Interworking Between SIP/SDP and H.323

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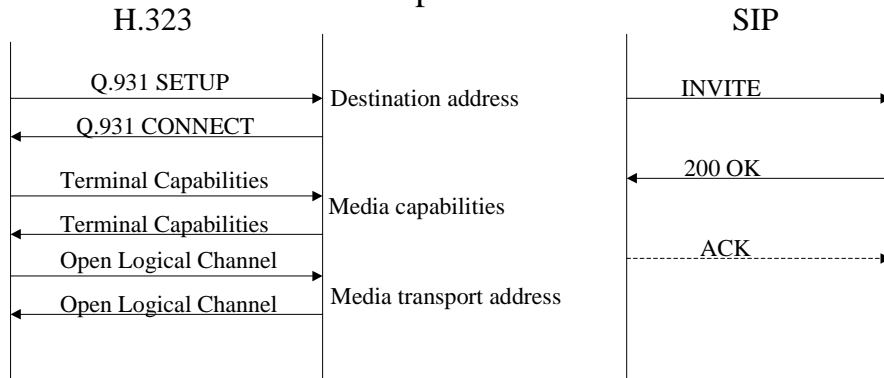

SIP   vs   H.323

SIP	H.323
• Text based request response	• Binary ASN.1 PER encoding
• SDP (media types and media transport address)	• Sub-protocols: H.245, H.225 (Q.931, RAS, RTP/RTCP), H.450.x...
• Server roles: registrar, proxy, redirect	• H.323 Gatekeeper

Both use RTP/RTCP over UDP/IP

Interworking Problems
Call setup translation

H.323		SIP

Q.931 SETUP → Destination address → INVITE →

← Q.931 CONNECT

Terminal Capabilities → Media capabilities ← 200 OK

← Terminal Capabilities

Open Logical Channel → Media transport address --- ACK --->

← Open Logical Channel
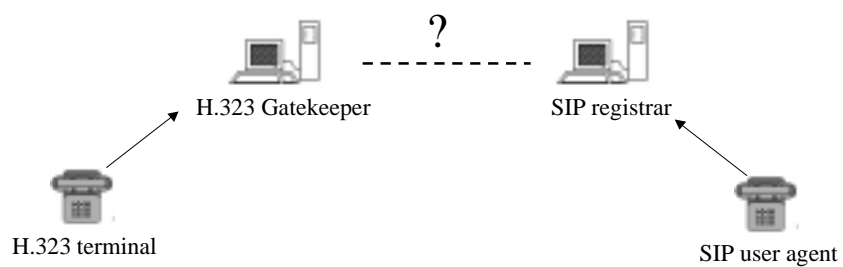
- Multi-stage dialing
- H.323v2 Fast-start is optional

Interworking Problems
User Registration

H.323   SIP

?

H.323 Gatekeeper   SIP registrar

H.323 terminal   SIP user agent

- 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
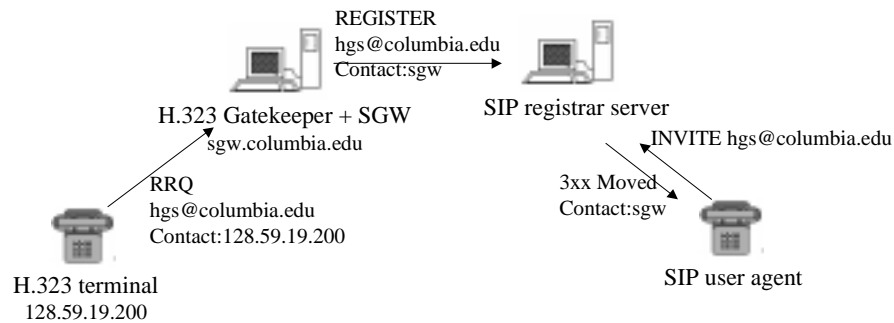- Algorithm selection by end-systems

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
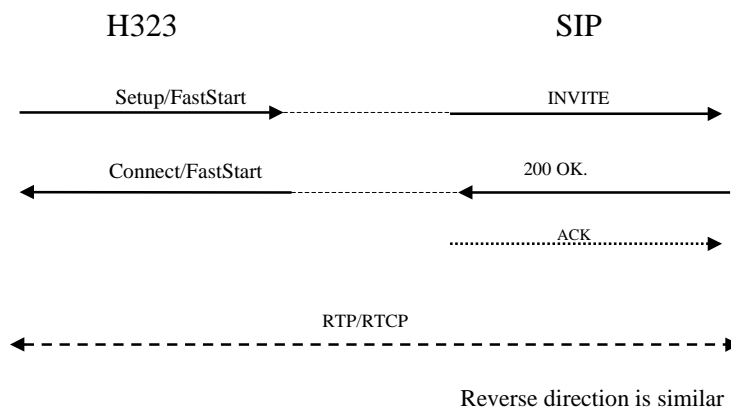- New headers : Also, Requested-By, Replaces

User registration

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

REGISTER
hgs@columbia.edu
Contact:sgw

SIP registrar server

H.323 Gatekeeper + SGW
sgw.columbia.edu

INVITE hgs@columbia.edu

RRQ
hgs@columbia.edu
Contact:128.59.19.200

3xx Moved
Contact:sgw

H.323 terminal
128.59.19.200

SIP user agent

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

Call Setup
with H.323v2 Fast Start
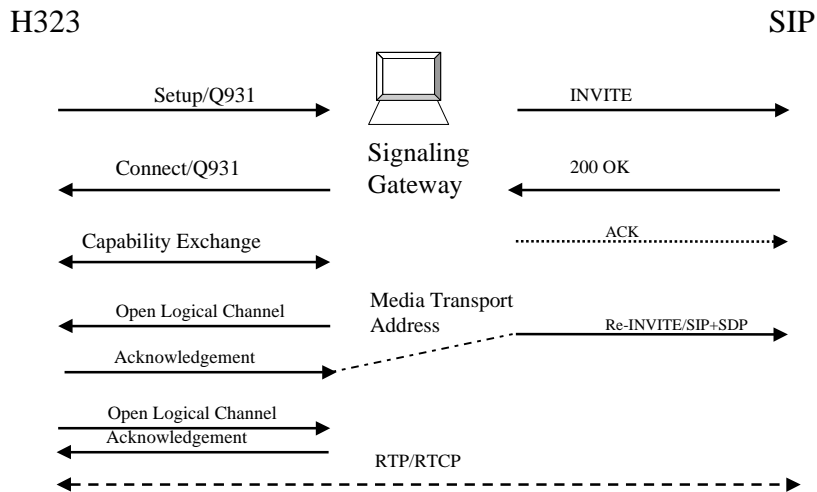
One-to-one mapping between SIP and H.323 messages.

H323   SIP

Setup/FastStart   INVITE

Connect/FastStart   200 OK.

ACK

RTP/RTCP

Reverse direction is similar

Call Setup
without Fast Start, SIP to H.323

H323   SIP

Setup/Q931   Signaling   INVITE
                                    Gateway

Connect/Q931

Capabilities/H245

Capabilities/H245.

Open Logical Channel/ H245   Media Transport   200 OK.
Acknowledgement   Address

Open Logical Channel / H245   ACK
Acknowledgement

                                                RTP/RTCP
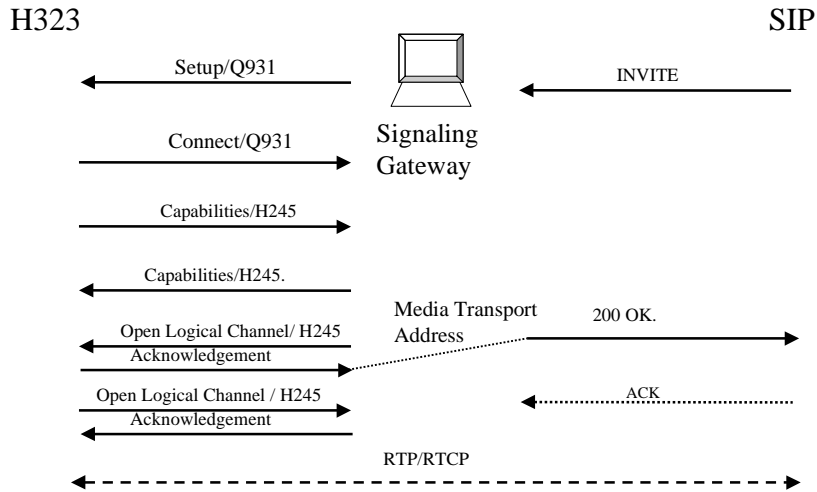
Call Setup
without Fast Start, H.323 to SIP

H323   SIP

Setup/Q931   Signaling   INVITE
                                    Gateway

Connect/Q931   200 OK

Capability Exchange   ACK

Open Logical Channel   Media Transport   Re-INVITE/SIP+SDP
                                    Address
Acknowledgement

Open Logical Channel
Acknowledgement   RTP/RTCP

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:
C1 = { [PCMU, PCMA, G.723.1] [H.261] }
C2 = { [PCMU, PCMA, G.729] [H.261] }
C1 ∩ C2= {[PCMU, PCMA] [H.261]}
operating modes = [audio=PCMU,video=H.261]

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