Internet Telephony: H.323
Overview

- H.323 architecture
- elements: gatekeepers, terminals, ...
- H.323 call setup
- H.323 features
- comparison with SIP
H.323 Components

H.323: overall architecture =

H.225.0: call control, RAS to gatekeeper: “may I?”, user location; RTP/RTCP

H.235: security for H.323 terminals

H.245: capabilities exchange, indications, notifications

H.246: interoperability with PSTN

H.332: large group conferences

H.450: supplementary services

H.246: interworking between H.323 and other H.xxx standards

Q.931: call setup = ISDN, similar to Q.2931 (ATM)
Q.932: supplementary services
H.323

- derived from H.320 (ISDN multimedia)
- mostly ASN.1 (PER) based
- signaling TCP-based, except for H.323v4
**H.323 Components**

**G.711:** 64 kb/s $\mu$/A-law audio

**G.723:** 5.6 or 6.3 kb/s audio

**G.729:** 8 kb/s audio

**H.261, H.263:** conference video coding

- ITU SG-16

- version 2 in use (Feb. 1998), version 3 (Sep. 1999)
H.323 Zones
H.323 Elements

**H.323 Terminal**: PC with H.323 software

**MCU**: multipoint control unit ➔ mixes audio and video

**MC**: multipoint controller ➔ performs signaling for centralized conferences

**MP**: multipoint process ➔ actual device for mixing

**gatekeeper**: session control: address translation, admission control, bandwidth control, zone management

**gateway**: interface between H.323 systems and other systems: PSTN, H.323 (PSTN mm), H.320 (ISDN), H.321 (ATM mm)
H.323 gatekeeper

- controls sessions
- performs user location and registration
- admission control
- reroutes signaling
- processes RAS (registration, admission, status) from H.323 terminals
Gatekeeper Discovery

endpoint

GRQ

GCF

GRJ

gatekeeper
Registration with Gatekeeper

Registration

RRQ

RCF
RRJ

URQ

UCF
URJ

URQ

UCF

endpoint-initiated unregistration

gatekeeper-initiated unregister request
H.323 Phases

Initialization: register with GK

GK admission: obtain permission
    GK resolves address

Call signaling: signaling connection to peer
    call initiation and completion/rejection

Negotiation/configuration: negotiate roles during call
    capability exchange
    determine mode of operation

Media exchange: configure and open logical channels
    transmit and receive data streams

Re-negotiation: change members, parameters, media, …
**Shutdown:** terminate the call/conference
deregister user on log-off
H.323 Channels

**RAS:** endpoint – gatekeeper: H.225.0 (UDP)

**Call signaling:** call control & supplementary services; ≈ Q.931 (TCP; v3: also UDP)

**H.245 control:** media control, capability exchange; open “logical channels” (TCP)

**Logical channel:** carry audio, video, media (UDP)
H.323v1 call setup (w/o fastStart)
Gatekeeper-routed Signaling

```
endpoint 1
  1
  2
  3
  8

gatekeeper cloud

endpoint 2
  4
  5
  6
  7
```

Flow details:
- ARQ from endpoint 1 to gatekeeper cloud
- ACF/ARJ from endpoint 1 to gatekeeper cloud
- setup from gatekeeper cloud to endpoint 2
- ARQ from gatekeeper cloud to endpoint 2
- ACF/ARJ from gatekeeper cloud to endpoint 2
- connect from endpoint 1 to gatekeeper cloud
- connect from gatekeeper cloud to endpoint 2
Both Endpoints Registered – GK Routed
**H.323v3 call setup**

- Caller
- Gatekeeper
- Callee

- **RAS** communication
- **SETUP** message
- **CALL PROCEEDING** message
- **ALERTING** message
- **CONNECT** message

- Audio
- Port 1720, 1300
- 2 dynamic ports
H.323 call holding: near & remote-end

served user A

near-end hold

FACILITY (holdNotific.inv)

music on hold (MOH)

FACILITY (remoteHold.inv)

remote-end hold

FACILITY (remoteHold.rr)

held user A

local MOH
H.323 call diversion

Originator A
(caller)

GK (rerouting)

Served B
(original destination)

Diverted-to C
(final destination)

SETUP

CALL PROCEEDING

FACILITY(CallRerouting.invoke)

FACILITY(CallRerouting.ReturnResult)

RELEASE COMPLETE

SETUP(diverting.LegInfo2.inv)

FACILITY (diverting.LegInfo2.inv)

ARQ

ACF

ALERTING

CONNECT(...)

CONNECT(diverting.LegInfo3.inv)
**H.323 blind call transfer**

transferring party ➔ transferred party ➔ transfer destination

- SETUP
- ALERTING
- CONNECT
- FACILITY.ctInvoke
- RELEASE COMPLETE.ctResult
- SETUP.ctTransfer=cid
- CALL PROCEEDING
- RELEASE COMPLETE
- CONNECT
H.323 problems

- very complex (200+ pages; 65 pages for call forwarding!)
- no multicast signaling
- limited multicast conferences (MCUs)
- call = TCP connection ↔ mobility, reliability
- but: better capability negotiation (H.245)
- no media servers
- agile ports → firewalls difficult
H.323 delay

- several TCP connections ➔ very long latency (6.5-8 RTTs)
- 1 TCP SYN loss ➔ delay of 6 seconds
- 2 TCP SYN losses ➔ delay of 24 seconds

➢➢ H.323v2 for fewer connections, UDP?
## SIP – H.323 comparison

<table>
<thead>
<tr>
<th></th>
<th>H.323</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Architecture</td>
<td>stack</td>
<td>element</td>
</tr>
<tr>
<td>Conference control</td>
<td>yes</td>
<td>no</td>
</tr>
<tr>
<td>Protocol</td>
<td>mostly TCP</td>
<td>mostly UDP</td>
</tr>
<tr>
<td>Encoding</td>
<td>ASN.1, Q.931</td>
<td>HTTPish</td>
</tr>
<tr>
<td>Emphasis</td>
<td>telephony</td>
<td>multimedia, multicast</td>
</tr>
<tr>
<td>Address</td>
<td>flat alias, E.164</td>
<td>SIP, E.164 URLs</td>
</tr>
</tbody>
</table>

Both SIP and H.323 are evolving: SIP additions, H.323v2 implemented, v3 to be decided.
SIP – H.323 comparison


H.323 Resource Reservation

- *local* admission decision
- prior to call setup → no information about bandwidth available
- works only for “yellow cable Ethernet”
- other applications have to notify GK
- SIP: RSVP, YESSIR, DiffServ + call preconditions
SIP vs. H.323 call setup

**H.323v1:** several TCP connections (H.245, Q.931) → very long latency (6.5-8 RTTs), particularly with packet loss; currently in *NetMeeting*

**H.323v2:** merge H.245 and Q.931 (“FastConnect”)

**H.323v3:** allow UDP

End systems need to support all versions.
### H.323 vs. SIP: basic call control

(modified from Dalgic and Fang, *Comparison of H.323 and SIP*)

<table>
<thead>
<tr>
<th>Service</th>
<th>H.323v1</th>
<th>v2</th>
<th>v3</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call holding</td>
<td>no</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
<tr>
<td>Call transfer</td>
<td>no</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
<tr>
<td>Call forwarding</td>
<td>no</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
<tr>
<td>Call waiting</td>
<td>no</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
</tbody>
</table>
H.323 vs. SIP: advanced features

<table>
<thead>
<tr>
<th>Service</th>
<th>H.323v1</th>
<th>v2</th>
<th>v3</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Third party control</td>
<td>no</td>
<td>no</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Conference</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
<tr>
<td>Click-to-dial</td>
<td>?</td>
<td>?</td>
<td>?</td>
<td>PINT</td>
</tr>
<tr>
<td>Capability exchange</td>
<td>better</td>
<td>better</td>
<td>better</td>
<td>yes</td>
</tr>
<tr>
<td>HTML transport</td>
<td>no</td>
<td>no</td>
<td>no</td>
<td>yes</td>
</tr>
</tbody>
</table>
# H.323 vs. SIP: services

<table>
<thead>
<tr>
<th>Service</th>
<th>H.323</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call transfer</td>
<td>H.450.2</td>
<td>“30x”</td>
</tr>
<tr>
<td>Call diversion</td>
<td>H.450.3</td>
<td>“30x”</td>
</tr>
<tr>
<td>Call hold</td>
<td>H.450.4</td>
<td>SDP-based</td>
</tr>
<tr>
<td>Call park</td>
<td>H.450.5</td>
<td>REGISTER</td>
</tr>
<tr>
<td>Call waiting</td>
<td>H.450.6</td>
<td>INVITE</td>
</tr>
<tr>
<td>Message waiting</td>
<td>H.450.7</td>
<td>email, NOTIFY</td>
</tr>
<tr>
<td>Call forward busy</td>
<td>H.450.9</td>
<td>“30x”</td>
</tr>
</tbody>
</table>
## H.323 vs. SIP: quality of service

<table>
<thead>
<tr>
<th></th>
<th>H.323v1</th>
<th>v2</th>
<th>v3</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call setup delay</td>
<td>6-7 RTT</td>
<td>3-4</td>
<td>1.5-2.5</td>
<td>1.5</td>
</tr>
<tr>
<td>Loss recovery</td>
<td>TCP</td>
<td>TCP</td>
<td>better</td>
<td>better</td>
</tr>
<tr>
<td>Fault detection</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
<tr>
<td>Mid-call failure</td>
<td>fail</td>
<td>fail</td>
<td>fail</td>
<td>live</td>
</tr>
<tr>
<td>Registrar failure</td>
<td>fail</td>
<td>fail</td>
<td>backup</td>
<td>multicast</td>
</tr>
<tr>
<td>GK/Proxy redundancy</td>
<td>no</td>
<td>no</td>
<td>backup</td>
<td>SLP, DNS, DHCP</td>
</tr>
<tr>
<td>Loop detection</td>
<td>no</td>
<td>no</td>
<td>PathValue</td>
<td>Via, hops, time</td>
</tr>
</tbody>
</table>
## H.323 vs. SIP: manageability

<table>
<thead>
<tr>
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<th>v2</th>
<th>v3</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Admission control</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
<td>no (RSVP)</td>
</tr>
<tr>
<td>Policy control</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
<td>ob proxy</td>
</tr>
<tr>
<td>Resource reservation</td>
<td>local</td>
<td>local</td>
<td>local</td>
<td>no (RSVP)</td>
</tr>
</tbody>
</table>
## H.323 vs. SIP: scalability

<table>
<thead>
<tr>
<th></th>
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<th>v2</th>
<th>v3</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Complexity</td>
<td>more</td>
<td>more</td>
<td>more+</td>
<td>less</td>
</tr>
<tr>
<td>Server processing</td>
<td>SF</td>
<td>SF</td>
<td>SF/SL, TSF</td>
<td>SL, TSF/TSL</td>
</tr>
<tr>
<td>Inter-server</td>
<td>no</td>
<td>no</td>
<td>yes</td>
<td>yes</td>
</tr>
</tbody>
</table>

TS: transaction state; SF: call statefull; SL: call stateless
## H.323 vs. SIP: flexibility

<table>
<thead>
<tr>
<th></th>
<th>H.323 v1</th>
<th>v2</th>
<th>v3</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport protocols</td>
<td>TCP</td>
<td>TCP</td>
<td>TCP/UDP</td>
<td>any</td>
</tr>
<tr>
<td>Extensibility</td>
<td>unlabeled vendor extensions</td>
<td>IANA, labeled</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Customization</td>
<td>harder</td>
<td></td>
<td></td>
<td>easier</td>
</tr>
<tr>
<td>Version compatibility</td>
<td>N/A</td>
<td>yes</td>
<td>yes</td>
<td>N/A</td>
</tr>
<tr>
<td>SCN interoperability</td>
<td>good</td>
<td>good</td>
<td>good</td>
<td>TBD</td>
</tr>
<tr>
<td>Protocol encoding</td>
<td>binary (ASN.1, Q.931)</td>
<td>text</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>