Interworking Between SIP/SDP and H.323

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SIP vs H.323

- Text based request response
- SDP (media types and media transport address)
- Server roles: registrar, proxy, redirect

Both use RTP/RTCP over UDP/IP

- Binary ASN.1 PER encoding
- Sub-protocols: H.245, H.225 (Q.931, RAS, RTP/RTCP), H.450.x...
- H.323 Gatekeeper
Interworking Problems

Call setup translation

H.323

Q.931 SETUP
Q.931 CONNECT
Terminal Capabilities
Terminal Capabilities
Open Logical Channel
Open Logical Channel

SIP

Destination address
Media capabilities
Media transport address

INVITE
200 OK
ACK

• Multi-stage dialing
• H.323v2 Fast-start is optional

Interworking Problems

User Registration

H.323

H.323 Gatekeeper
SIP registrar

SIP user agent

H.323 terminal

• Location independent user identifier?
• Use information from both networks
Interworking Problems
Media Description

H.323/H.245
Supports inter-media constraints

{ [G.711 Mu law, G.711 A law][H.261 video]} { [G.723.1] [no video] }

SIP/SDP
List of alternative set of algorithms.

audio  G.711 Mu law, G.723.1, G.728
video  H.261

• Translation in both directions
• Algorithm selection by end-systems

Interworking Problems
Advanced Services

• H.323 Conferencing: centralized signaling control, MC (Multi-point Controller)
• Supplementary services: H.450.x

• SIP Conferencing: centralized bridged + decentralized distributed

• New headers: Also, Requested-By, Replaces
User registration

- Registration info to foreign network
- Three ways: SGW + GK, SGW + proxy/registrar, SGW

```
REGISTER
hgs@columbia.edu
Contact: sgw
```

```
H.323 Gatekeeper + SGW
sgw.columbia.edu
```

```
SIP registrar server
```

```
RRQ
hgs@columbia.edu
Contact: 128.59.19.200
```

```
SIP user agent
```

```
INVITE hgs@columbia.edu
```

```
3xx Moved
Contact: sgw
```

```
REGISTER
hgs@columbia.edu
Contact: sgw
```

```
INVITE hgs@columbia.edu
```

```
3xx Moved
Contact: sgw
```

```
REGISTER
hgs@columbia.edu
Contact: sgw
```

```
INVITE hgs@columbia.edu
```

```
3xx Moved
Contact: sgw
```

```
REGISTER
hgs@columbia.edu
Contact: sgw
```

```
INVITE hgs@columbia.edu
```

```
3xx Moved
Contact: sgw
```

- Independent SGW preferable - use SIP OPTIONS and H.323 LRQ

Call Setup

with H.323v2 Fast Start

One-to-one mapping between SIP and H.323 messages.

```
INVITE
```

```
ACK
```

```
200 OK
```

RTP/RTCP

Reverse direction is similar
Call Setup
without Fast Start, SIP to H.323

H323
→
INVITE

SIP
←
200 OK

Signaling Gateway

Capabilities/H245

Open Logical Channel / H245
Acknowledgement

Capabilities/H245

Open Logical Channel / H245
Acknowledgement

Media Transport Address

200 OK

RAR/TCP

Call Setup
without Fast Start, H.323 to SIP

H323
→
INVITE

SIP
←
200 OK

Signaling Gateway

Setup/Q931

Connect/Q931

Capability Exchange

Open Logical Channel
Acknowledgement

Open Logical Channel
Acknowledgement

Media Transport Address

RAR/TCP

Re-INVITE/SIP+SDP
**Capability Set**

- Capability set in each direction
- Maximal intersection and current operating modes
- Re-INVITE or change in H.323 mode or logical channels, whenever it changes

Example:

\[ C_1 = \{ [\text{PCMU, PCMA, G.723.1}] [\text{H.261}] \} \]
\[ C_2 = \{ [\text{PCMU, PCMA, G.729}] [\text{H.261}] \} \]
\[ C_1 \cap C_2 = \{ [\text{PCMU, PCMA}] [\text{H.261}] \} \]

operating modes = [audio=PCMU, video=H.261]

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**Conclusion and Future Work**

- Ad-hoc conferencing
- SIP centered or H.323 centered conferencing
- Basic call setup ⇒ other supplementary services

- Our demonstration setup (openh323+Columbia stack) for basic audio call
- IETF, ITU and ETSI TIPHON
- Convergence between SIP and H.323 in newer versions