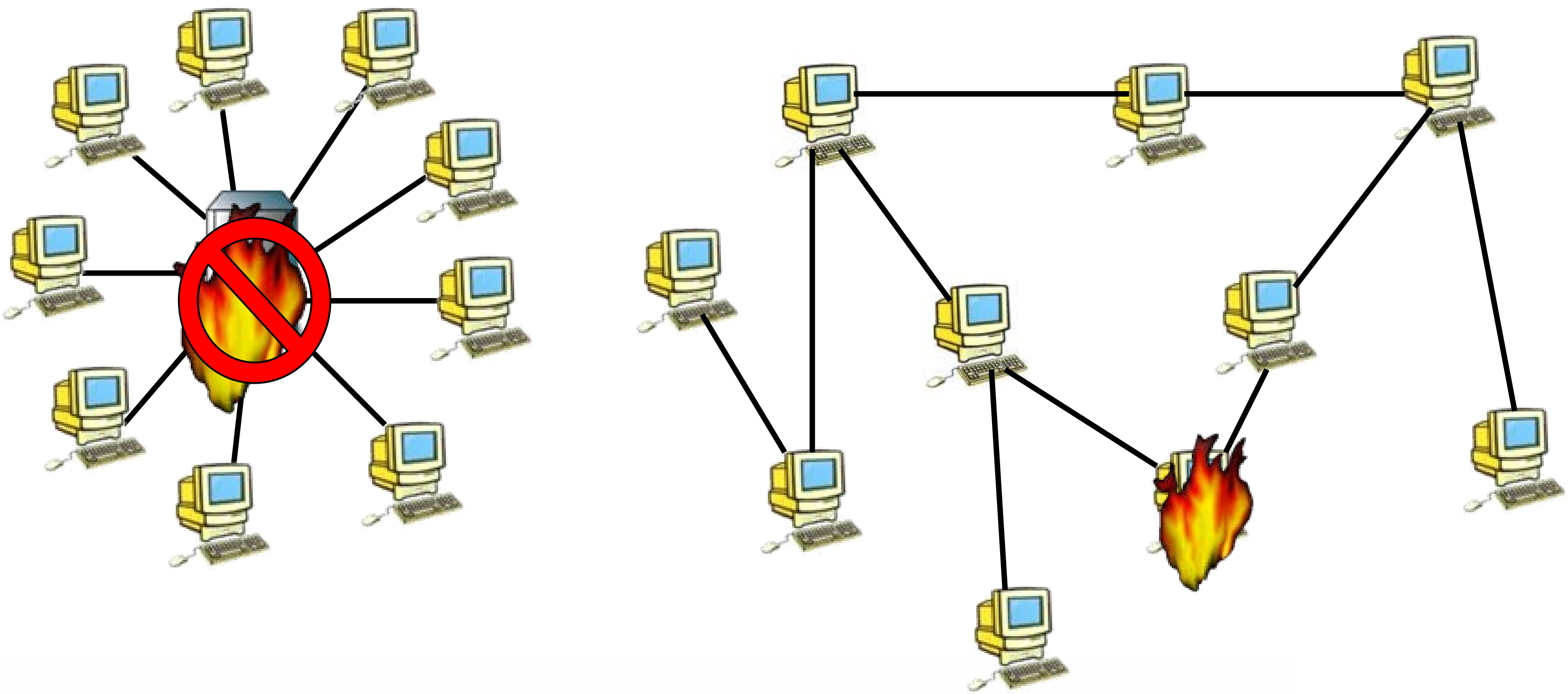
- **Low Cost**
  - There is no need to buy more special machines to be servers. Every computer can be a server and a client at the same time.





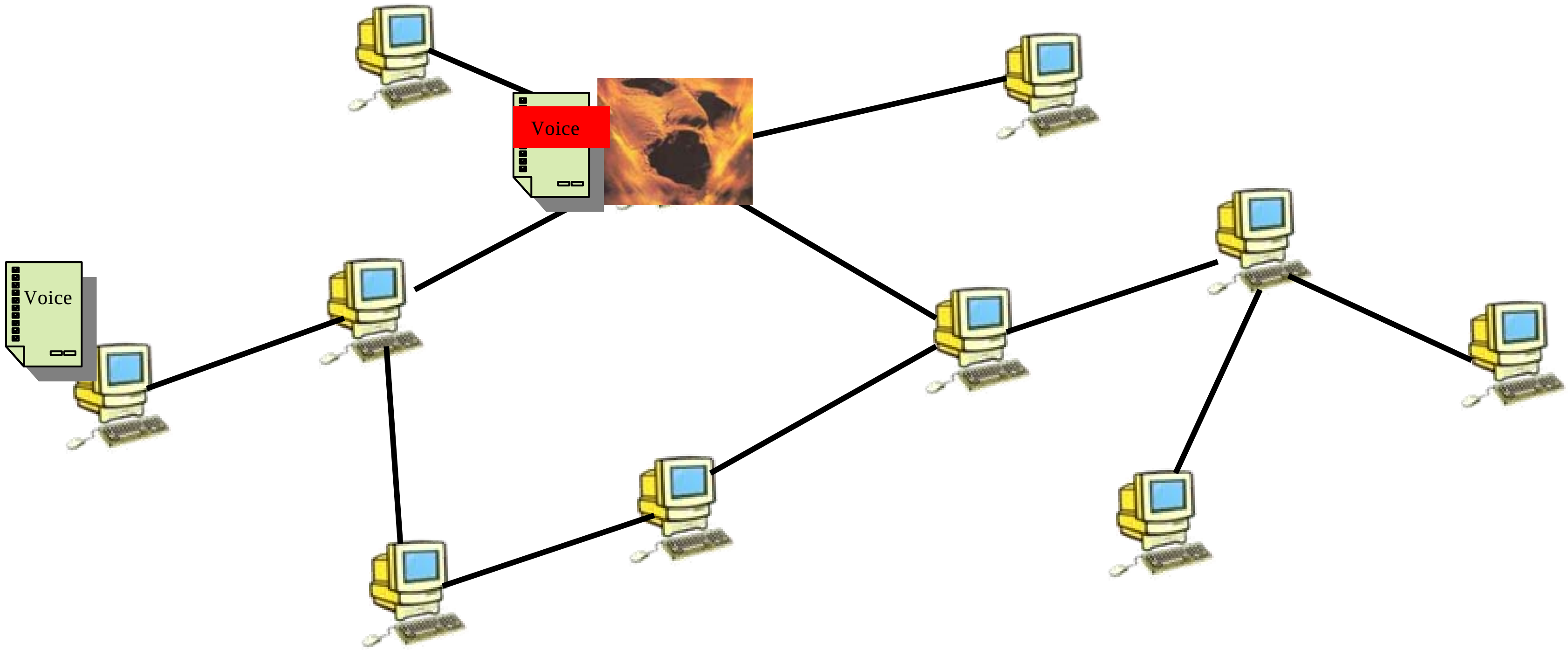
Advantages of P2P (3/3)

- **Robustness and Reliability**
  - It could work without centralized server.
  - Increased Network Connectivity



Issues of P2P (1/2)

- **Security**
  - Integrity and authenticity.
  - Privacy and confidentiality.



Issues of P2P (2/2)

- **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

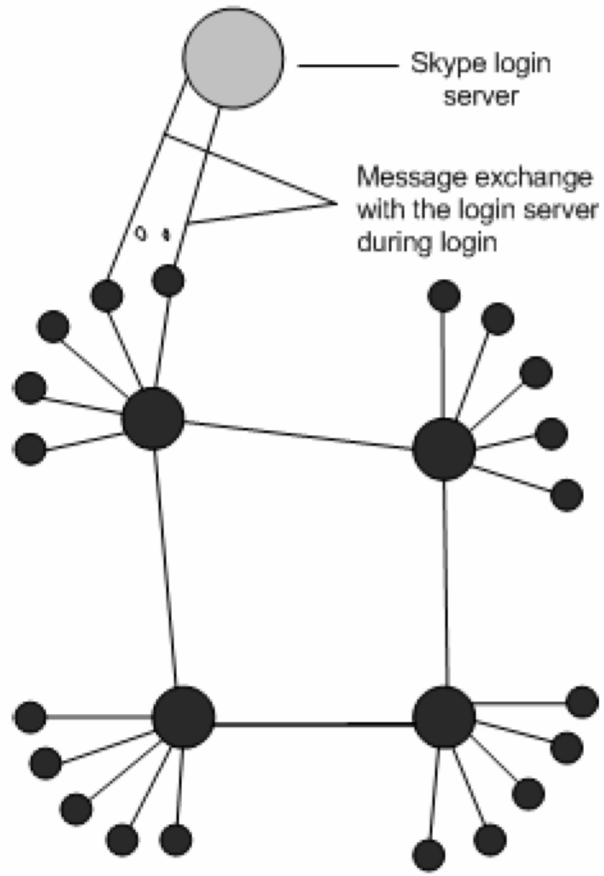
An Example of Voice over Overlay Network

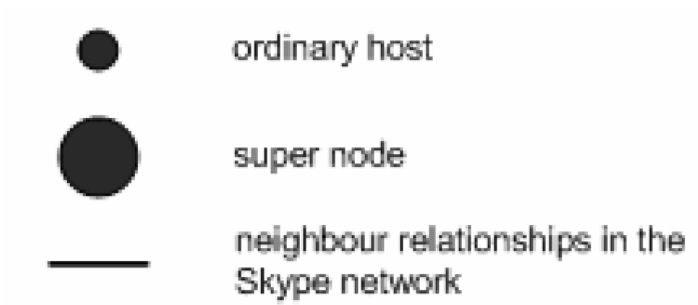
Introduction

- **Skype is a peer-to-peer VoIP client developed by KaZaa in 2003**
- **Skype claims that**
  - It can work almost seamlessly across NATs and firewalls
  - It has better voice quality than the MSN and Yahoo IM applications
- **The key Skype functions include**
  - Login
  - NAT and firewall traversal
  - Call establishment and teardown
  - Media transfer
  - Codecs
  - Conferencing

Skype Network

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





Key Components of Skype Software [1/2]

- **Ports**
  - SC opens a TCP and an UDP listening port
  - SC also opens port 80 (HTTP) and port 443 (HTTPS)
  - There is no default TCP or UDP listening port
- **Host Cache (HC)**
  - The HC is a list of super node IP:Port pairs
  - A SC stores HC in the Windows registry at HKEY_CURRENT_USER / SOFTWARE / SKYPE / PHONE / LIB / CONNECTION / HOSTCACHE
  - HC contains a maximum of 200 entries
- **Codecs**
  - The white paper observes that Skype uses **iLBC**, **iSAC**, or a third unknown codec
  - Skype codecs allow frequency between 50-8000 Hz to pass through

Key Components of Skype Software [2/2]

- **Buddy List**
  - Skype stores its buddy information in the Windows registry
    - Digitally signed and encrypted
  - The buddy list is local to one machine and is not stored on a central server
- **Encryption**
  - Skype uses AES (Advanced Encryption Standard)
    - 256-bit key (1.1×10^{77} possible keys)
  - Skype uses 1536 to 2048 bit RSA to negotiate symmetric AES keys

Experimental Setup

- **Version 0.97.0.6**
  - Latest version 1.0.0.106
- **Under three different network setups**
  1) Both Skype users were on machines with public IP address
  2) One Skype user was behind port-restricted NAT
  3) Both Skype users were behind port-restricted NAT and UDP-restricted firewall
- **Ethereal was used to monitor network traffic**
- **NetPeeker was used to tune the bandwidth**

Skype Functions

- **Startup**
  - When SC was run for the first time after installation
    - sent a HTTP 1.1 GET request (contains the keyword "installed") to the Skype server
  - During subsequent startups
    - a SC only sent a HTTP 1.1 GET request to determine if a new version is available
- **Login**
- **User Search**
- **Call Establishment and Teardown**
- **Media Transfer and Codec**
- **Keep-alive Messages**
  - The SC sent a refresh message to its SN over TCP every 60s

Login

- **Login is perhaps the most critical function to the Skype operation**
- **During this process, a 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

Login Server and Bootstrap Super Nodes

- **Login Server**
  - The only central component in the Skype network
  - IP address: 80.160.91.11
    - ns14.inet.tele.dk and ns15.inet.tele.dk
- **Bootstrap Super Nodes**
  - HC was initialized with 7 IP:Port pairs
  - Bootstrap SNs are connected to the Internet through 4 ISPs
  - If the HC was flushed after the first login, SC was unable to connect to the Skype Network

First-time Login Process [1/2]

- **There are only 7 entries in the SC host cache upon installation**
- **A SC must connect to well known Skype nodes in order to log on to the Skype Network**
  - By sending UDP packets to some bootstrap SNs and then wait for their response
    - It is not clear how SC selects among bootstrap SNs to send UDP packets to
  - SC then established a TCP connection with the bootstrap SN that responded

First-time Login Process [2/2]

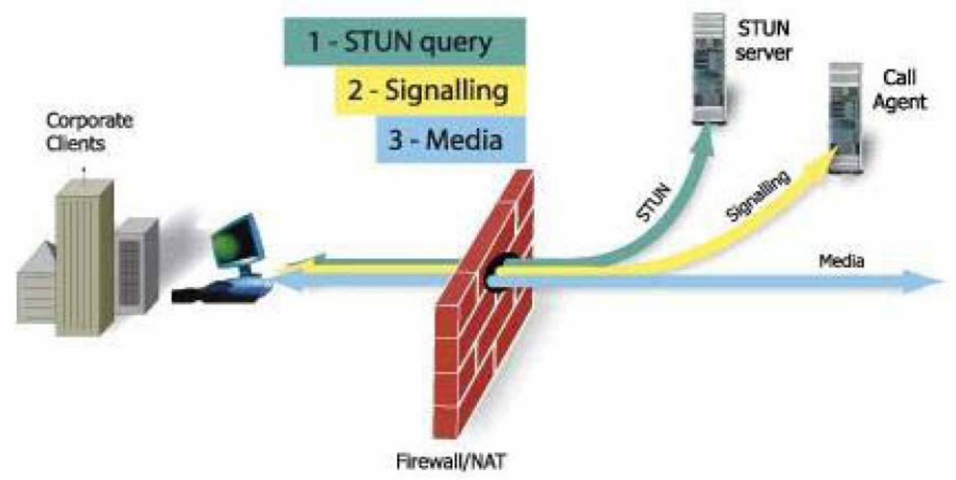
- **A SC running on a machine with public IP address**
  - Exchange some packets with SN over TCP
  - Then establishes a TCP connection with the login server
  - The TCP connection with the SN persisted as long as SN was alive
  - The total data is about 9k bytes
- **A SC behind a port-restricted NAT**
  - Roughly the same as for a SC on a public IP address
  - The total data is about 10k bytes
- **A SC behind a port-restricted NAT and UDP-restricted firewall**
  - Unable to receive any UDP packets from machines outside the firewall
  - It exchanged 8.5k bytes of data

NAT and Firewall Determination

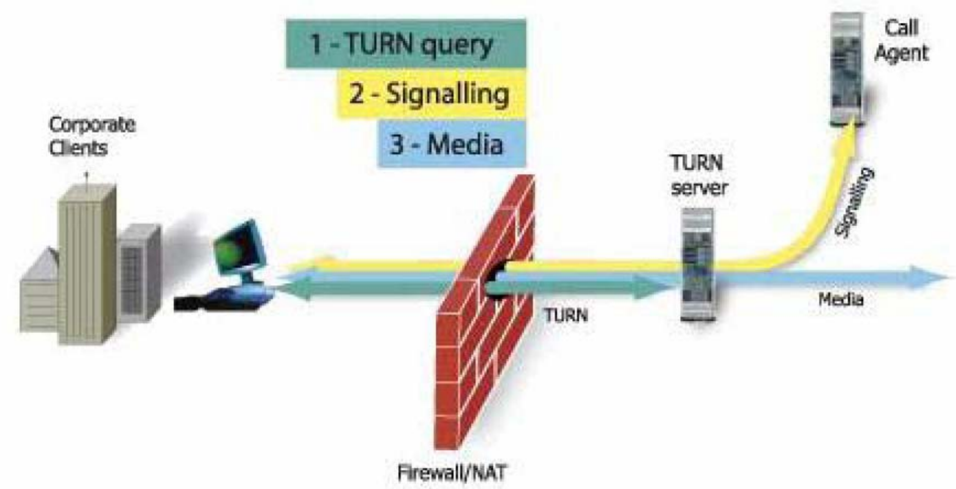
- **The authors conjecture that a SC is able to determine at login if it is behind a NAT and firewall**
  - By exchanging messages with **its SN** or **some nodes** using a variant of the STUN protocol
- **Once determined, the SC stores this information in the Windows registry**
- **SC refreshes this information periodically**

STUN and TURN

- **STUN**
  - Simple Traversal of UDP through NAT
  - Doesn’t work through symmetric NAT



- **TURN**
  - Traversal Using Relay NAT
  - Increase latency
  - Server load



Login Procedures

- **Alternate Node Table**
  - **SC sends UDP packets to about 20 distinct nodes at the end of login process**
    - To advertise its arrival on the network
  - **Upon receiving a response from them, SC builds a table of online nodes**
    - Alternate node table
    - It is with these nodes a SC can connect to, if its SN becomes unavailable
- **Subsequent Login Process**
  - Quite similar to the first-time login process
- **Login Process Time**
  - Scenario (1) and (2): **3-7** seconds
  - Scenario (3): about **34** seconds

User Search

- **Skype uses its Global Index (GI) technology to search for user**
  - A distributed algorithm
  - Guarantee to find a user if it exits and has logged in during the last 72 hours
- **For SC on a public IP address**
  - SC sent a TCP packet to its SN
  - SN gave SC the IP:Port of **4** nodes to query
    - If it could not find the user, it informed the SN over TCP
    - It appears that the SN now asked it to contact **8** different nodes
  - This process continued until the SC found the user or it determined that the user did not exist
  - The search took 3 to 4 seconds
- **Search Result Caching**

Call Establishment and Teardown [1/2]

- **The call signaling is always carried over TCP**
- **For users that are not in the buddy list**
  - Call placement = user search + call signaling
- **Both users were on public IP address**
  - The caller SC established a TCP connection with the callee SC
- **The caller was behind port-restricted NAT and callee was on public IP address**
  - The caller sent 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 Establishment and Teardown [2/2]

- **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**

Media Transfer and Codecs [1/2]

- **The total uplink and downlink bandwidth used for voice traffic is 5k bytes/s**
  - This bandwidth usage corresponds with the Skype claim of 3k-16k bytes/s
- **No silence suppression is supported in Skype**
  - It maintains the UDP bindings at NAT
  - These packets can be used to play some background noise at the peer
- **Skype allows peers to hold a call**
  - To ensure UDP binding, a SC sends three UDP packets per second to the call peer on average

Media Transfer and Codecs [2/2]

- **Codec Frequency Range**
  - The min. and max. audible frequency Skype codecs allow to pass through are 50 Hz and 8000 Hz
- **Congestion**
  - 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

Conferencing

- A acts as a mixer, mixing its own packets with those of B and sending to C and vice versa
  - The most powerful machine will be elected as conference host and mixer
- Two-way call: 36k bytes/s
- Three-user conference: 54k bytes/s