ΟΙΚΟΝΟΜΙΚΟ ΠΑΝΕΠΙΣΤΗΜΙΟ ΑΘΗΝΩΝ
ATHENS UNIVERSITY OF ECONOMICS AND BUSINESS

Multimedia Technology

Section # 23: Conferencing

Instructor: George Xylomenos

Department: Informatics

Contents

- H.320 and H.324
- H.323
- SIP and SDP
- ICE, STUN and TURN
- Multiparty conferencing

H.320 and H.324

Class: Multimedia Technology, **Section # 23:** Conferencing

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H.320 (1 of 4)

- H.32x conferencing standards series
  - Standardized by the ITU-T
  - Initially: circuit switching (PSTN or ISDN)
  - Later: packet switching (Internet)
  - Limited compatibility between standards
- H.320: conferencing over ISDN
  - Exchange of audio, video, images and text
  - Incorporates other standards

H.320 (2 of 4)

- H.320 components: control plane
  - H.221: Multiplexing of audio and video
    - Based on ISDN B channels (64 kbps)
  - H.230: Multiplexing of sync and control
  - H.231: Multiparty conferencing
  - Q.931: Call establishment
  - H.233/4: Data encryption
  - H.242/3: Negotiation with two/more terminals

H.320 (3 of 4)

- H.320 components: media
  - H.261: Video compression at CIF/QCIF
    - CIF: 352x288, QCIF: 176x144
  - G.711: 3.7 KHz audio at 64 Kbps
  - G.722: 7.5 KHz audio at 64 Kbps
  - G.728: 3.7 KHz audio at 16 Kbps
  - Only G.711 is mandatory

H.320 (4 of 4)

- Bandwidth: p x 64 Kbps
  - Video requires at least p=2 (128 Kbps)
    - Either an ISDN BRI (2 B channels)
    - H.261 with QCIF and low frame rates
  - Ideally, 384 Kbps (6 B channels)
    - 3 BRI lines or part of a PRI line
    - H.261 with CIF and/or higher frame rates

H.324 (1 of 3)

- ISDN should succeed PSTN
  - Making circuits digital
- B-ISDN should succeed ISDN
  - ATM: Cell (tiny packet) switching
- H.320 was extended for ATM
  - H.321: variant for ATM WANs
  - H.322: variant for ATM LANs
  - Little use, like ATM!

H.324 (2 of 3)

- H.324: variant for PSTN (56 Kbps)
  - After it became clear that ISDN was dead
  - Video: H.263 for reduced bitrates
  - Audio: G.723.1 (5.3 and 6.3 Kbps)
  - Alternatively, G.729 (8 Kbps)
- Adaptation of protocols to PSTN
  - PSTN lines are very noisy
  - Their bandwidth can fluctuate

H.324 (3 of 3)

- H.245: negotiation of codecs
  - Can add extra delay to audio
  - Allows changing the bitrates
- H.223: media multiplexing
  - Error detection and recovery
  - Multiple was of multiplexing media
- V.25: call establishment via PSTN modems

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H.323

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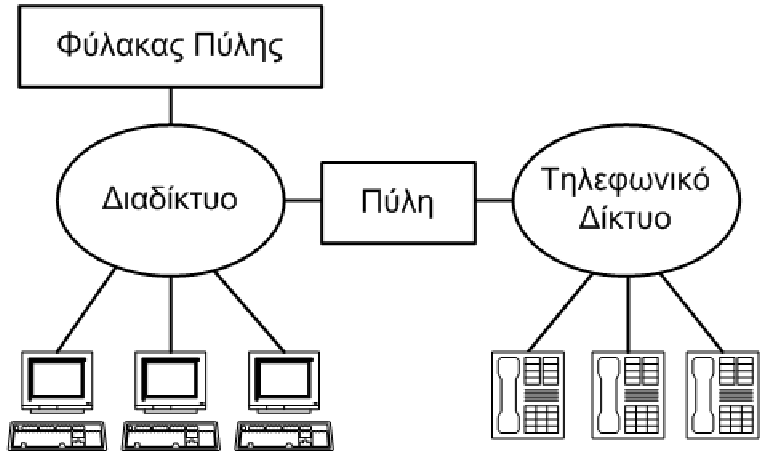
H.323 (1 of 5)

- H.323: Conferencing over the Internet
  - Compatibility between different terminals
- H.323 terminals
  - User devices (hardware)
  - Software applications
- H.323 gateways
  - Interconnection with telephony
    - Analog or digital

H.323 (2 of 5)



- H.323 gatekeepers
  - Conversion between addresses
  - Access control to conferences
  - Bandwidth management
  - Charging and billing

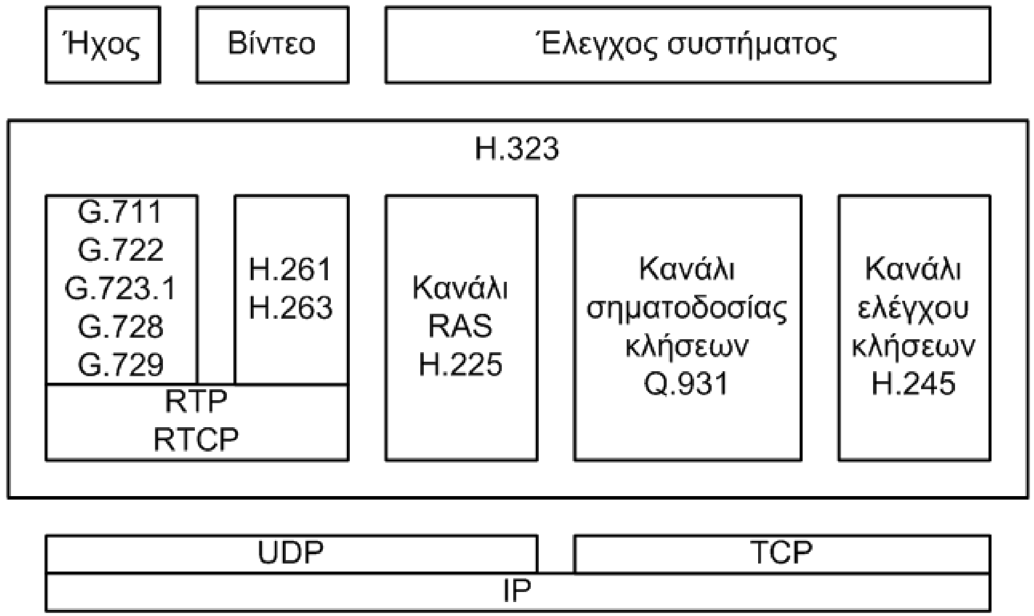
H.323 (3 of 5)

- Required protocols
  - RTP: media transport over UDP
    - RTCP for media control (optional)
  - H.245: codec negotiation
  - Q.931: call establishment
    - Also used in ISDN
  - H.225 (RAS): communication with gatekeeper
    - Registration, admission and state protocol

H.323 (4 of 5)

- Video coding
  - Not required by the standard
    - Cheaper terminals are voice only
  - Minimum (if video supported): H.261
    - Required: QCIF (176x144 luma pixels)
    - Compatibility with older standards
  - Optional: H.263
    - Optional resolutions: CIF, 4CIF and 16CIF

H.323 (5 of 5)



- Audio encoding
  - G.711 is required
  - All other options of H.320/H.324 are optional

H.323 Channels

- Multiple media channels: RTP/UDP
  – One channel per medium (per direction)
- Call signaling channel: Q.931/TCP
  – Dialtones (before dialing)
  – Ringing (after dialing)
- Call control channel: H.245/TCP
  – Exchanges terminal capabilities
  – Opens and closes media channels

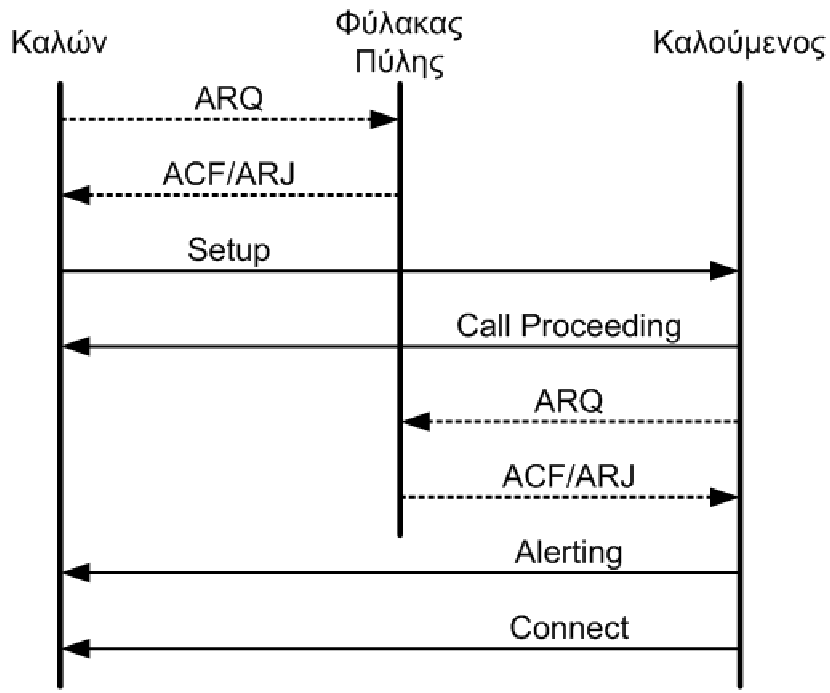
H.323 Gatekeeper (1 of 2)

- The gatekeeper manages a zone
  – Example zone: an organization
  – If there is a gatekeeper, it must be used
  – Checks if calls are allowed
    - Similar to a PBX (private branch exchange)
  – Uses the RAS protocol over TCP
- Bandwidth management
  – Limits the number of concurrent conferences

H.323 Gatekeeper (2 of 2)

- Terminals register to the gatekeeper
  - During initialization (power up)
    - IP address and caller alias
- Address translation
  - Translates aliases to IP addresses
    - Gathered during terminal initialization
    - May require connection with other gatekeepers
- Gatekeeper approves individual calls

H.323 Signaling (1 of 2)



- RAS: connection request (ARQ)
  - May be accepted or rejected (ACF or ARJ)
- Q.931: call setup, call proceeding (routed)

H.323 Signaling (2 of 2)

- Gatekeeper must approve incoming call
- The two gatekeepers may need to talk
  - If they are in different zones
  - This allows translating callee's alias to IP
- Alerting (ringing)
- Call establishment
- Media negotiation follows

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SIP and SDP

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What is SIP? (1 of 2)

- SIP (Session Initiation Protocol)
  - Standardized by the IETF
  - Establishment of multimedia session
  - Codec negotiation
  - Modification of session parameters
  - Session teardown
  - Other protocols used for media exchange

What is SIP? (2 of 2)

- SIP differs from telephony protocols
  - Simple and easy to implement
    - Messages are coded in plain text
  - Covers functionality of Q.931 and RAS
  - Works with other Internet protocols
    - RTP for media transport
    - SDP for media description

SIP Addresses (1 of 2)

- URI (uniform resource identifier)
  - Describes a communication resource
  - Similar to e-mail or web address (URL)
  - Can be used as a hyperlink
- SIP URI for a physical person
  - Location independent
  - Terminal independent
  - sip:name@organization

SIP Addresses (2 of 2)

- Telephone number
  - sip:+302108203693@PSTN-provider
- Group
  - sip:helpdesk@organization
- Media server
  - sip:gameserver@microsoft.com
- All these need to be translated!
- Terminal at fixed IP address
  - sip:225.251.234.1

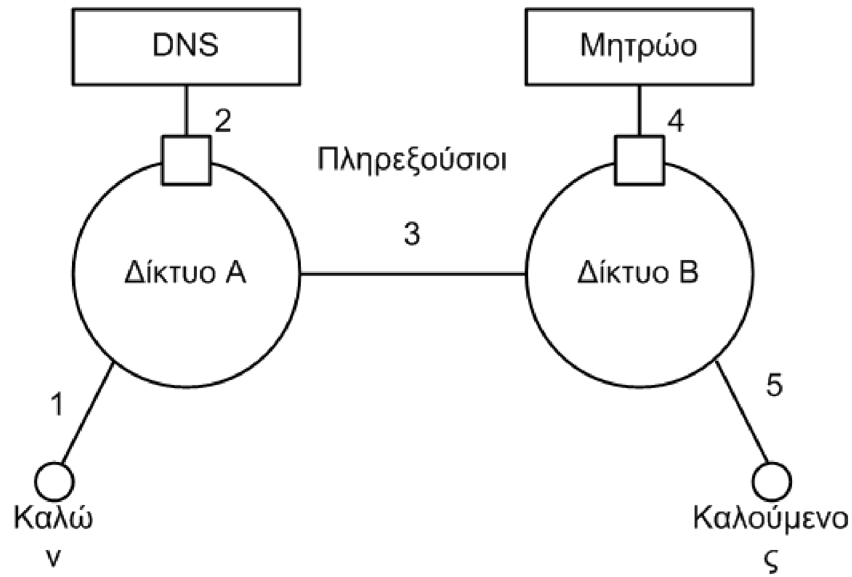
SIP Methods

- Six types of messages (methods)
  - Session establishment: INVITE and ACK
  - Session teardown: BYE
  - Cancelation during establishment: CANCEL
  - Terminal registration: REGISTER
    - Allows address translation
    - Works with dynamic addresses (from DHCP)
  - Information exchange: OPTIONS

SIP Entities

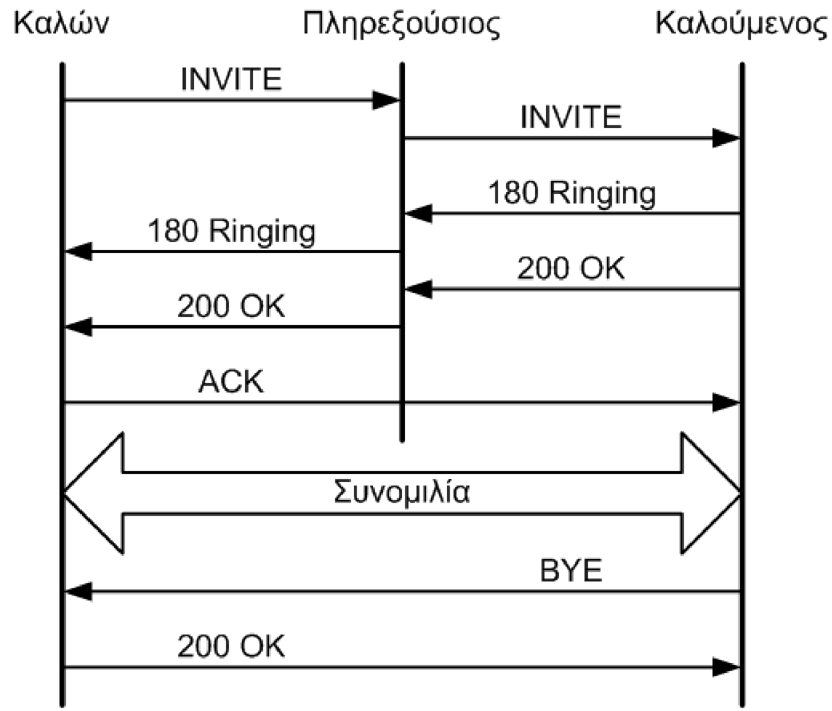
- SIP User Agent
  - The user’s terminal
- SIP Proxy
  - The PBX
  - First point of contact
- SIP registrar
  - May be combined with proxy
- Software or hardware implementation
- Gateways used to talk to telephones

SIP Signaling (1 of 3)



- Call establishment with two parties
  - INVITE with URI of callee sent to caller’s proxy
  - Forwarded to the callee’s proxy
  - Message forwarded to the callee

SIP Signaling (2 of 3)



- INVITE received (180 Ringing)
- Caller picked up (200 OK)
- Call parameters finalized (ACK)

SIP Signaling (3 of 3)

- Message path
  - Initial signaling
    - Goes through the proxies
    - Allows locating the callee
  - Remaining signaling
    - Can be transmitted directly
    - Or via the proxy, to control the call
  - Media are always transmitted directly

SDP (1 of 2)

- SDP (Session Description Protocol)
  - Describes codecs and ports to be used
  - Encapsulated inside SIP messages
- Example: INVITE and OK messages
  - From/To: SIP URI of caller and callee
  - o (originator): user and connection
  - c (connection data): address to receive data
  - m (media description): codec, port, profile

SDP (2 of 2)

Αίτηση	**Απόκριση**
INVITE sip:UserB@there.com SIP2/0	SIP/2.0 200OK
Via:SIP/2.0/UDP here.com:5060	Via:SIP/2.0/UDP here.com:5060
From:sip:UserA@here.com	From:sip:UserA@here.com
To:sip:UserB@there.com	To:sip:UserB@there.com;tag=65a35
Call-ID:12345600@here.com	Call-ID:12345600@here.com
CSeq:1 INVITE	CSeq:1 INVITE
Contact:sip:userA@here.com	Contact:sip:userB@there.com
Content-type:application/sdp	Content-type:application/sdp
v=0	v=0
o=UserA 289084 2890 IN IP here.com	o=UserB 493834 462 IN IP there.com
c=IN IP4 100.101.102.103	c=IN IP4 110.111.112.113
m=audio 49172 RTP/AVP0	m=audio 3456 RTP/AVP0

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ICE, STUN and TURN

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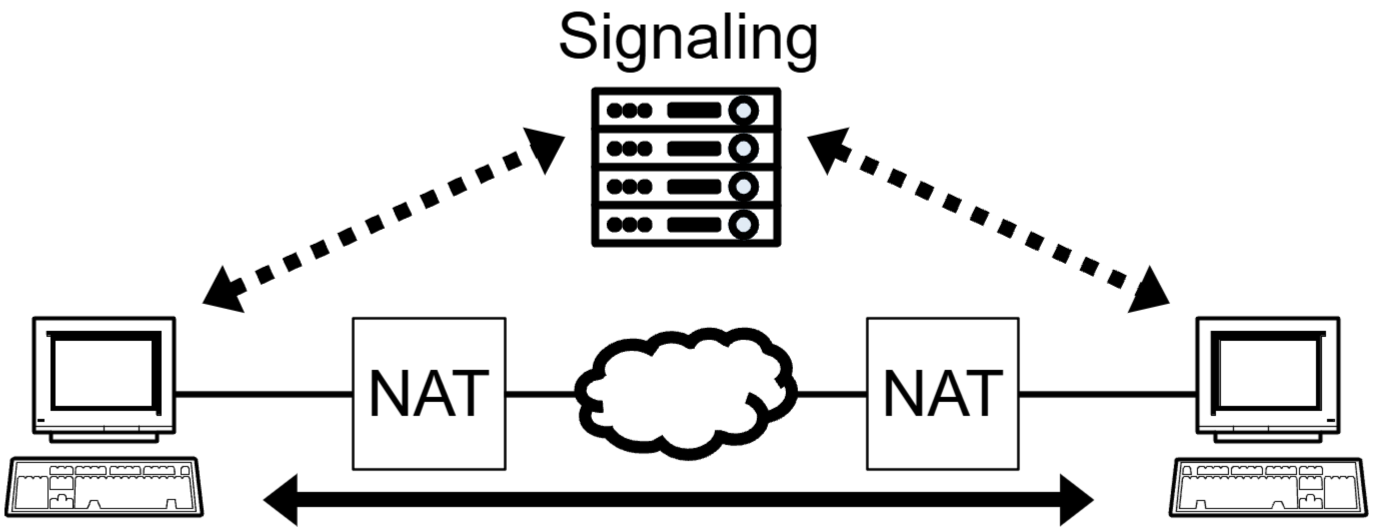
The need for ICE, STUN and TURN

- Conferencing was initially over telephony
  - H.320 and H.324 assumed custom equipment
- Then it moved to the Internet
  - Initially with H.323 and custom equipment
  - Then came SIP and software endpoints
- But this does not work from our home!
  - NATs change public addresses to private
  - Firewalls block most traffic (ports and protocols)

What is ICE? (1 of 2)

- Interactive Connectivity Establishment (ICE)
  - A protocol for NAT traversal
    - Also, for firewall traversal!
  - Each endpoint has an ICE agent
  - The ICE agent discovers ICE candidates
    - Addresses and ports where endpoint is reachable
  - Then it talks to a signaling server
    - Which has a public IP address

What is ICE? (2 of 2)

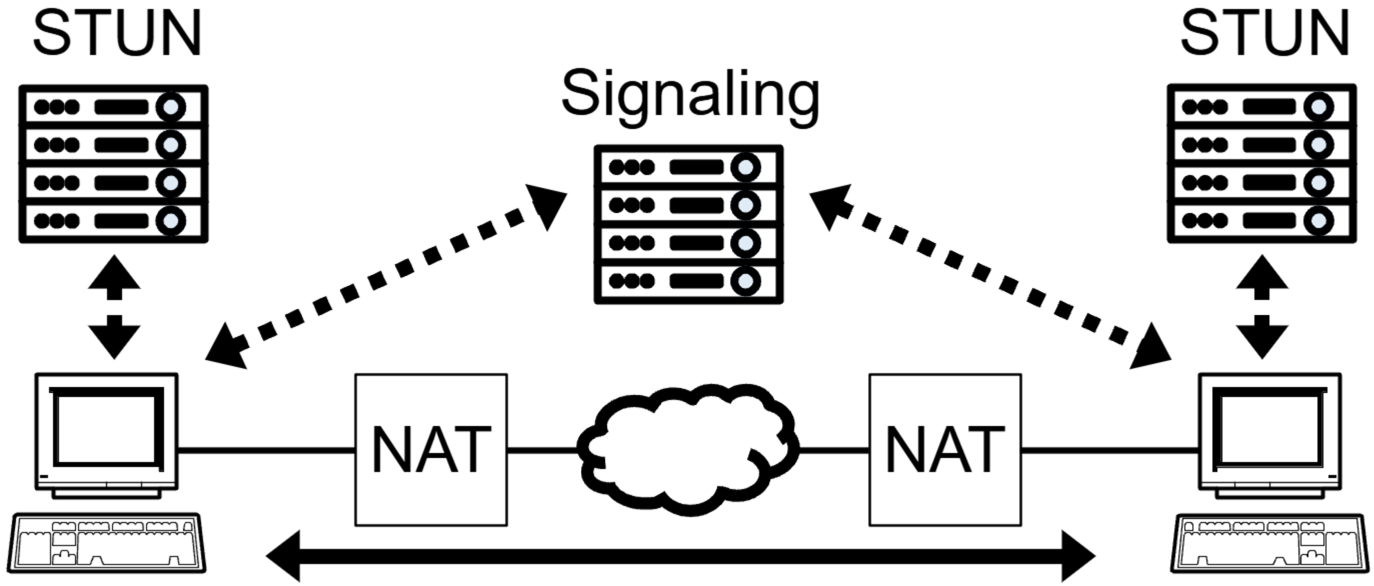


- Signaling can use different protocols
  - For example, SIP can carry SDP and ICE data
- Best case: the hosts have public IPs
  - Then, they can communicate directly

What is STUN? (1 of 2)

- Session Traversal Utilities over NAT (STUN)
  - What if you do not have a public IP?
  - The ICE agent can talk to a STUN server
    - Which has a public IP address
  - The STUN server sends back the public IP
    - As translated by the endpoint's NAT
  - This address is another ICE candidate
  - Which can be exchanged via the signaling server

What is STUN? (2 of 2)

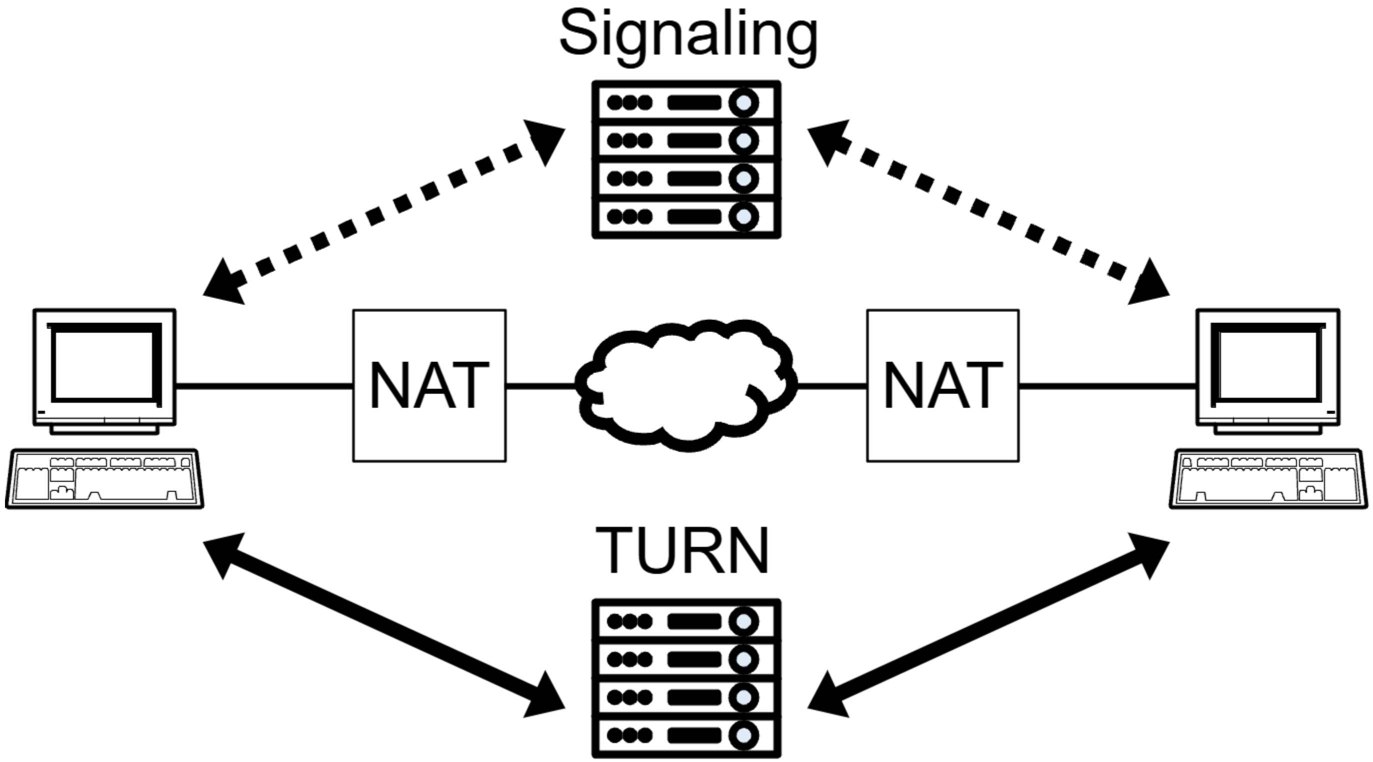


- The endpoints can try the ICE candidates
  - This works for some types of NATs/firewalls
  - If it does, the endpoints can talk directly

What is TURN? (1 of 2)

- Traversal Using Relays around NAT (TURN)
  - What if STUN addresses do not work?
    - This is the case with some NATs (and firewalls)
  - Then we contact a TURN server
    - Which has a public IP address
  - Each endpoint only talks to the TURN server
    - Separate connection / session per endpoint
    - All communication goes through the TURN server

What is TURN? (2 of 2)



- The TURN server is essentially a relay
  - Reachable by both endpoints

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Multiparty conferencing

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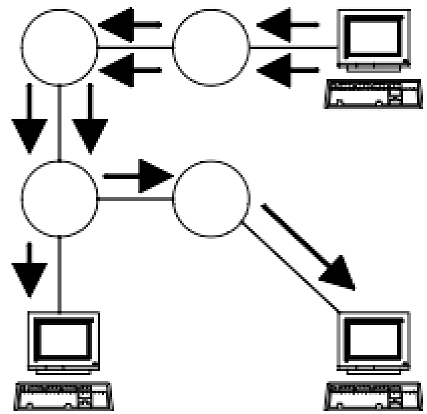
Multiparty conferencing

- How can 3+ endpoints communicate?
  - With 2 endpoints, direct connection
  - Possibly with some intermediates
    - Gateways, gatekeepers, proxies, TURN servers
- Example: conferencing with 3 endpoints
  - With video and audio
  - Every user sees all other users
  - Every uses hears a mix of all other users

Basic topologies

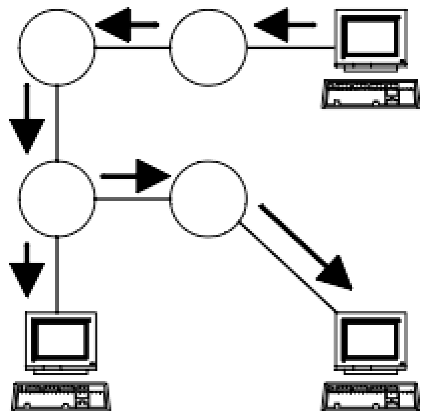
- Peer-to-peer
  - Every user transmits to every other user
    - Ideally, using multicast
  - Low delay but not scalable
- Server-based
  - Everybody connects to a single server
  - The server retransmits media streams
  - Higher delay but only one stream sent

Peer-to-peer (1 of 2)



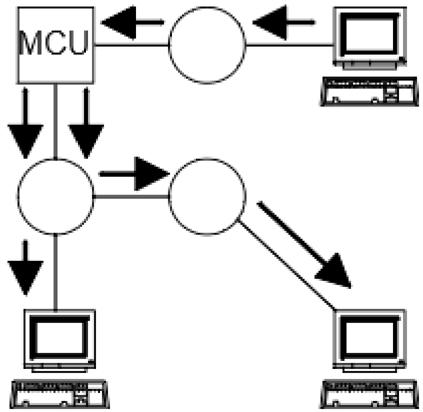
- Simple P2P topology
  - Everyone sends to everyone else
  - Send/receive n-1 flows for n users
  - Congestion close to the server
    - And the receiver

Peer-to-peer (2 of 2)



- P2P with multicasting
  - All users belong to a group
  - Each sender sends a single flow
    - The network replicates it
  - Congestion only close to the receiver

Server-based (1 of 4)

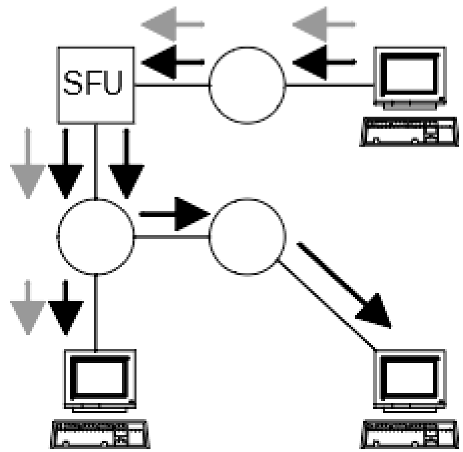


- MCU: Multipoint Conference (or Control) Unit
  – Central server for conferencing
  – Each sender only transmits to the MCU
  – The MCU creates a single flow for all receivers
    - In principle, can be sent with multicasting

Server-based (2 of 4)

- An MCU mixes the media
  - Small window per participant or speaker only
  - Audio mix or selection of one participant
  - Composes a single outgoing flow
  - This means decoding, transforming, recoding
  - Considerable processing delay
  - Plus, possible network delay
    - Depending on the MCUs location

Server-based (3 of 4)



- SFU: Selective Forwarding Unit
  - Just copies and relays media
    - May drop some of them
    - No processing delays (only copying)
    - Only network delays (depending on location)

Server-based (4 of 4)

- An SFU does not relay everything
  - Otherwise, the receiver will suffer!
  - Use of layered coding in each stream
  - Each receiver chooses what to receive
    - Receiver talks to the SFU
    - Can ask for low resolution video from all
    - Or for high resolution video from speaker
    - Plus audio from speaker

End of Section # 23

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