An Analysis of skype P2P Internet Telephony Protocol

Presented by

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Overview

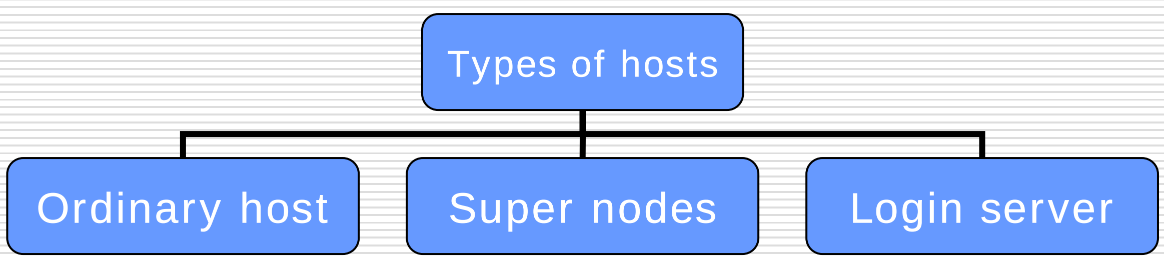
- Introduction to Skype
- P2P architecture
- Key components of Skype
- Skype functions
- Conferencing
- Conclusion

Introduction to Skype

Based on peer-to-peer network

- Minimal network infrastructure
- Utilize its users' computers to do the work

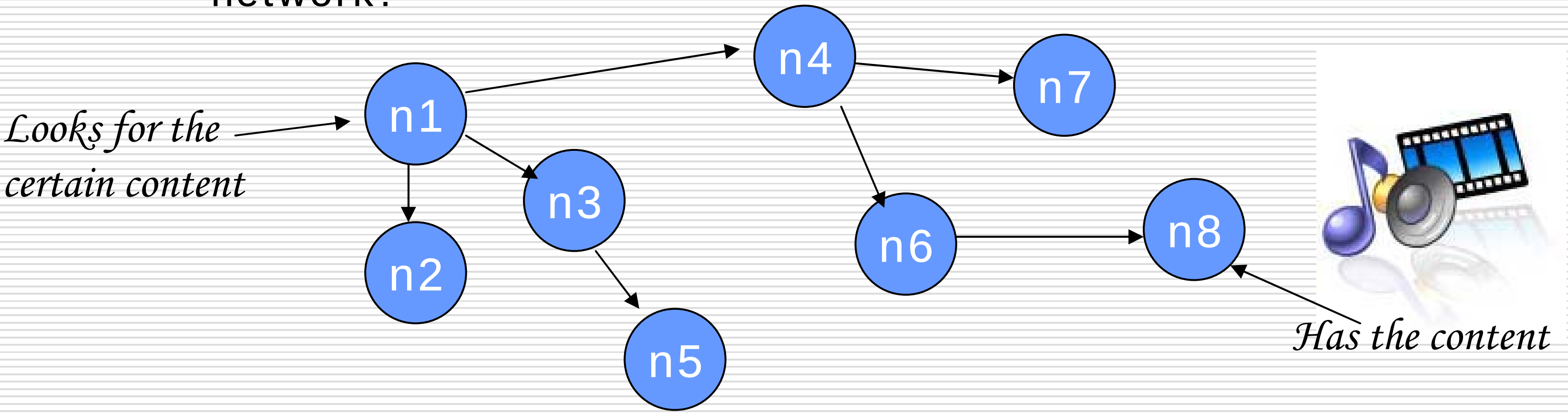
- Claims to have implemented a Global Index Technology
- Uses Wide band codecs
- It is a proprietary protocol in contrast to SIP and H.323



P2P Introduction

- Peer-to-Peer (P2P) is a communications model in which
  – each communication node (peer) has both server and client capabilities
  – either party can initiate a communication session
  – applications connect with each other directly

- How to find specific peer that hosts desired data within decentralized network?



Skype Network

- Skypehas a similar architecture as its predecessor KaZaA
- It uses TCP for signaling and both UDP and TCP for transporting media traffic
- Any node with a public IP address having sufficient CPU,memory and network band width is a candidate to become a super node
- An ordinary host must connect to a super node and must register itself with the Skype login server

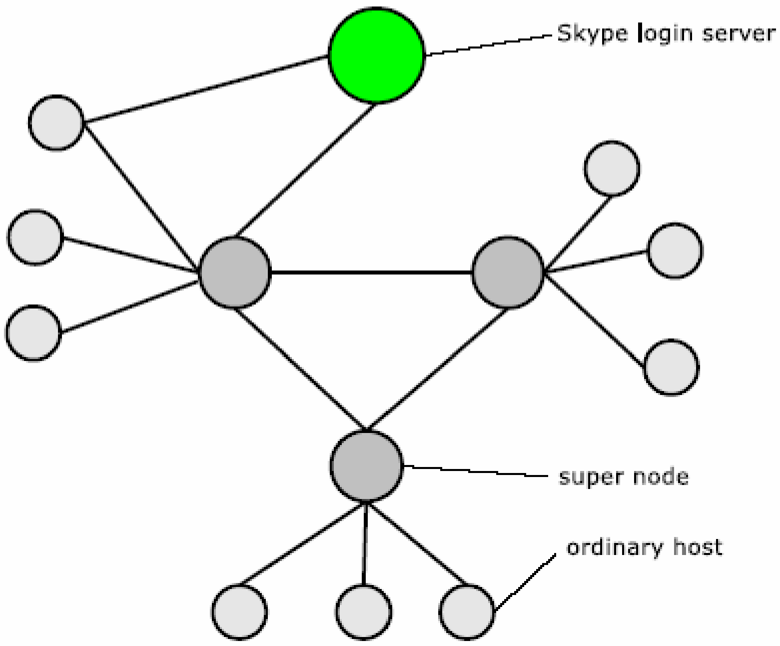


Figure 1: The Skype Network[1]

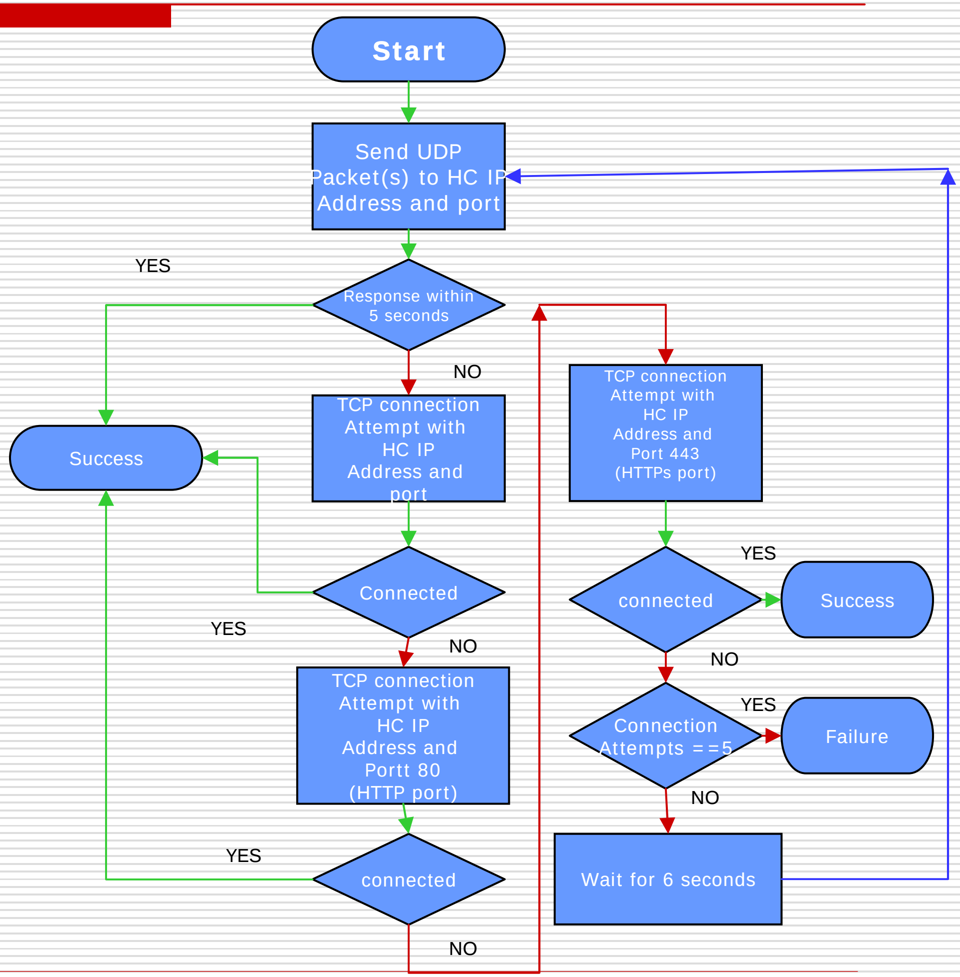
Key Components of Skype

Ports	A Skype client (SC) opens a TCP and a UDP listening port configured in its connection dialog box
Host Cache (HC)	A list of super node IP address and port pairs that SC builds and refreshes regularly A SC stores HC in the Windows registry
Codecs	A wideband codec allowing frequencies between 50-8K Hz, which is Implemented by Global IP Sound
Buddy List	Skype stores buddy information in Windows registry Buddy list is digitally signed and encrypted, local to machine and not on a central server
Encryption	Skype uses 256-bit AES encryption
NAT and Firewall	SC uses a variation of the STUN and TURN protocols to determine the type of NAT and firewall

Skype functions

- Startup
  - HTTP 1.1 GET
    - Installed
    - Get latest version
- Skype login algorithm
  - Only one entry is present in host cache
- Login server
  - SC exchanged data over TCP with a node of IP address **80.160.91.11**
  - Reverse lookup

ns14.inet.tele.dk



Bootstrap super node (SN)

- After logging in for the first time after installation, HC was initialized with 7 IP addresses and port pairs

IP address:port	Reverse lookup result
66.235.180.9:33033	sls-cb10p6.dca2.superb.net
66.235.181.9:33033	ip9.181.susc.suscom.net
80.161.91.25:33033	0x50a15b19.boanxx15.adsl-dhcp.tele.dk
80.160.91.12:33033	0x50a15b0c.albnxx9.adsl-dhcp.tele.dk
64.246.49.60:33033	rs-64-246-49-60.ev1.net
64.246.49.61:33033	rs-64-246-49-61.ev1.net
64.246.48.23:33033	ns2.ev1.net

- Upon first login, the SC sent UDP packets to at least four nodes in the bootstrap node list

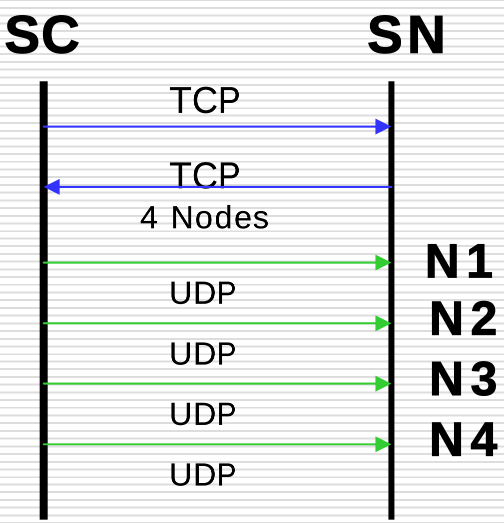
- Bootstrap IP address and port information is hard coded in the SC

First-time login process

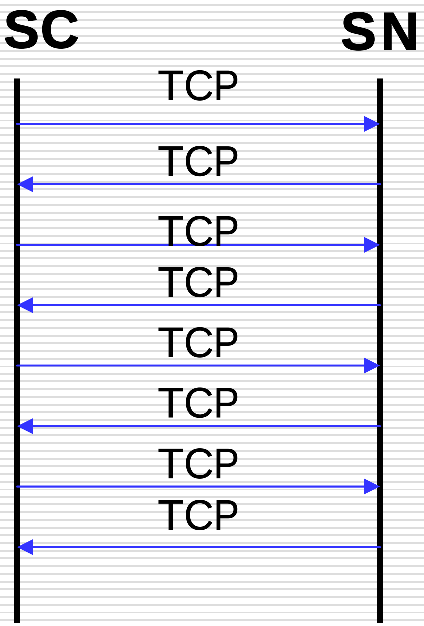
- A SC must connect to well known Skype nodes in order to log on to the Skype network

- **Step 1** : Sends UDP packets to some bootstrap super nodes.
- **Step 2** : Establishes a TCP connection with the bootstrap super node.
- **Step 3** : Acquiring the address of the login server.
- **Step 4** : Establishing a TCP connection with the login server.
- **Step 5** : Advertise its arrival.

User search

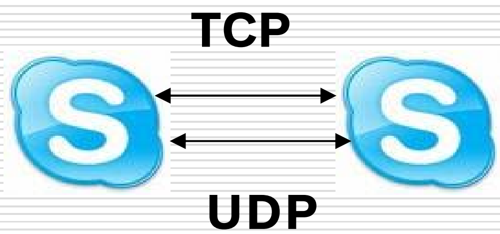


Skype client in public address



Skype behind a port-restricted NAT and UDP-Restricted firewall

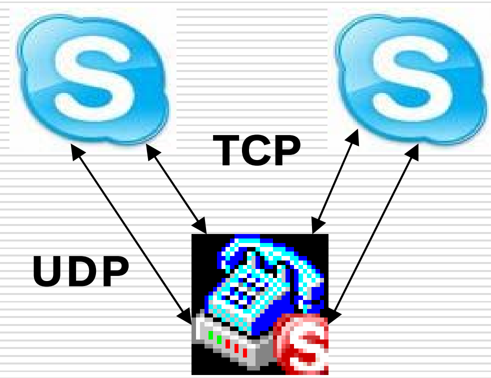
Call Establishment



Public IP Address

Signalling	TCP
Media	UDP
Packet payload size	67 bytes

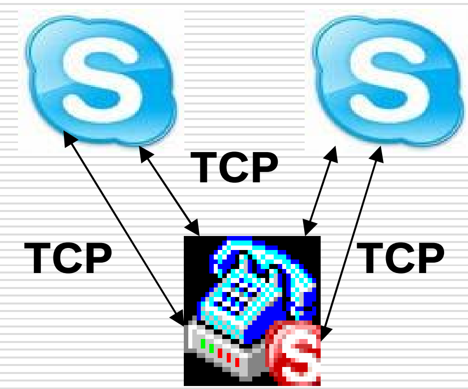
Behind port restricted NAT



Online Skype node

Signalling	TCP
Media	UDP
Packet payload size	67 bytes

Behind port-restricted NAT UDP- restricted firewall

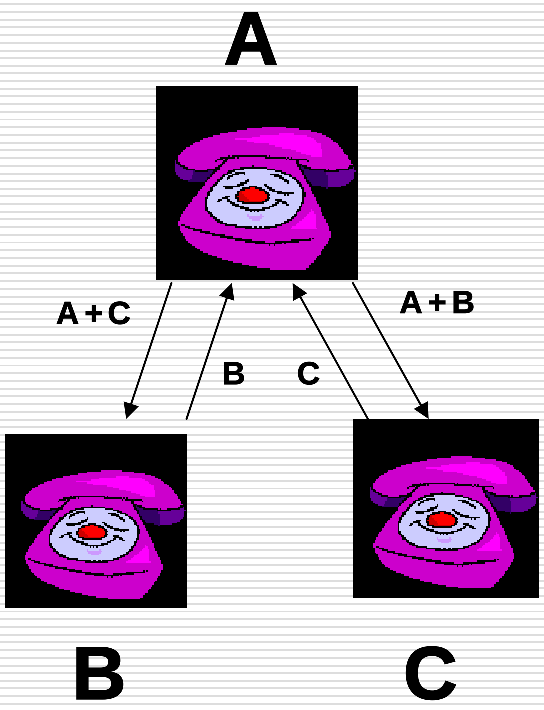


Online Skype node

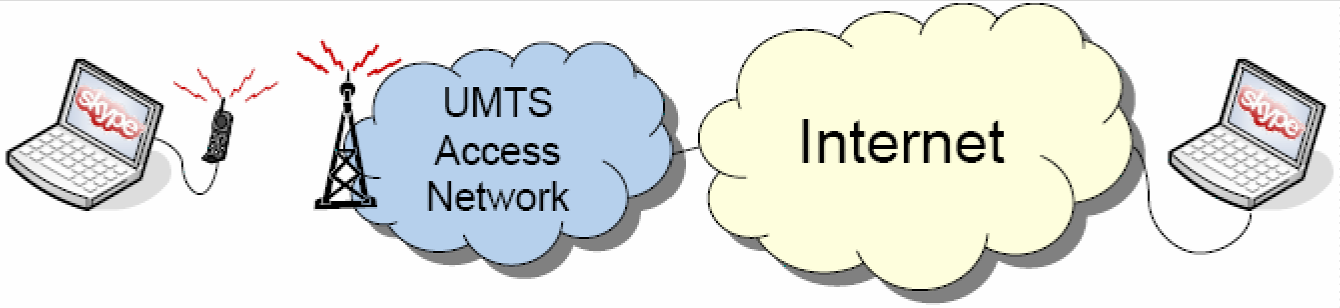
Signalling	TCP
Media	TCP
Packet payload size	69 bytes

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
- **Setup**
  - **A ,B ,C in public internet**
  - **B , C behind port restricted NAT and A on public internet**
  - **B , C behind port restricted NAT and UDP restricted firewall**



Skype over UMTS



- Mobile Voip as inexpensive alternate for voice calls
- UMTS sufficient to make mobile voip calls with skype possible
- Based on **experienced end-to-end quality** Skype implements
  - dynamic QoS adaptation onto environment
  - application-driven re-routing
- Connection relayed if ...
  - packet loss too high (>25%)
  - round trip time too high (>4s)

Unsolved issues

- In the first time login process skype sends ICMP messages to some nodes in the network . The reason is not clear

- It is not clear how SC terminates search if it is unable to find an user

- At login SC determines if it is behind a NAT and firewall . Once determined it stores the information in windows registry and it refreshes this information periodically. We are not clear how SC refreshes the information

Conclusion

- Skype is the first VoIP client based on peer-to-peer technology. We think that three factors are responsible for its increasing popularity.

- 1. Better voice quality than MSN and Yahoo IM clients.

- 2. Work almost seamlessly behind NATs and firewalls.

- 3. Extremely easy to install and use.

References

An Analysis of the skype PeertoPeer Internet Telephony Protocol Salman A. Baset and Henning Schulzrinne

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Thank you