STATE KEY LABORATORY OF SWITCHING TECHNOLOGY AND TELECOMMUNICATION NETWORK

北京邮电大学

BEIJING UNIVERSITY OF POSTS AND TELECOMMUNICATIONS

SIP协议分析

—— SIP协议基础架构（续）

理解协议处理的基本原理与实现模型

- 协议的结构
- 消息的处理规则

- 用户管理
- 组网与路由
- 会话建立与媒体协商

用户管理

- SIP/SIPS用户地址 — SIP/SIPS URI
  - SIP and SIPS Uniform Resource Indicators
  - 格式：

    sip:user:password@host:port;uri-parameters?headers

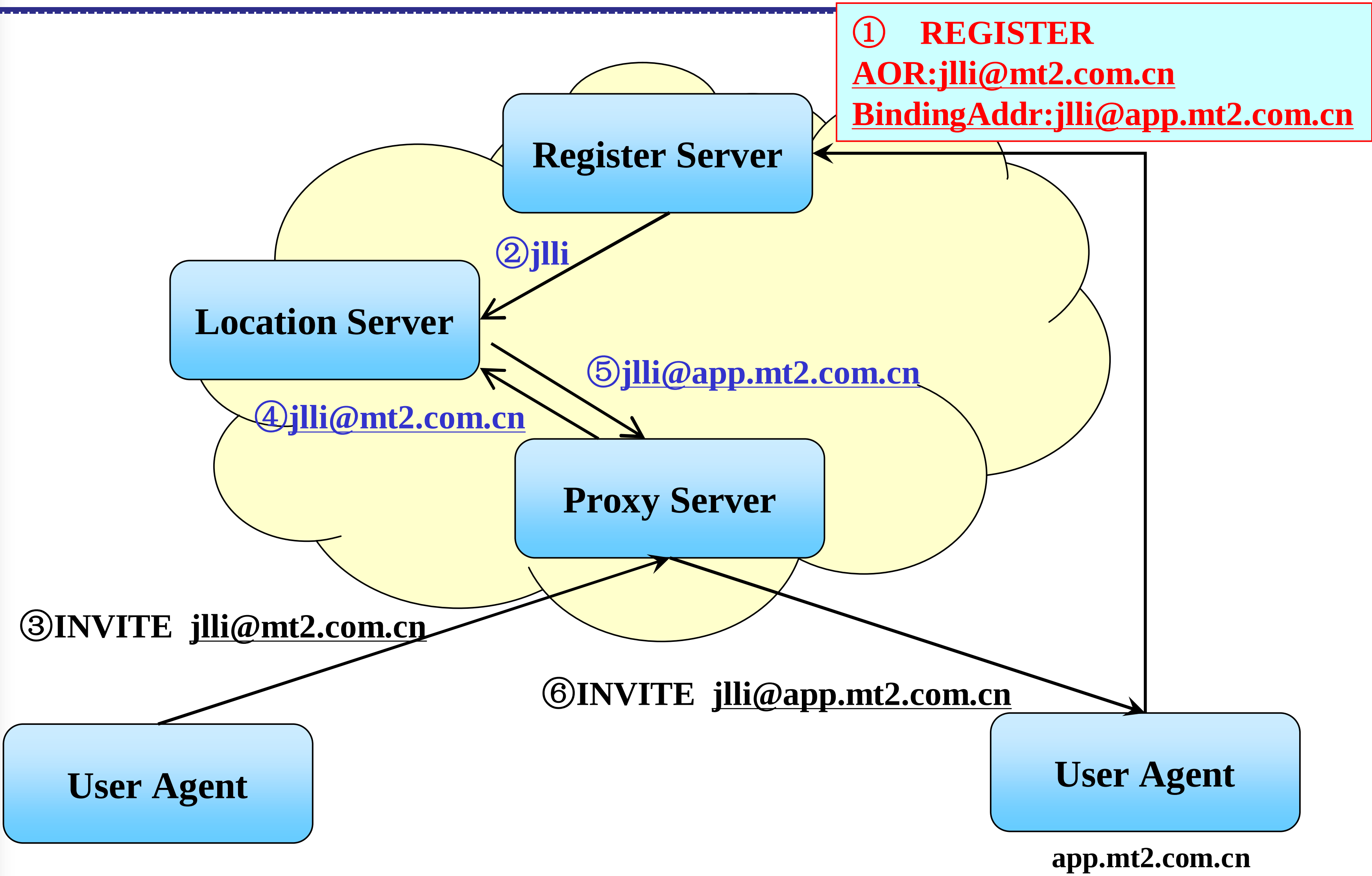
  - 合法的SIP URI
    - sip:alice@atlanta.com
    - sip:bob@biloxi.com:5060
    - sip:bob@biloxi.com;transport=udp
    - sip:bob@192.0.2.4
    - sip:bob@biloxi.com:5060;lr
  - 合法的URI
    - tel:+358-555-1234567;postd=pp22
    - sip:+358-555-1234567;postd=pp22@foo.com;user=phone

    - RFC2806 (tel: URLs for Telephone Calls)
    - RFC2396 (Uniform Resource Identifiers (URI))
    - 等等

用户管理的基本概念

- Address-of-Record（AOR）
  - An address-of-record (AOR) is a SIP or SIPS URI
  - AOR points to a domain with a location service that can map the URI to another URI where the user might be available
  - Typically, the location service is populated through registrations
  - An AOR is frequently thought of as the "public address" of the user
- Home Domain（归属控制域）
  - The domain providing service to a SIP user
  - Typically, this is the domain present in the URI in the address-of-record (AOR) of a registration
- AOR － HOME Domain的关系
  - AOR是用户的唯一号码
  - HOME Domain维护用户唯一号码与实际号码的映射关系

注册与用户管理过程


① REGISTER
AOR:jlli@mt2.com.cn
BindingAddr:jlli@app.mt2.com.cn
Register Server
②jlli
Location Server
⑤jlli@app.mt2.com.cn
④jlli@mt2.com.cn
Proxy Server
③INVITE jlli@mt2.com.cn
⑥INVITE jlli@app.mt2.com.cn
User Agent
User Agent
app.mt2.com.cn

注册请求与响应

注册到的Register Server

REGISTER sips:ss2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061;bran
Max-Forwards: 70
To: Bob <sips:bob@biloxi.example.com>
From: Bob <sips:bob@biloxi.example.com>;tag=ja743k
Call-ID: 1j9FpLxk3uxtm8tn@biloxi.example.com
CSeq: 2 REGISTER
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0

要注册的AOR
必须是SIP / SIPS地址

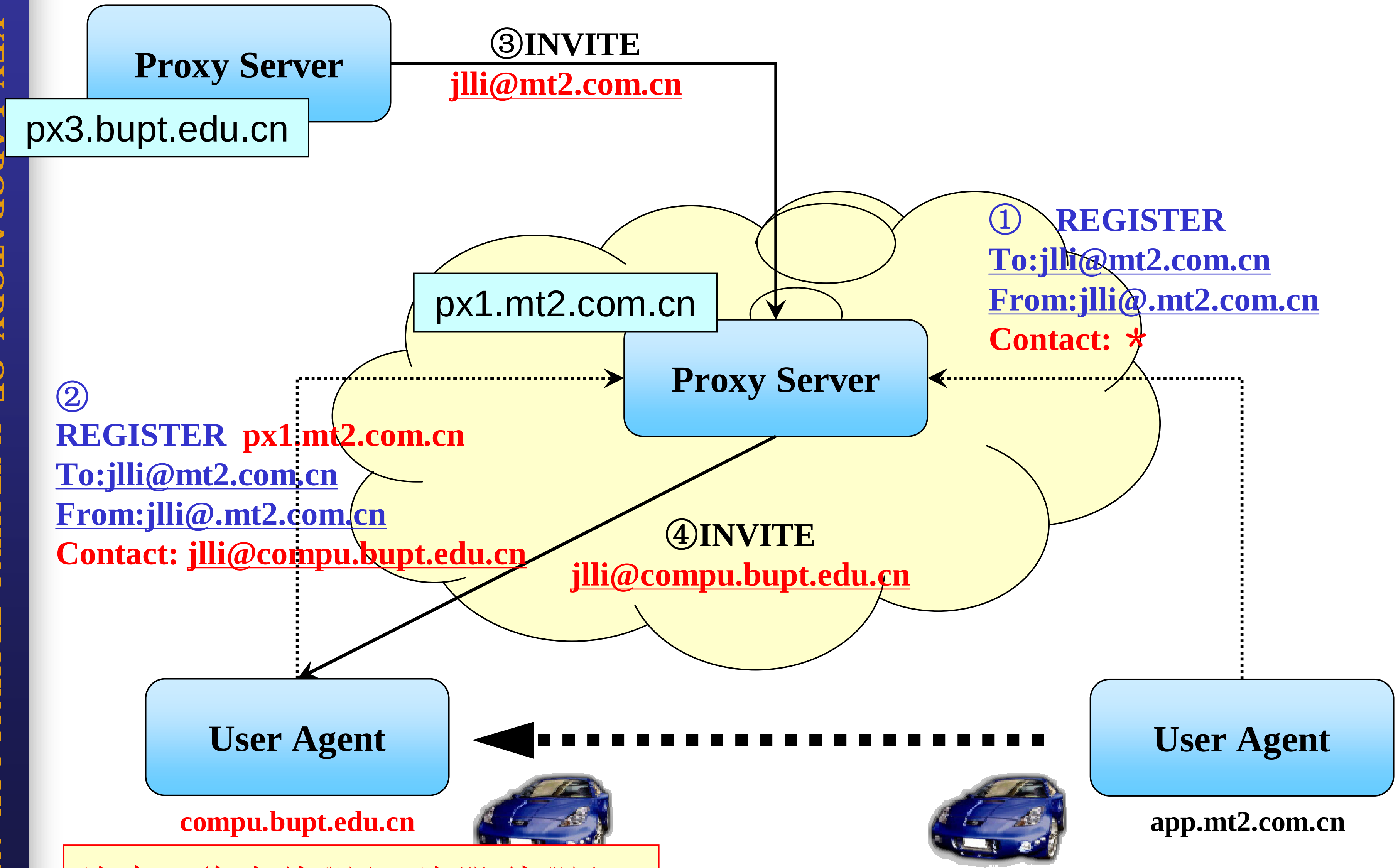
发起注册请求的AOR
缺省From=To
特殊：第三方注册

注册AOR所绑定的实际联系地址
可以携带0～n个合法的URL地址
Contact: * = 取消注册
不携带Contect =查询

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:50
received=192.0.2.201
From: Bob <sips:bob@biloxi.example.com>;tag=ja743ks76zlflH
To: Bob <sips:bob@biloxi.example.com>;tag=37GkEh
Call-ID: 1j9FpLxk3uxtm8tn@biloxi.example.com
CSeq: 2 REGISTER
Contact: <sips:bob@client.biloxi.example.com>;expires=3600
Contact: <mailto:bob@biloxi.example.com>;expires=4294967295
Content-Length: 0

当前已绑定的联系地址
及相应的生存周期
取消注册: 返回空

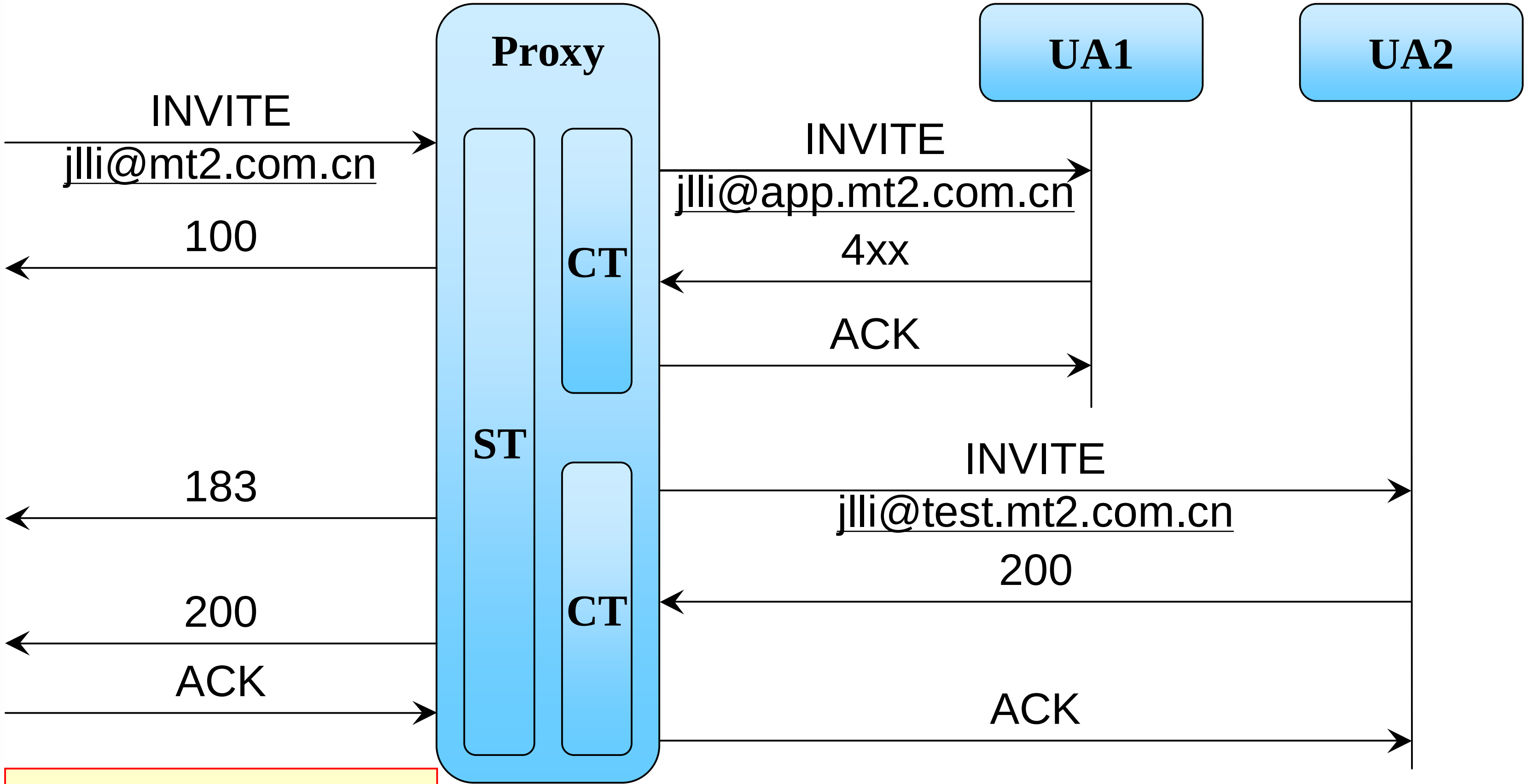
Personal Mobility



注意：移动到哪里？注册到哪里？

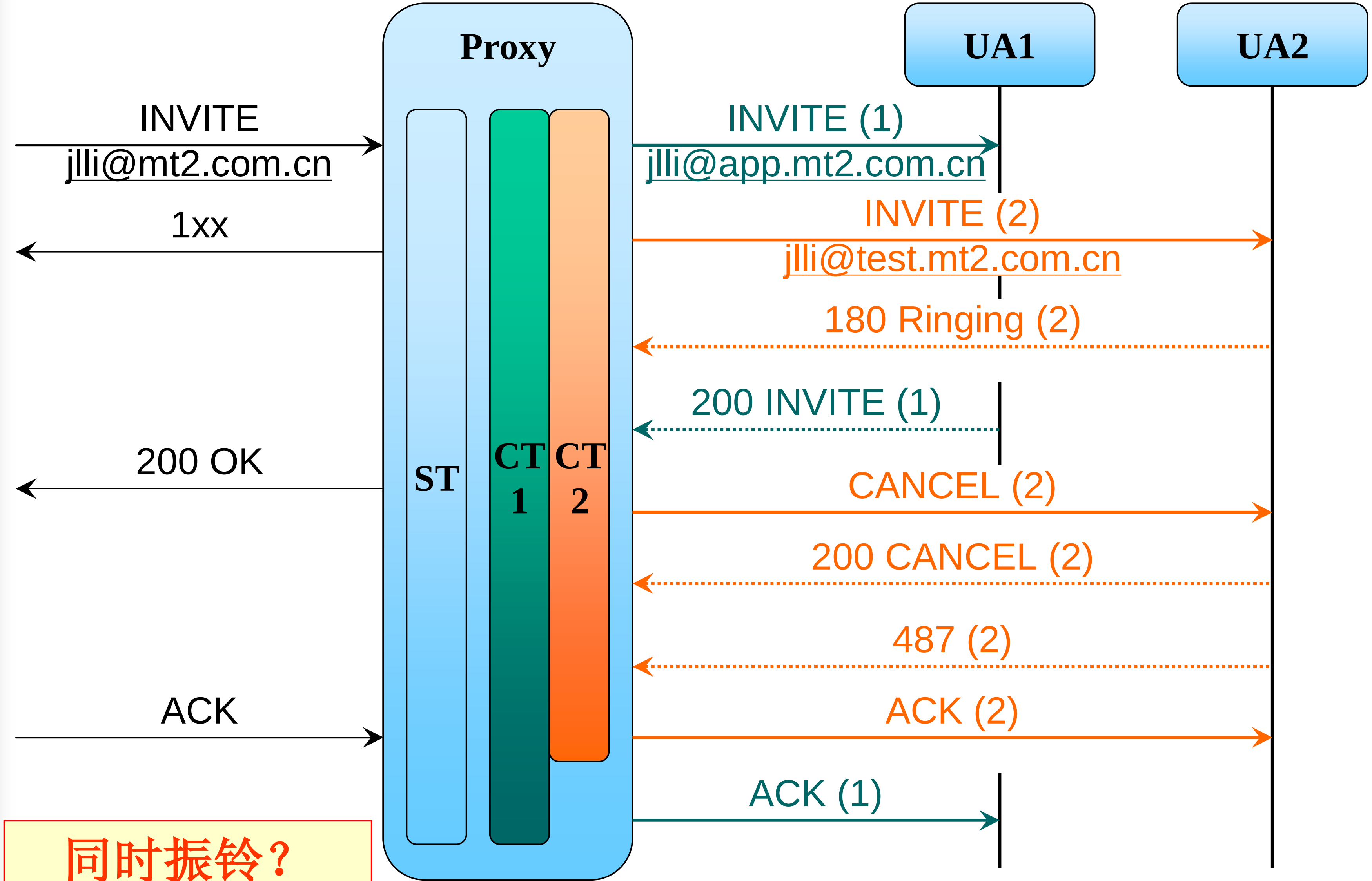
绑定多个有效地址

- AOR: jlli@mt2.com.cn
  - Binding Address 1: jlli@app.mt2.com.cn
  - Binding Address 2: jlli@test.mt2.com.cn



依次振铃？

Forking



同时振铃？

用户管理小结

- Domain
- AOR
- 注册请求与响应
- 注册过程

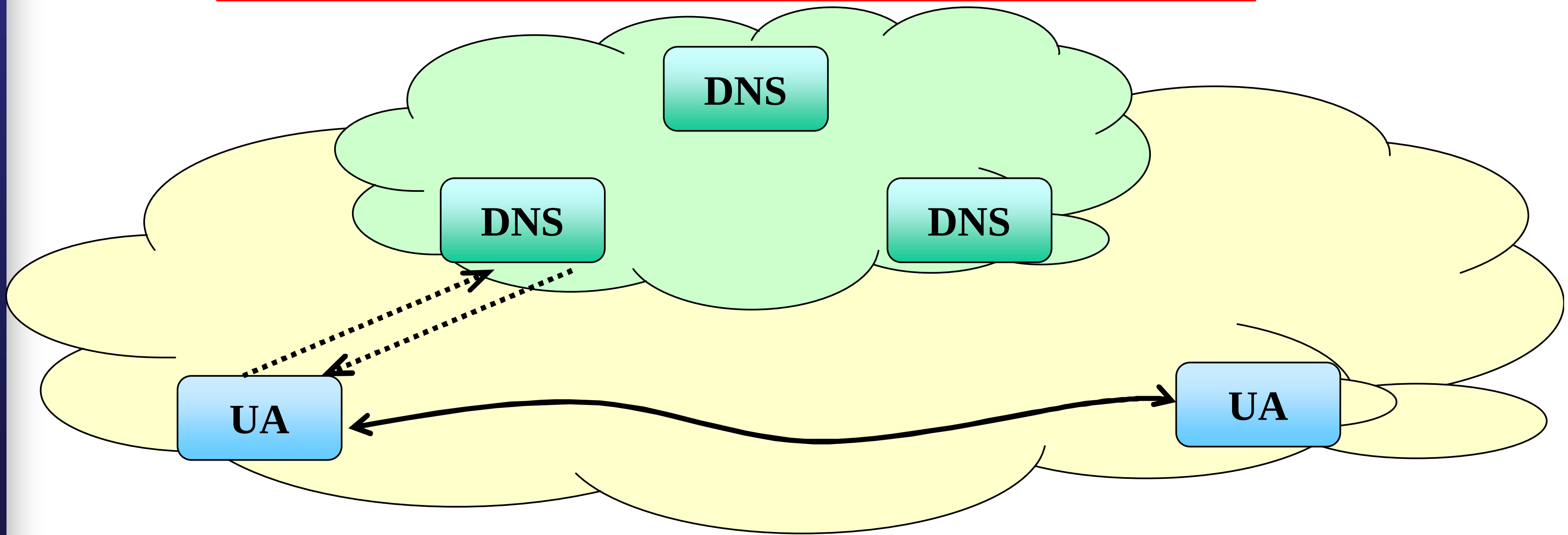
- 基于Target Set可以做的工作
  - Personal Mobility
  - 依次振铃
  - 同时振铃

这些是SIP协议支持的业务能力吗？

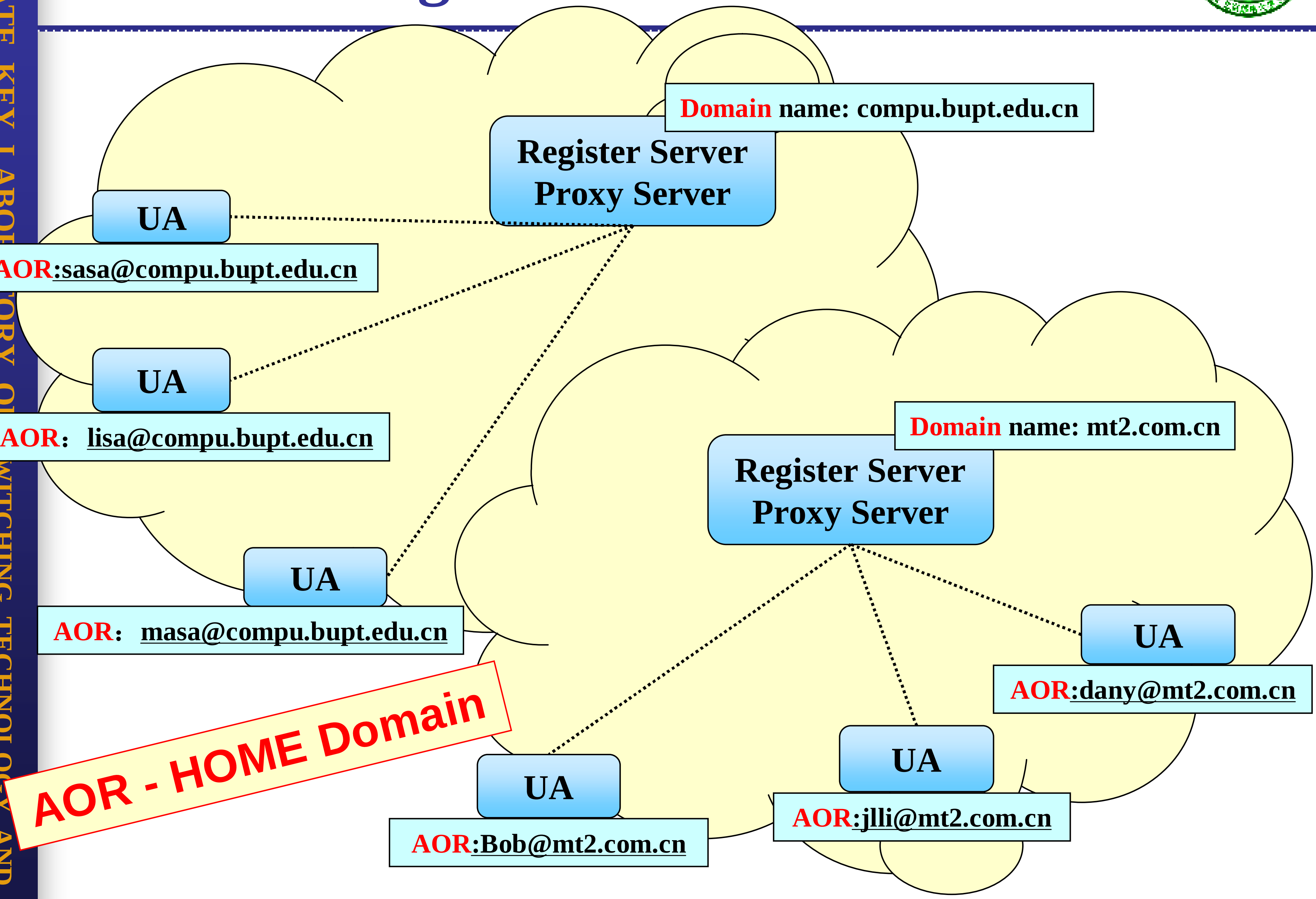
组网与路由Networking & Routing

- Networking
  - SIP协议需要组网吗？
  - DNS解析被叫用户地址不可以吗？
  - 回想 SIP协议注册机制：AOR－HOME Domain

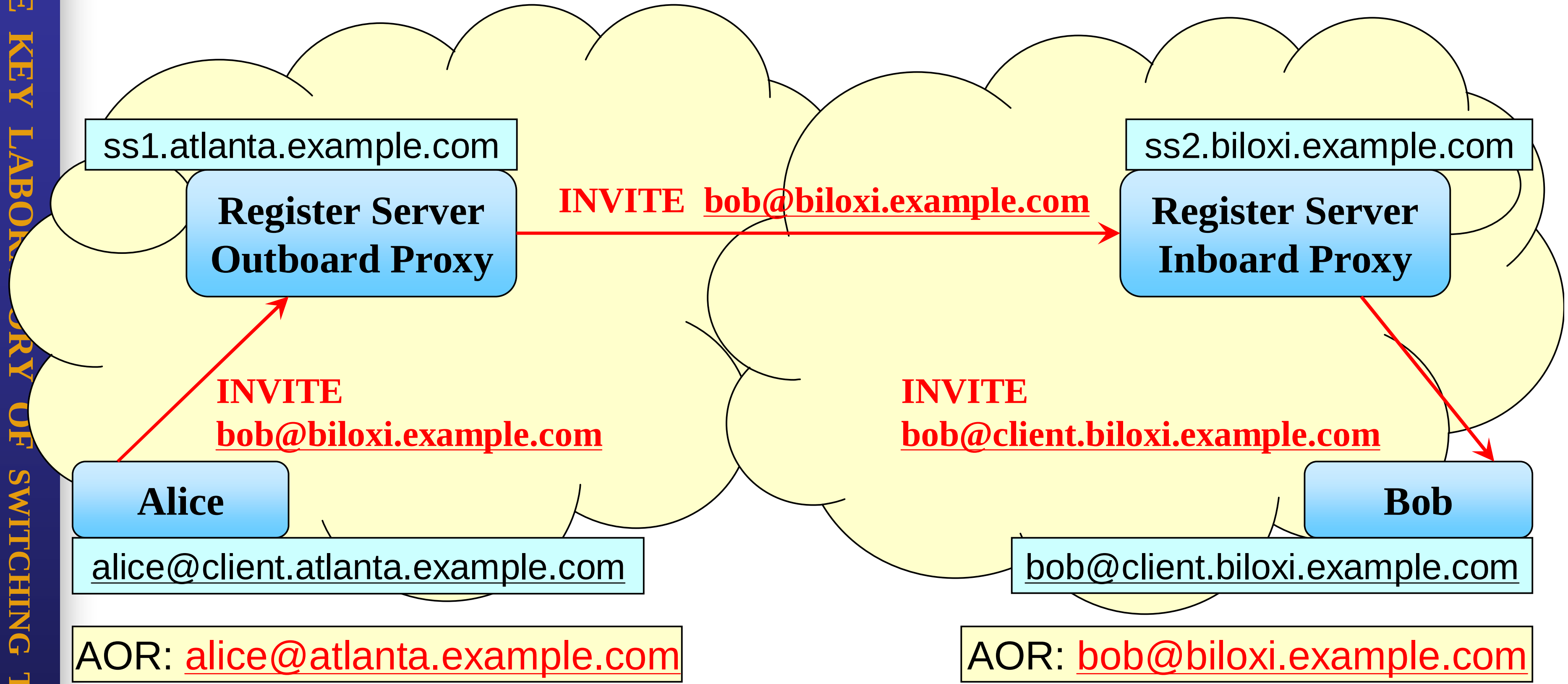
SIP组网的目的是完成用户管理



Networking


Domain name: compu.bupt.edu.cn
Register Server
Proxy Server
UA
AOR:sasa@compu.bupt.edu.cn
UA
AOR: lisa@compu.bupt.edu.cn
UA
AOR: masa@compu.bupt.edu.cn
Domain name: mt2.com.cn
Register Server
Proxy Server
UA
AOR:dany@mt2.com.cn
UA
AOR:jlli@mt2.com.cn
UA
AOR:Bob@mt2.com.cn
AOR - HOME Domain

路由过程



- 如何将请求路由到目的地
  - Alice-Proxy1：UAC如何转发请求
  - Proxy1-Proxy2：Proxy如何转发目的地非本控制域的请求
  - Proxy2-Bob： Proxy如何转发目的地为本控制域的请求

路由过程

Alice-Proxy1

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.example.com:5060;br
Max-Forwards: 70
Route: <sip:ss1.atlanta.example.com;lr>
Contact: <sip:alice@client.atlanta.example.com;transport=tcp>

Request-URI 被叫AOR

Top Via主叫地址

指定Route URL中包含 lr 参数

主叫实际地址

Proxy1-Proxy2

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.example.com;lr>
Contact: <sip:alice@client.atlanta.example.com;transport=tcp>

Request-URI不变

TopVia添加实际上一跳IP地址；添加TopVia；

递减

删除Route,添加Record-Route

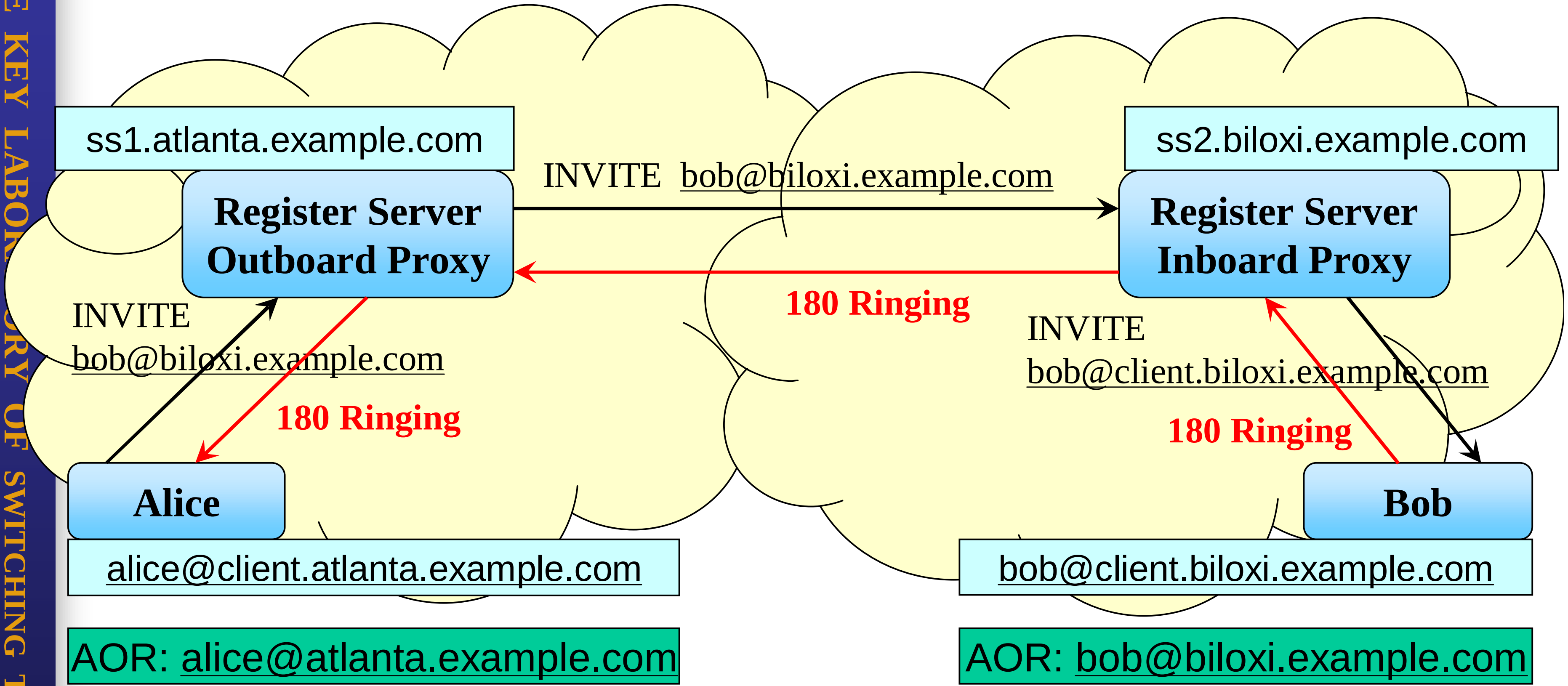
Proxy2-Bob

INVITE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060;branch=z9hG4bK721e4.1
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG
Max-Forwards: 68
Record-Route: <sip:ss2.biloxi.example.com;lr>,<sip:ss1.atlanta.
Contact: <sip:alice@client.atlanta.example.com;transport=tcp>

Bob属于本控制域，Request-URI改变为AOR绑定地址

带给了终端Bob
- **实际转发路径（Via）**
- **主叫实际地址（Contect）**
- **强制路由（Record-Route）**

路由过程



- 如何将请求的响应返回到发端
  - Bob-Proxy2：UAS如何返回响应
  - Proxy2-Proxy1：Proxy如何转发响应
  - Proxy1-Alice： Proxy如何转发响应

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路由过程

Bob-Proxy2

TopVia添加实际上一跳IP地址

SIP/2.0 200 OK
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060;branch=z9hG4bK721e4.1;received=192.0.2.222
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr>,<sip:ss1.atlanta.example.com;lr>
Contact: <sip:bob@client.biloxi.example.com;transport=tcp>

Record-Route不变

Proxy2-Proxy1

Contact变为被叫实际地址

SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr>,<sip:ss1.atlanta.example.com;lr>
Contact: <sip:bob@client.biloxi.example.com;transport=tcp>

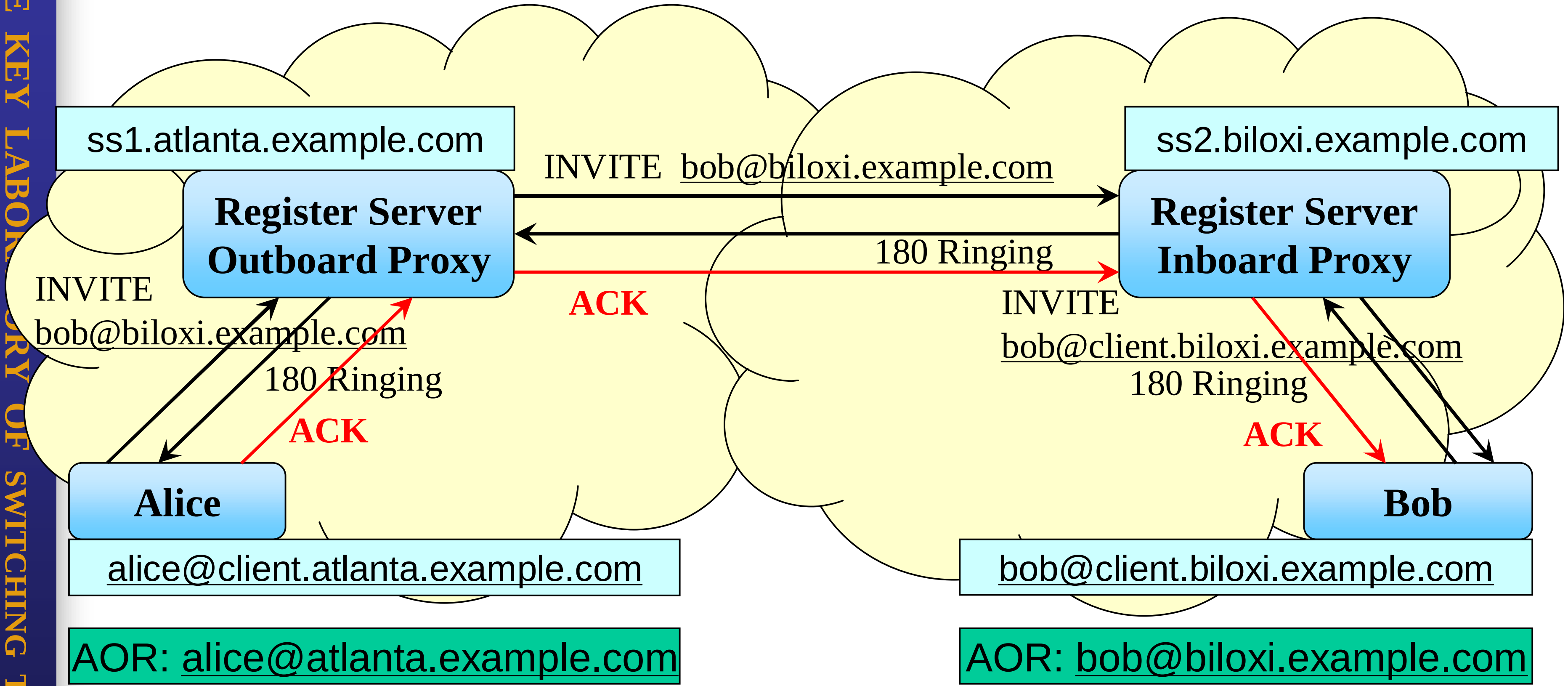
删除TopVia

Proxy1-Alice

SIP/2.0 200 OK
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr>,<sip:ss1.atlanta.example.com;lr>
Contact: <sip:bob@client.biloxi.example.com;transport=tcp>

带给了终端Alice
被叫实际地址（Contect）
强制路由（Record-Route）

路由过程



- 如何转发后继请求
  - Alice-Proxy1：UAC如何处理后继请求
  - Proxy1-Proxy2：Proxy如何转发后继请求
  - Proxy2-Bob： Proxy如何转发目的地为本控制域的后继请求

路由过程

Request-URI变为实际的被叫地址

↓ Alice-Proxy1

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG4bK74b76
Max-Forwards: 70
Route: <sip:ss1.atlanta.example.com;lr>,<sip:ss2.biloxi.example.com;lr>

根据Record-Route
记录设置的Route

↓ Proxy1-Proxy2

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG4bK74b76;received=192.0.2.101
Max-Forwards: 69
Route: <sip:ss2.biloxi.example.com;lr>

删除本Proxy的
Route记录

↓ Proxy2-Bob

直接根据Request-URI
就可以转发了

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060;branch=z9hG4bK721e4.1
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060;branch=z9hG4bK74b76;received=192.0.2.101
Max-Forwards: 68

虽然Request-URI填写的是实际被叫地址，但是ACK将发送给
Top Route 指定的地址：强制路由
最后一跳，不需要再次进行地址查询就可以直接转发

路由过程总结

- 请求
  - 依据Route记录转发请求
  - 如果没有Route，则依据Request-URI转发请求
  - 添加Record-Route记录，将Proxy记录在后继请求转发路径中
  - 添加Via记录，将Proxy记录在响应转发路径中
  - 路由跳数Max-Forward减一
- 响应
  - 依据Via转发响应
- 作用
  - Via保证了响应与请求的路径一致（响应与请求的对应）
  - Record-Route / Route保证了关心这一Dialog的Proxy能够保留在后继请求的转发路径中
  - Max-Forward保证消息不会在网络中无限制的生存下去
  - Request-URI保证了请求目标不变

松散路由与严格路由

- 松散路由（Loose Routing）
  - A proxy is said to be loose routing if it follows the procedures defined in this specification for processing of the Route header field.
  - These procedures **separate** the destination of the request (present in the **Request-URI**) from the set of proxies that need to be visited along the way (present in the **Route** header field).
  - A proxy compliant to these mechanisms is also known as a loose router
- 严格路由（Strict Routing）
  - A proxy is said to be strict routing if it follows the Route processing rules of RFC 2543 and many prior work in progress versions of this RFC.
  - That rule caused proxies to **destroy** the contents of the **Request-URI** when a **Route** header field was **present**.
  - Strict routing behavior is not used in this specification, in favor of a loose routing behavior.
  - Proxies that perform strict routing are also known as strict routers.

松散路由与严格路由过程比较

alice@client.atlanta.example.com
ss1.atlanta.example.com

松散路由	严格路由
INVITE bob@biloxi.example.com To: bob@biloxi.example.com Route: ss1.atlanta.example.com	INVITE bob@biloxi.example.com To: bob@biloxi.example.com Route
接收端删除Route	发送端删除Route

ss1.atlanta.example.com
ss2.biloxi.example.com

松散路由	严格路由
INVITE bob@biloxi.example.com To:bob@biloxi.example.com Route-Request:ss1.atlanta.example.com	INVITE ss2.biloxi.example.com To:bob@biloxi.example.com Route-Request:ss1.atlanta.example.com
Request-URI不变	Request-URI修改为Proxy2

ss2.biloxi.example.com
bob@client.biloxi.example.com

松散路由	严格路由
INVITE bob@client.biloxi.example.com To: bob@biloxi.example.com Route-Request:ss2.biloxi.example.com Route-Request:ss1.atlanta.example.com	INVITE bob@client.biloxi.example.com To: bob@biloxi.example.com Route-Request:ss2.biloxi.example.com Route-Request:ss1.atlanta.example.com

松散路由与严格路由过程比较

↓ alice@client.atlanta.example.com
ss1.atlanta.example.com

松散路由	严格路由
ACK bob@client.biloxi.example.com To: bob@biloxi.example.com Route: ss1.atlanta.example.com Route: ss2.biloxi.example.com	ACK ss1.atlanta.example.com To: bob@biloxi.example.com Route Route: ss2.biloxi.example.com
Request-URI直接是Bob	为了支持发送端删除Route，Request-URI修改为TopRoute

↓ ss1.atlanta.example.com
ss2.biloxi.example.com

松散路由	严格路由
ACK bob@client.biloxi.example.com To:bob@biloxi.example.com Route: ss2.biloxi.example.com	ACK ss2.biloxi.example.com To: bob@biloxi.example.com Route
Request-URI不变	Request-URI修改为Proxy2

↓ ss2.biloxi.example.com
bob@client.biloxi.example.com

松散路由	严格路由
ACK bob@client.biloxi.example.com To: bob@biloxi.example.com	ACK bob@client.biloxi.example.com To: bob@biloxi.example.com

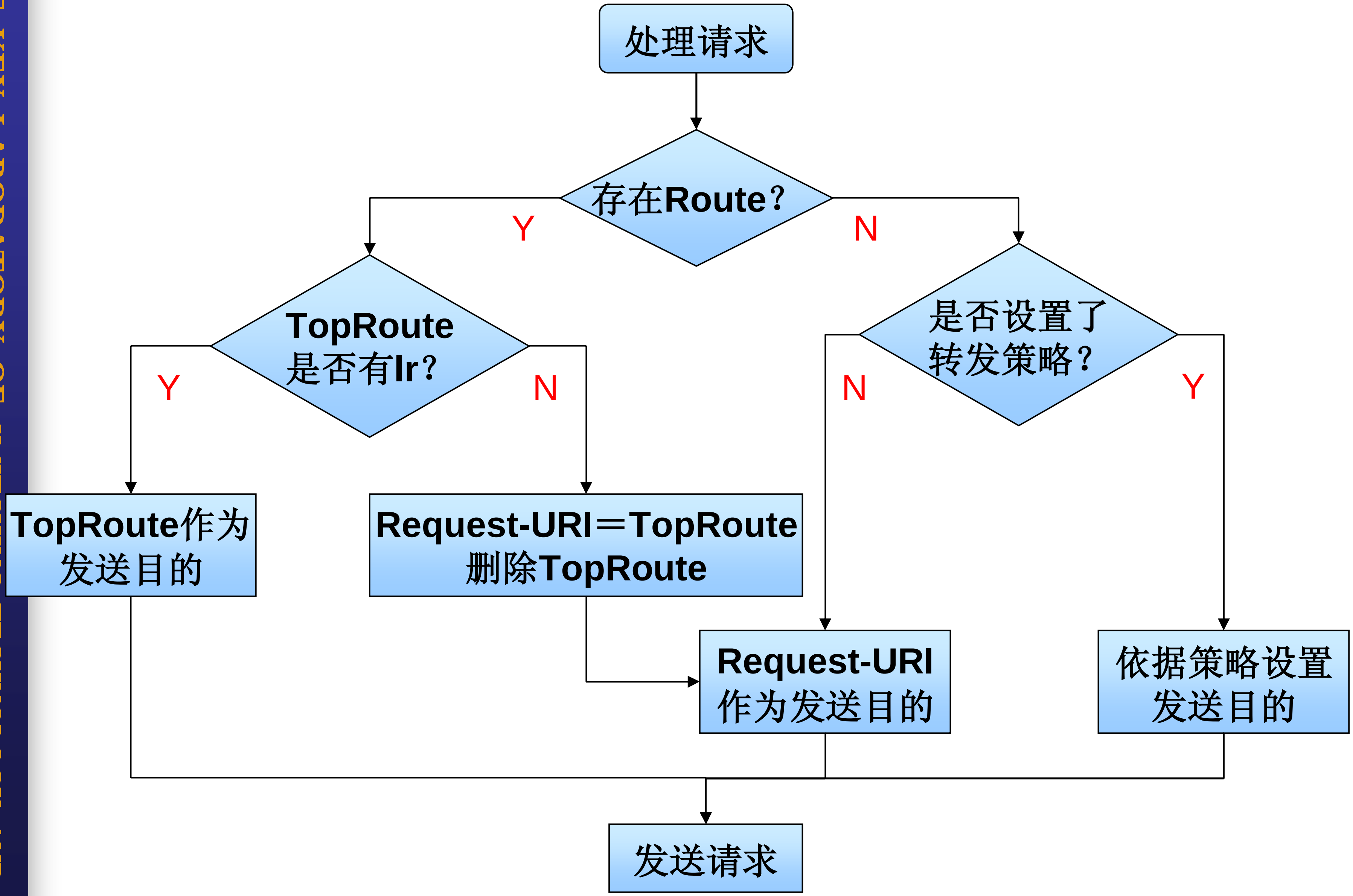
松散路由与严格路由的区别

- 松散路由
  - 如果请求中Route存在，则依据Route路由
  - 如果请求中Route不存在，则依据Request-URI路由
  - 在整个路由路径中，Request-URI不变，指向最终目的
  - 后继请求使用Contact作为Request-URI，到达最终目标控制域之后仍然可以依据Request-URI直接转发
- 严格路由
  - 请求依据Request-URI路由
  - 如果请求中存在Route，则使用Route修改Request-URI
  - 在整个路由路径中，Request-URI指向下一跳地址
  - 后继请求即便使用Contact作为Request-URI，在到达最终目标控制域的时候，依然只能使用To指示路由目标，可能还需要进行一次路由查询
- 松散路由的优点
  - 谁能最准确的确定目标？
  - Request-URI ＝ To？
  - 实际From由 Client请求中的Contact 标识
  - 实际To由 Server最终响应中的Contact标识
  - From-To-CallID演变为 Dialog标识

目标所属的控制域

Request-URI可能包含更丰富的信息

兼容lr与sr的请求消息路由处理过程



组网与路由小结

- 组网
- 路由过程
- 松散路由
- 严格路由
- 兼容松散路由与严格路由

严格路由 — B2BUA？

会话建立与媒体协商

- 会话建立的两层含义
  - 实体之间逻辑连接关系的建立
  - 实体之间媒体流的建立
- SIP协议完成的是“会话”建立吗
  - SIP协议完成的是：维持Pear-Pear之间的逻辑连接关系
  - SIP协议本身不关心媒体流的建立
- SIP协议“族”中有媒体建立与协商协议吗？
  - SDP：Session Description Protocol
    - SDP is intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.
    - SDP完成的仅仅是如何描述Multimedia Sessions

SIP协议是一个满足会话控制与媒体控制分离的协议吗

H.323协议的媒体控制

- **H.245 — 媒体控制协议**
- **H.245 session**:
  - The part of the call that begins with the establishment of an H.245 Control Channel and ends with the receipt of the H.245 EndSessionCommand or termination due to failure.
  - Not to be confused with a call, which is delineated by the H.225.0 Setup and Release Complete messages.
- H.245的主要目的是
  - Master/slave determination 主从确定
  - Capability Exchange 能力交换
  - Logical Channel Signalling 逻辑通道信令
  - Bidirectional Logical Channel Signalling
  - Close Logical Channel Signalling
  - Mode Request
  - Round Trip Delay Determination
  - Maintenance Loop Signalling

IP网络上需要媒体控制协议吗

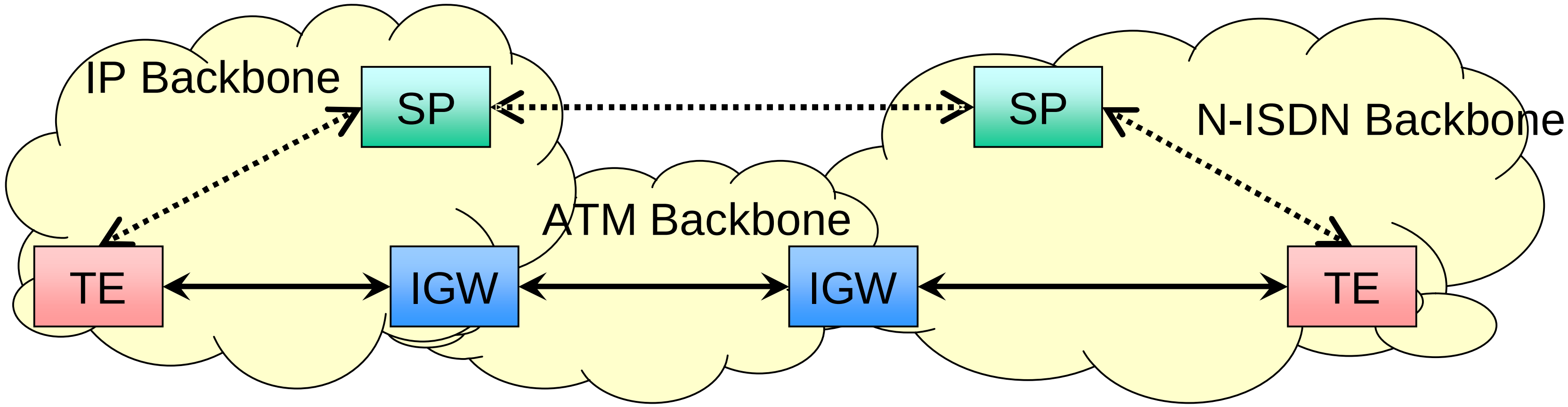
- IP网络上需要媒体控制协议吗
  - IP网络连接并进行数据传输的基本要求
    - TCP/UDP+IP地址+端口号 — 主从决定
    - 媒体类型描述与协商 — 能力交换
  - IP网络连接并进行数据传输的扩展要求
    - 实时媒体传输与控制 — RTP/RTCP

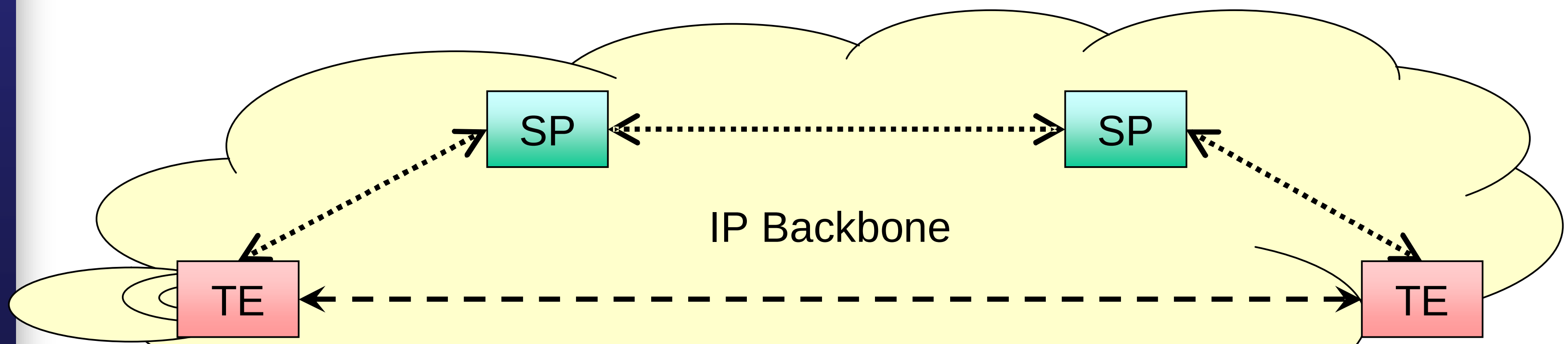
需要吗?

- H.323基于IP网络，为什么还有媒体控制协议
  - 在ITU-T的场景下，要适应广义的网络
  - H.3xx系列协议并不仅仅针对IP网络
  - H.245也不仅仅针对IP网络的媒体连接控制

需要吗?

媒体控制协议与媒体控制能力





媒体能力的协商语义 走信令通道还是走媒体通道还有区别吗

SIP协议是一个满足会话控制与媒体控制分离的协议吗

SDP

- Session name and purpose
- Time(s) the session is active
  - An arbitrary list of start and stop times bounding the session
  - For each bound, repeat times such as "every Wednesday at 10am for one hour"
- The media comprising the session
  - The type of media (video, audio, etc)
  - The transport protocol (RTP/UDP/IP, H.320, etc)
  - The format of the media (H.261 video, MPEG video, etc)
  - Multicast
    - Multicast address for media
    - Transport Port for media
  - Unicast
    - Remote address for media
    - Transport port for contact address
- Information to receive those media (addresses, ports, formats and so on)
- Information about the bandwidth to be used by the conference
- Contact information for the person responsible for the session

SDP描述

- **Session description**
  - **v**= (protocol version)
  - **o**= (owner/creator and session identifier).
    - o=<username> <session id> <version> <network type> <address type> <address>
  - **s**= (session name)
  - **c**= (connection information - not required if included in all media)
    - c=<network type> <address type> <connection address>
- **Time description**
  - **t**= (time the session is active)
    - t=<start time> <stop time>
- **Media description**
  - **m**= (media name and transport address)
    - m=<media> <port> <transport> <fmt list>
    - m=<media> <port>/<number of ports> <transport> <fmt list>
  - **a**=* (zero or more media attribute lines)
    - a=recvonly / sendrecv / sendonly / inactive
    - a=fmtp:<format> <format specific parameters>

SIP如何携带SDP

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice  1234  4321  IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly
```

消息体的类型
Application/sdp

消息体的长度

Username、session id、version、network type、address type、address

media、port、transport、fmt list

fmtp: format、format specific Parameters

SDP消息体

SDP是作为净荷承载的，SIP并不关心SDP所要表达的内容

SDP与Offer-Answer模型

Offer

```
v=0
o=alice 1234 1234 IN IP4 ua1.com.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 62986 RTP/AVP 0 4 18
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
a=rtpmap:18 G729/8000
a=sendrecv
m=video 51372 RTP/AVP 31
a=rtpmap:31 H261/90000
m=video 53000 RTP/AVP 32
a=rtpmap:32 MPV/90000
```

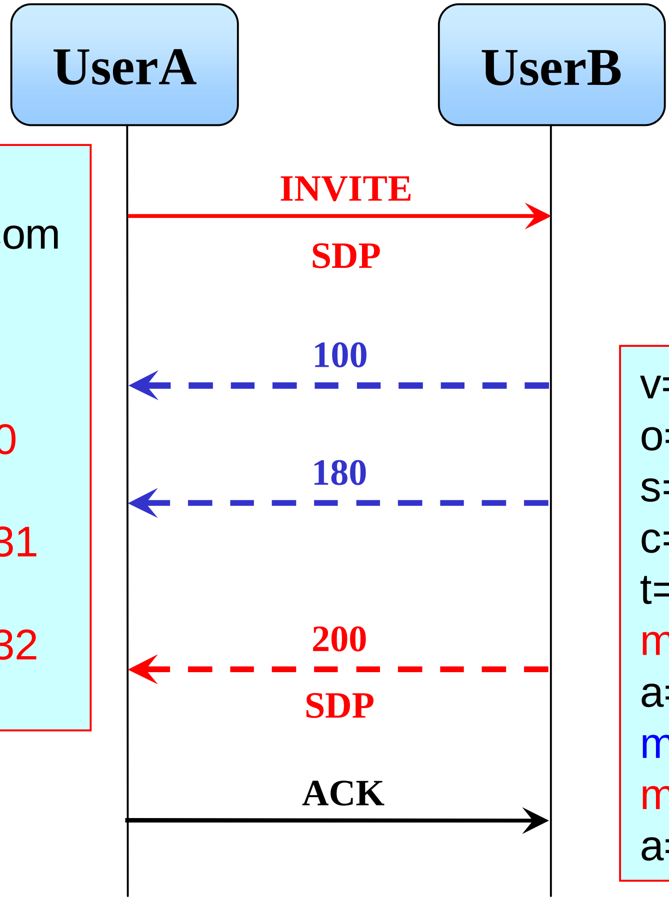
Answer

```
v=0
o=bob 5678 5678 IN IP4 ua2.com.cn
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 49920 RTP/AVP 0 4
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
a=sendonly
m=video 0 RTP/AVP 31
m=video 53000 RTP/AVP 32
a=rtpmap:32 MPV/90000
```

SIP协议与Offer-Answer模型

Offer

v=0
o=usera 1 2 IN IP4 usera.com
s=
c=IN IP4 usera.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
m=video 51372 RTP/AVP 31
a=rtpmap:31 H261/90000
m=video 53000 RTP/AVP 32
a=rtpmap:32 MPV/90000



Answer

v=0
o=userb 2 3 IN IP4 userb.com
s=
c=IN IP4 userb.com
t=0 0
m=audio 49920 RTP/AVP 0
a=rtpmap:0 PCMU/8000
m=video 0 RTP/AVP 31
m=video 53000 RTP/AVP 32
a=rtpmap:32 MPV/90000

SDP是呈现给处理SDP的终端／网关的，SIP并不关心Offer-Answer的过程

SDP一定是承载在SIP的可靠消息中的

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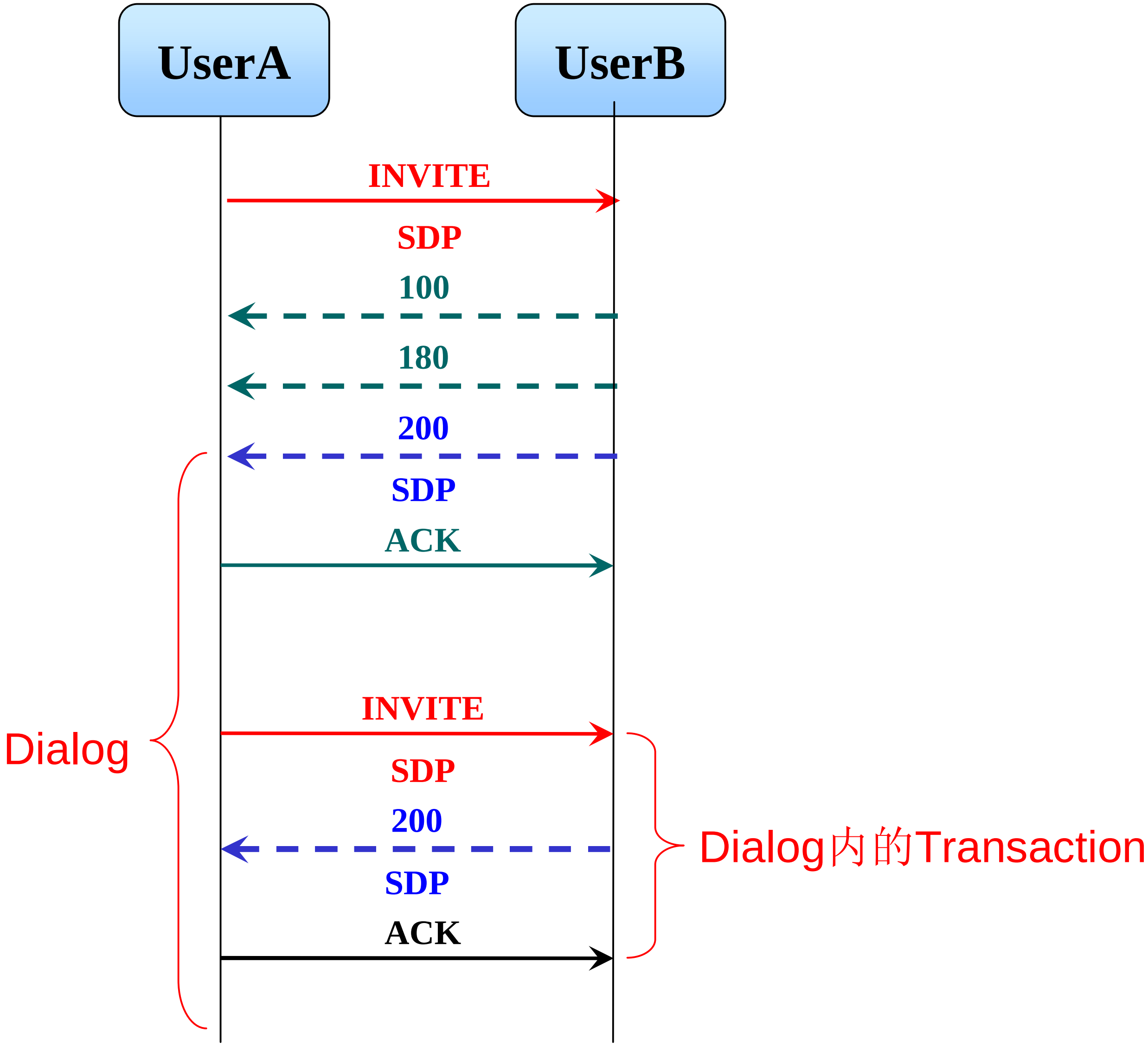
会话过程中的媒体修改



会话建立与媒体协商相关协议

- RFC2327－SDP: Session Description Protocol
- RFC3264－An Offer/Answer Model with the Session Description Protocol (SDP)

会话建立与媒体协商小结

- SIP不是媒体建立 / 协商协议
- SDP的定义
- SDP如何描述会话
- Offer-Answer过程
- SDP与SIP的关系