Advanced Networking

P2P Voice Applications

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Credits for part of the original material to Saverio Niccolini NEC Heidelberg

The Client/Server model in conversationsl communications

- User-plan communication is nearly always in direct mode, i.e., logically it is P2P
- C/S is used in traditional telephone networks (also over IP) for look-up and signaling
- This requires a very expensive network of servers and resources …
- … it's just like calling always a 12** number instead of looking in your address book!

From POTS to VoP2P: Step 1

- H.323
  - Tries to reproduce the traditional telephony over a packet, IP-based network
  - Adds services that are not conceivable in traditional telephony (e.g. voice-web integration)

From POTS to VoP2P: Step 2

- SIP
  - Has the standard "internet" philosophy
  - Move service logic and intelligence in terminals
  - Distributed and flexible (indeed, SIP goes far beyond telephony)
  - Security problems

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From POTS to VoP2P: Step 3

- (Nearly) Server-Less systems
  - Are a transition from the C/S model, which is still rooted in H.323 and SIP) to a service model where each terminal tries to be as autonomous as possible with a flexible hierarchy for seracing services and userse (DNS like) … The P2P paradigm is the ending point.
  - Without a service provider P2P systems are open and closed at the same time: anybody can be part of a system, but different systems do not talk one another
  - However a communication service either has a monopoly (protocols, formats, syntax – like IP) or there are gateways to cross different service domains

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VoP2P Standardization

- Many approaches:
  - An interest forum …
  - Pilot products …
  - A Task Force IETF talking to an ITU group talking with …
- But indeed looking into the problem …
  - SIP is compatible with a P2P model
    - Just substitute Registrars and Proxys with distributed data-bases and distributed search engines
  - If the idea is a win-win scenario some idea will be the winner … not necessarily SIPeer

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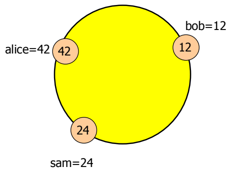
SIPeer

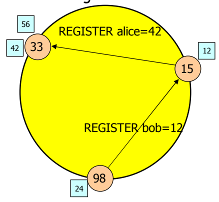
- Goals
  - P2P standardization project based on SIP primitives
  - 0-configuration
  - Audio and messaging
  - Backward compatible sith standard SIP systems
- Use existing DHT (Distributed Hash Tables) systems
  - Key=hash(user@domain)

Users' Search

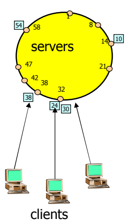
- Without "REGISTER"
  - The key is computed based on the user ID
  - Nodes enter the P2P overlay with their user ID
  - One node ⇔ One user



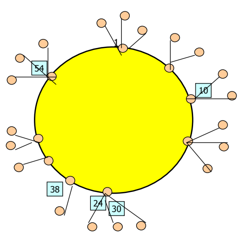
With "REGISTER"
  - The users "REGISTER" with some nodes responsible for its key
  - Periodic refresh
  - Enabled off-line message exchange



Design alternatives



DHTs are located on distributed, low-cost servers



Hierarchical approach: standard, but stable and powerful nodes maintain the DHT – similar to skype

P2P real-time: Users perspective

- Ease of usage
- No user configuration required
- Working across all networking environments
  - Network Address Translators (NATs)
  - Firewalls (FWs)
- P2P real-time applications are not standard-based but they "just work"
- Different user experience with respect to standard-based real-time applications
  - e.g. H.323-based or SIP-based

Identification of issues with P2P SIP

- Goal
  - Identify potential issues of SIP-based P2P communication related to Middleboxes (NAT and firewall) traversal
    - to be considered when designing standards for a SIP-based P2P infrastructure
- Non-Goals
  - Constrain a future P2P SIP architecture in any way
  - Still we need to list potential communication steps that might raise issues
  - Those steps are not necessary part of the final SIP-based P2P solution
  - Suggest NAT traversal methods to be selected for P2P solution

Potential Communication Steps

- Steps considered
  - middlebox detection
  - registration
  - search for relays
  - address lookup
  - call setup
  - call termination
- Not all steps might be necessary
- Several steps may be combined into one

Middlebox Detection

- Detect Middleboxes
  - on the signaling path
  - on the data path
- Communication means detection for
  - registration
  - incoming / outgoing signaling
  - data streaming to and from other terminals or relays
- Checks to be performed
  - sending and receiving UDP packets
  - opening incoming and outgoing TCP connections
  - use of certain fixed port numbers
  - the option to relay or tunnel signaling messages and streamed data
- NAT parameter detection
  - full cone, half cone, etc…

Registration

- Authentication of the user
- Notification of communication capability and willingness
- Registration of contact parameters
- Notification of service provisioning capability and willingness

Further Steps

- Search and Connect Relay
  - Candidate relays may be suggested by infrastructure
- Address Lookup
  - Per-call lookup
  - Buddy list lookup
- Connection Establishment and Termination

Middlebox Traversal Methods

- Tunneling
  - in highly restricted environments only
  - controversial:
    - HTTP and DNS tunneling are not legitimate
    - TURN could be OK
- Network-initiated Middlebox Signaling
  - not the right choice for P2P SIP
- Terminal-initiated Middlebox Signaling
  - several methods known

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Terminal-initiated Middlebox Signaling

- Standards
  - STUN (IETF RFC3489)
  - UPnP (UPnP Forum)
  - SOCKS (IETF RFC 1928)
  - RSIP (IETF RFC 3103)
- Under development
  - STUN update (IETF behave WG)
  - ICE (IETF mmusic WG)
  - NSIS (IETF nsis WG)
- Middlebox traversal using relays
  - STUN relay (previously TURN) (IETF mmusic WG)

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Open Issues for SIP-based P2P

- SIP-unrelated
  - middlebox detection beyond UDP
- SIP-related
  - terminal reachability
  - communication service requirements
  - communication service offers
- The relevance of these issues strongly depends on the choice of P2P architecture

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Middlebox Detection Beyond UDP

- Limited or no middlebox detection for TCP and DCCP (Datagram Congestion Control Protocol) available
  - Middlebox signaling for TCP is covered by UPnP, SOCKS, RSIP, NSIS
- TCP considered for signaling and for data
  - Several SIP-signaled services use TCP
  - RTP over TCP used when UDP is blocked
- Might get solved partially by ICE TCP
  - still in early state

Terminal Reachability

- Relevance depends on registration and relay detection process
- Terminal might need to register first and then find and connect to a relay in order to be reachable
- In between these two steps it would be reachable for signaling but unreachable for data transmission and should be registered as such
- Currently, the SIP protocol does not provide explicit means for signaling such a state

Communication Service Requirement

- The terminal might need to express its needs for relaying
  - signaling messages
  - lookup requests
  - data streams
- Infrastructure nodes might need to suggest relays to be used by terminal
- For both, request and suggestion, signaling means are required
  - Extension Header Field for Service Route Discovery During Registration (RFC 3608) might offer means

Communication Service Offering

- A terminal in an unrestricted (or just slightly restricted) environment might be able (and the user willing) to offer services to other peers, such as relay services and lookup services
- Currently, the SIP protocol does not provide explicit means for signaling such offers

P2P SIP: how to locate peers?

- Basic idea is that what you are looking for has an identifier
  - Locate items in the overlay based on the identifier
  - Distributed Hash Table (DHT), Content Addressable Networks (CAN)
  - Since "everything has its place", eliminate false negatives
  - Since you can go (close to) directly to the item you want, more efficient

Applying this to SIP

- Use pure Distributed Hash Tables (DHT) to find the other UAs
  - Problems
    - currently no DHT standardized
    - some firewalls block DHT traffic as "file sharing"
- Use DHT for location, but implemented as SIP messages
  - Essentially, use DHT as another registration/location mechanism
- Use standard SIP to signal once resources are located

Problems with P2P SIP

- Like most things SIP, NATs
  - Same problems, plus some new ones
  - Super nodes?
- Security
  - Sybil attacks
  - DoS (through traffic and true denial)
  - Encryption
  - Information "leakage"
  - Choosing node locations to divert/block

Advanced Networking

Skype

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Skype characteristics

- Skype is a well known P2P program for real time communications
  - Voice calls
  - Video (from version 2.0)
  - File sharing and instant messaging when in a call
- Seems to work with no problems in all network conditions compared to similar P2P applications
- One of the reasons of its success is its ability to work in network scenarios with middleboxes
  - such as firewalls and Network Address Translators (NATs)
  - usually, this is a problem for P2P applications

How Skype works

- Skype overlay network
  - network structure
  - entities involved
- Skype function analysis
- Lesson learned
- Skype security analysis
  - Binary
  - Network protocol
  - Skype authentication
  - Traffic encryption

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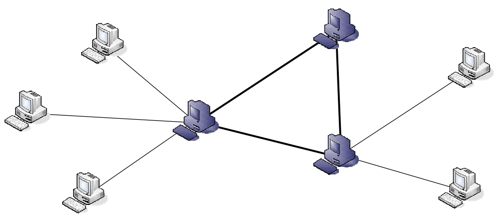
Skype overlay network (I)

- Skype network relies on distributed nodes:
  - Skype Clients (SCs)
  - Supernodes (SNs)



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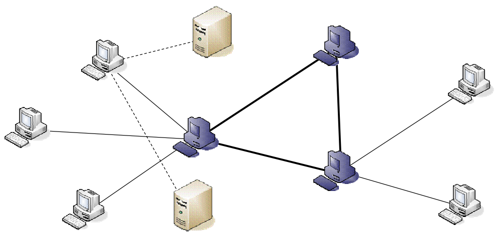
Skype overlay network (II)

- Although there are also centralized entities:
  - HTTP Server
  - Login Server



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Skype overlay network (III)

Skype Client
- used to place voice calls and send instant messages
- connection to skype network possible through a supernode (SN)
- connection with the SN (via TCP) maintained for the whole time the client is on-line
- client configuration and SN addresses are stored locally and refreshed periodically to maintain a coherent view of Skype network

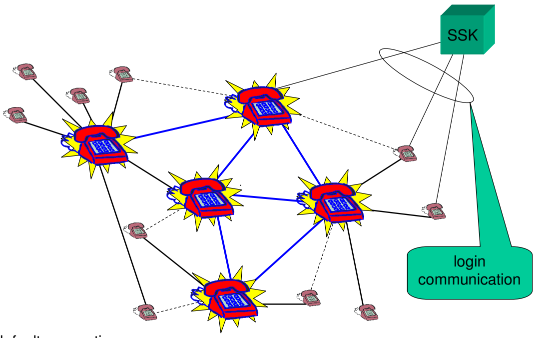
Skype overlay network (IV)

Supernode
- Normal Skype Client that can accepts incoming TCP connections, with enough CPU, memory and bandwidth
- There are also a number of “default” Supernodes, used to increase network robustness and stability

Skype overlay network (V)

Servers
- Login server ensures that names are unique across Skype namespace. Also central point for authentication
- HTTP Server used by clients to check for updates

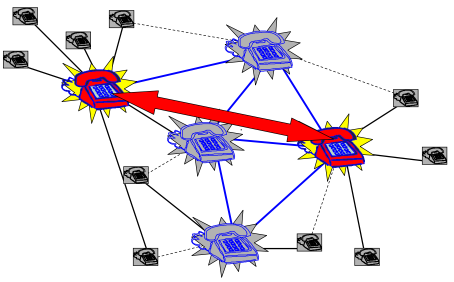
Topology



—— default connection

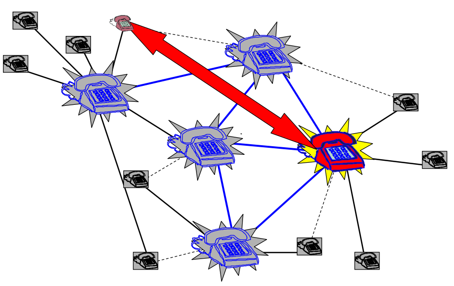
Topology: calls

Supernodes communicate directly ...



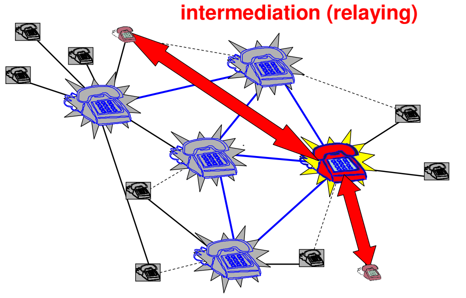
Topology: calls

... also with normal peers



Topology: calls

normal nodes require a supernode intermediation (relaying)



Some caratteristics

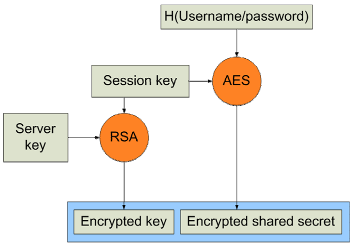
- CODECs
  - Default is a wideband (8 kHz-16kHz sampling) resulting in a transmission rate of 40 kbit/s in each direction (140 pck/s with payload of 67 bytes)
  - Quality in normal conditions is very good, much better than PCM telephony
  - No narrowband coding is provided, congestion is not considered a problem generated by skype
  - Under lab conditions over UDP the system works well even with only 16--20 kbit/s; below 12 kbit/s the system cannot work

Some Characteristics

- Ports
  - 80 (HTTP) e 443 (HTTPS) on TCP for signalin, random choice on UDP or TCP for voice
  - Ports are announced on the P2P network
- Encryption
  - All communications are AES (Advanced Encryption Standard) encoded

Skype Encryption

- Authentication
  - At login time the client generate a RSA session key and uses it to encrypt his credentials.
  - Then encrypts the session key using the server's public key
  - and sends this information to the login server



Some Characteristics

- Host Cache
  - List of supernodes (IP, Port) used to make the search phase faster
  - Roughly 200 entries dynamically updated
  - If the host cache is void skype does not work (some defaults entry are there from the beginning)
  - Une of the critical points for skype functioning
  - The idea is not new to P2P networks and answer to the bootstrap problem … albeit in a naive way

Skype functions analysis

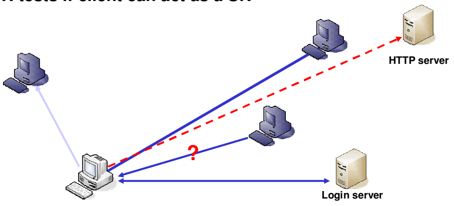
- Essentials
  - Login
  - Search
  - Buddy list signaling
  - Call establishment

Login function

- Join and maintain overlay network:
  - Interaction with central servers
    - login server manage authentication and ensures unique names
    - HTTP server ensures client software updates
  - Refresh of shared.xml
    - file stored on the client containing SNs list and parameters identifying middlebox
  - Network tests if joining client can act as a SN

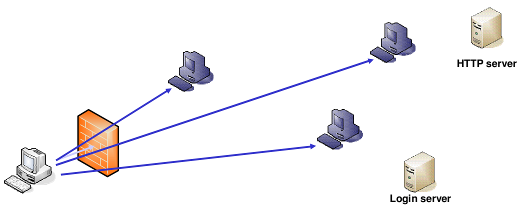
Login procedure

- At startup the client contacts the HTTP server to check for updates
- Sends UDP datagram to a -default SN- to refresh the list of supernodes
- Connects via TCP to a SN (connection maintained throughout Skype session) and exchanges info on online nodes
- Verify username and password via TCP with the Login server
- Another SN tests if client can act as a SN



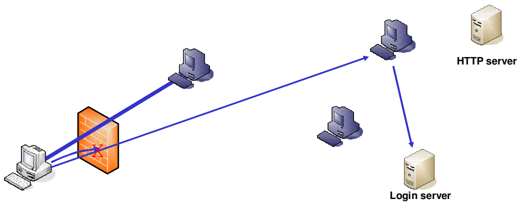
Login: Firewall blocks UDP

- Firewall prevents UDP exchange for SN list refreshing
- Client establishes several TCP connections with SNs to gather information, when finished all but one are torn down



Login: Firewall blocks Login sever

- After connection with the SN, attempt to connect with the Login server fails
- Client connect to the Login using a SN as a relay

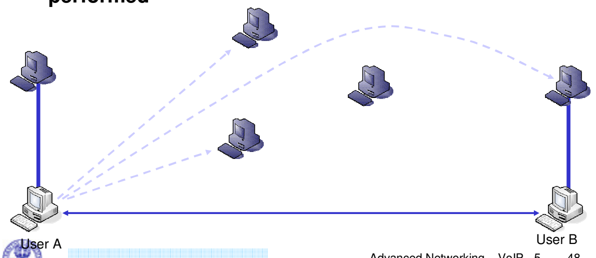


Search function

- Procedure performed when a user wants to add someone to his buddy's list and communicate for the first time
- Search is performed using username as key
  - possible since names are unique
  - this is why there is the need for central servers

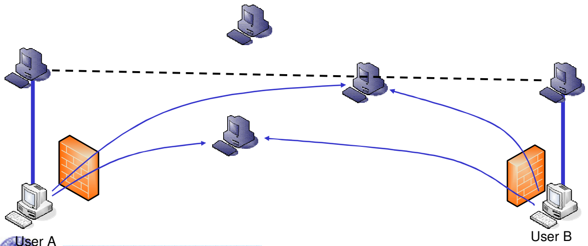
Search procedure

- User A exchanges info with its SN and gather 3 SNs addresses
- A query the 3 SNs via UDP asking if they know the public IP of B
- Once A gets the address of B authorization exchange is performed



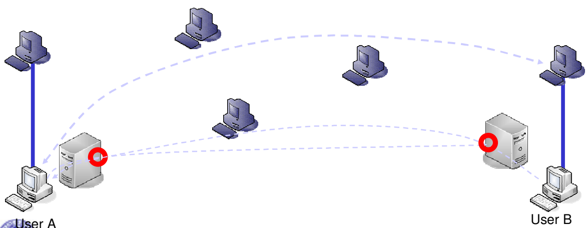
Search: Firewall blocks UDP

- Firewall blocks UDP
  - preventing direct connection w/ the SNs or another user
  - the SN of A communicate to B (via his SN) the address of A
- Both A and B establish TCP connections with the same 2 SN to exchange authorization



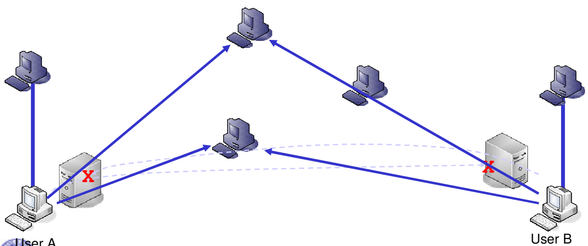
Search: Port restricted NAT

- Once user A gather the address of SN of B, sends a UDP query containing his external address. SN of B replies with user B external address.
- User A send an UDP datagram to user B external address in order to create a mapping in his NAT, anyway packet will be filtered by NAT of B
- User B does the same but this datagram reaches user A
- Once exchanged authorization a TCP connection via 2 SNs as relay is established, as depicted in previous slide



Search: Symmetric NAT

- Clients try the technique depicted for Port restricted NAT
  - but it fails due to symmetric NAT behvior
- Clients exchange authorization via TCP using 2 SNs as relay

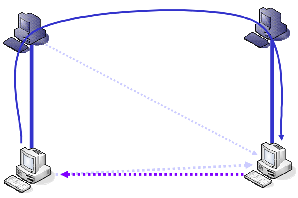


Buddy list signaling

- Buddy list is a list of “friend” users
- Skype allow a user to know if buddies are online/offline
  - overlay network informs buddies when user change status

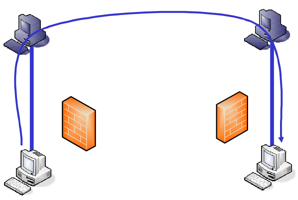
Buddy List signaling procedure

- A user going on-line informs his buddies either directly using UDP or via the SNs.
- When going off-line, a user tear down the TCP connection with the SN.
- The SN informs via UDP the buddies that the user is going off-line
- To have a confirmation buddies try to ping the user.



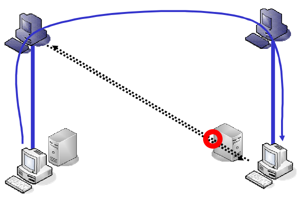
Buddy List signalling: Firewall blocking UDP

- Since UDP traffic is blocked, on-line/off-line signalling is performed via the SNs



Buddy List signaling: Port restricted NAT

- On-line/off-line signaling is performed in a way similar to that depicted in previous slide.
- As a difference after the change of status, buddies query the SN of the user for confirmation.

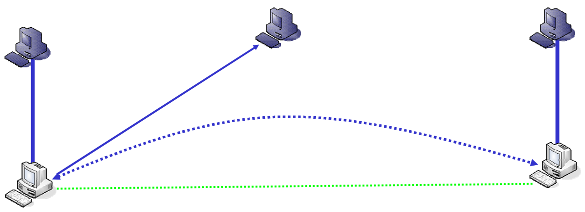


Call establishment function

- Signaling performed using TCP connection
  - overlay network used only if otherwise impossible
- Media carried over UDP when possible
  - in case relay servers are used

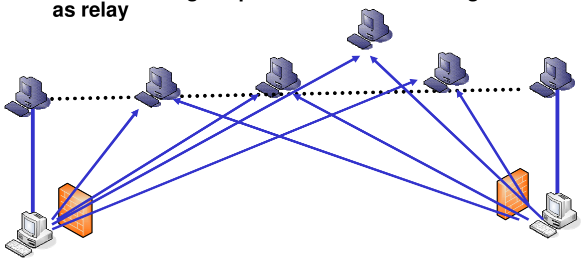
Call establishment procedure

- User A wants to call user B, so he query some SNs for user B address.
- Once he gets user B address they exchange signaling over TCP
- Voice traffic carried via UDP



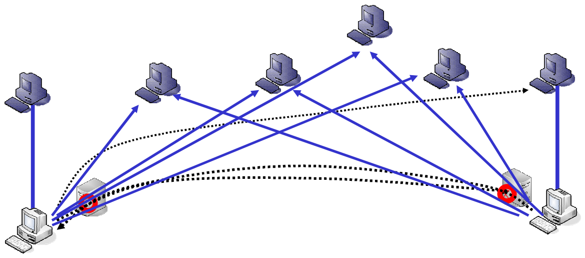
Call establishment: firewall blocks UDP

- Signaling exchanges are performed by the SNs on behalf of the users
- Media exchange is performed via TCP using 4 SNs as relay



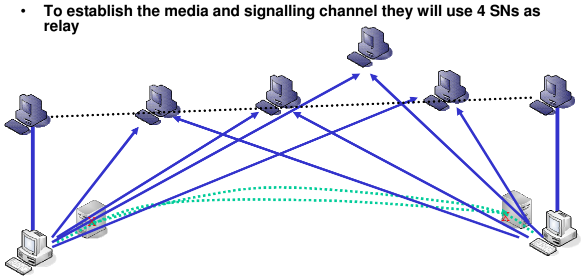
Call establishment: Port restricted NAT

- Once User A gets the address of the SN responsible for user B he queries for his address. SN informs B that user A wants to call him, and tells external address of B to A.
- A and B establish UDP flow using reverse hole punching
- They also establish TCP connection using 4 SNs as relay



Call establishment: Symmetric NAT

- User A and B communicate their addresses via their SNs
- They try reverse hole punching but it won't work because of NATs restrictions
- To establish the media and signalling channel they will use 4 SNs as relay



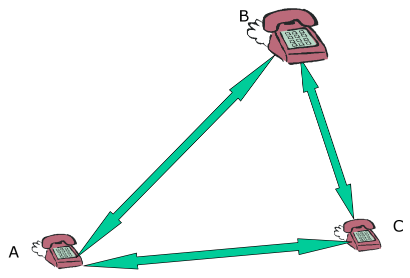
Lesson learned

- Traversal is well possible in many cases without explicit signaling to the middlebox
  - open public access network
  - protected enterprise networks
- Reverse hole punching and tunneling techniques workarounds allow Peer-to-peer communications in almost every scenario
  - Skype only fails completely if firewall blocks TCP but in fact that is a very uncommon case
- Explicit middlebox signaling protocols (like IETF MIDCOM MIB, CheckPoint OPSEC, NEC's SIMCO) are still required for
  - highly protected access network
  - applying security policies by network operator
  - anyway Skype will undermine many of these policies
- Skype tries to use IP network instead of overlay
  - SNs can't assure constant presence
  - avoid overlay congestion

Audio Conference

- Based on traffic mixing in one of the nodes
- Limited to few nodes (5-6)
- Works also with some nodes behind NAT/FW
- The mix node is elected based on it elaboration capabilities, since mixing is CPU intensive
- It does not need to be the conference initiator

Audio Conference: signaling



Audio Conference: audio flows