北京邮电大学
现代交换原理实验报告

实验名称：	SIP 信令实验
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0. 目录

1. 实验目的

结合课堂所讲的 SIP 信令工作流程，对软电话呼叫的信令进行抓包分析，理解 VoIP 呼叫中会话信令、媒体协商信令的作用，加深对 VoIP 的理解。

2. 实验内容

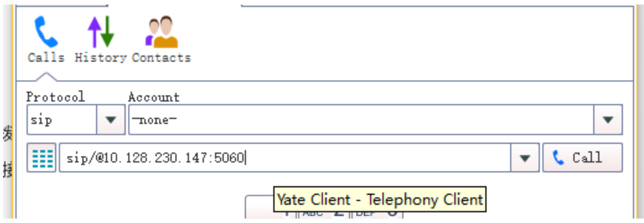
自己找一个软件拨打一通 SIP 电话，通过抓包软件观看 SIP 流程,通过两台电脑的 IP 地址即可

3. 实验环境

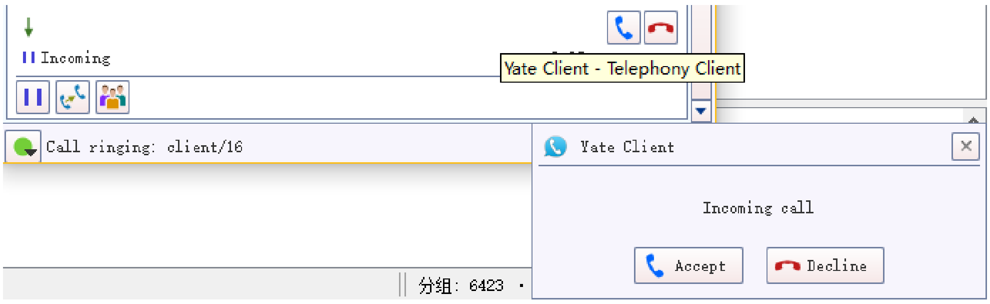
- Windows 10
- 软终端：Yate-6.0.0-1
- 分析工具：WireShark
- 通话双方 IP:
  主叫：10.128.236.68
  被叫：10.128.230.147

4. 实验步骤

1. 被叫：打开 WireShark,准备抓取报文
2. 主叫：在 YateClient 输入框中输入：sip/@10.128.230.147:5060 点击拨打，进行呼叫



3. 被叫：YateClient 收到呼叫，点击 Accpet 接受请求
4. 通话完成：任一方点击挂断图标结束通话



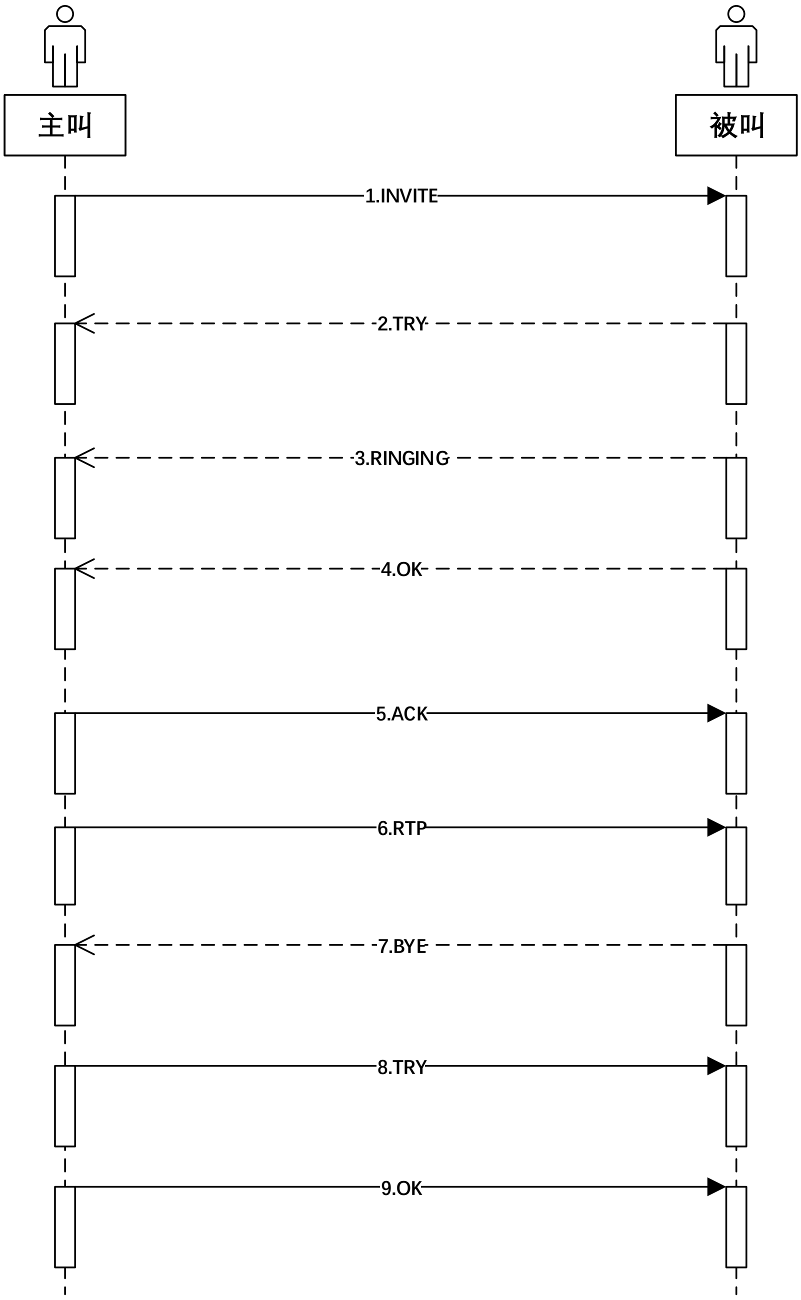
5. 结束抓取报文，保存文件

5. SIP 报文抓取实例分析

5.1 总体流程

sip

No.	Time	Source	Destination	Protocol	Length	Info
313	43.119358	10.128.236.68	10.128.230.147	SIP/SDP	947	Request: INVITE sip:10.128.230.147 |
316	43.154881	10.128.230.147	10.128.236.68	SIP	316	Status: 100 Trying |
317	43.188856	10.128.230.147	10.128.236.68	SIP	416	Status: 180 Ringing |
340	44.987501	10.128.230.147	10.128.236.68	SIP/SDP	952	Status: 200 OK |
343	45.026038	10.128.236.68	10.128.230.147	SIP	375	Request: ACK sip:10.128.230.147:5060 |
6236	102.361498	10.128.230.147	10.128.236.68	SIP	442	Request: BYE sip:10.128.236.68:5060 |
6241	102.415522	10.128.236.68	10.128.230.147	SIP	331	Status: 100 Trying |
6242	102.415523	10.128.236.68	10.128.230.147	SIP	429	Status: 200 OK |



5.2 基本过程

5.2.1 INVITE

```
Session Initiation Protocol (INVITE)
  Request-Line: INVITE sip:10.128.230.147 SIP/2.0
    Method: INVITE
    Request-URI: sip:10.128.230.147
    [Resent Packet: False]
  Message Header
    Max-Forwards: 20
    Via: SIP/2.0/UDP 10.128.236.68:5060;rport;branch=z9hG4bK396265751
    From: <sip:10.128.236.68>;tag=897616244
    To: <sip:10.128.230.147>
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    CSeq: 6 INVITE
    User-Agent: YATE/6.0.0
    Contact: <sip:10.128.236.68:5060>
    Allow: ACK, INVITE, BYE, CANCEL, OPTIONS, INFO
    Content-Type: application/sdp
    Content-Length: 506
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): yate 1558275733 1558275733 IN IP4 10.128.236.68
      Session Name (s): SIP Call
      Connection Information (c): IN IP4 10.128.236.68
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 31170 RTP/AVP 0 8 3 11 98 97 102 10
      Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a): rtpmap:8 PCMA/8000
      Media Attribute (a): rtpmap:3 GSM/8000
      Media Attribute (a): rtpmap:11 L16/8000
      Media Attribute (a): rtpmap:98 iLBC/8000
      Media Attribute (a): fmtp:98 mode=20
```

- Request-Line: INVITE sip:10.128.230.147 SIP/2.0　　//主叫向被叫发起呼叫请求，SIP 版本：2.0

  Method: INVITE　　//INVITE、ACK、CANCEL、OPTIONS、BYE、REGISTER

  Request-URI: sip:10.128.230.147　　//指示被邀请用户的当前地址

- Message Header

  Via: SIP/2.0/UDP 10.128.236.68:5060;rport;branch=z9hG4bK396265751 //通过主叫 10.128.236.68:5060

  From: <sip:10.128.236.68>;tag=897616244　　//可以是用户名或 URI

  To: <sip:10.128.230.147>

  Call-ID: 1040351045@10.128.236.68　　// 用来将消息分组的唯一标识，表示一个呼叫或对话

  CSeq: 6 INVITE　　// 消息的序列号，区分新消息和重试消息

  User-Agent: YATE/6.0.0

  Contact: <sip:10.128.236.68:5060>　　//实际地址

  Allow: ACK, INVITE, BYE, CANCEL, OPTIONS, INFO

  Content-Type: application/sdp　　//消息实体类型:SDP

  Content-Length: 506　　//消息实体长度

- Message Body

  Session Description Protocol Version (v): 0 //SDP 协议版本

  Owner/Creator, Session Id (o): yate 1558275733 1558275733 IN IP4 10.128.236.68 //创建者

  Session Name (s): SIP Call //会话名称

  Connection Information (c): IN IP4 10.128.236.68

  Time Description, active time (t): 0 0

  Media Description, name and address (m): audio 31170 RTP/AVP 0 8 3 11 98 97 102 103 104 105 … //媒体名称和传输地址

  //编解码器列表省略

  主叫向被叫发送了一个 SIP 请求 INVITE，表示主叫希望与被叫进行会话；此请求包含了语音流协议的详细信息。为此，在消息体中使用了会话描述协议 SDP。SDP 消息包含了主叫支持的所有媒体的编解码列表（这些编解码器使用 RTP 实时协议传输）

5.2.2 TRYING

```
Session Initiation Protocol (100)
  Status-Line: SIP/2.0 100 Trying
  Message Header
    Via: SIP/2.0/UDP 10.128.236.68:5060;rport=5060;branch=z9hG4bK396265751;received=10.128.2
    From: <sip:10.128.236.68>;tag=897616244
    To: <sip:10.128.230.147>
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    CSeq: 6 INVITE
    Server: YATE/6.0.0
    Content-Length: 0
```

- Status-Line: SIP/2.0 100 Trying //被叫尝试读取主叫请求，然后告诉主叫已经收到请求

5.2.3 RINGING

```
Status-Line: SIP/2.0 180 Ringing
    Status-Code: 180
    [Resent Packet: False]
    [Request Frame: 313]
    [Response Time (ms): 69]
Message Header
    Via: SIP/2.0/UDP 10.128.236.68:5060;rport=5060;branch=z9hG4bK396265751;received=10.128.2
    From: <sip:10.128.236.68>;tag=897616244
    To: <sip:10.128.230.147>;tag=1791049864
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    CSeq: 6 INVITE
    Server: YATE/6.0.0
    Contact: <sip:10.128.230.147:5060>
    Allow: ACK, INVITE, BYE, CANCEL, OPTIONS, INFO
    Content-Length: 0
```

- Status-Line: SIP/2.0 180 Ringing //电话响铃时，被叫向主叫发送临时消息以避免超时和放弃

5.2.4 OK

```
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 OK
  Message Header
    Via: SIP/2.0/UDP 10.128.236.68:5060;rport=5060;branch=z9hG4bK396265751;received=10.128
    From: <sip:10.128.236.68>;tag=897616244
    To: <sip:10.128.230.147>;tag=1791049864
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    CSeq: 6 INVITE
    Server: YATE/6.0.0
    Contact: <sip:10.128.230.147:5060>
    Allow: ACK, INVITE, BYE, CANCEL, OPTIONS, INFO
    Content-Type: application/sdp
    Content-Length: 508
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): yate 1558275737 1558275737 IN IP4 10.128.230.147
      Session Name (s): SIP Call
      Connection Information (c): IN IP4 10.128.230.147
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 20812 RTP/AVP 0 8 3 11 98 97 102 103
      Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a): rtpmap:8 PCMA/8000
      Media Attribute (a): rtpmap:3 GSM/8000
      Media Attribute (a): rtpmap:11 L16/8000
      Media Attribute (a): rtpmap:98 iLBC/8000
      Media Attribute (a): fmtp:98 mode=20
      Media Attribute (a): rtpmap:97 iLBC/8000
      Media Attribute (a): fmtp:97 mode=30
      Media Attribute (a): rtpmap:102 SPEEX/8000
      Media Attribute (a): rtpmap:103 SPEEX/16000
```

- Status-Line: SIP/2.0 200 OK　//被叫接受呼叫

被叫决定接受呼叫；此时，被叫向主叫发送 OK 响应。在响应的消息体中，还有另一个 SDP 消息，它包含了主叫被叫均支持的一组媒体编解码器。200 类型的响应可以接受所有类型的 SIP 请求。

5.2.5 ACK

```
Session Initiation Protocol (ACK)
  Request-Line: ACK sip:10.128.230.147:5060 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 10.128.236.68:5060;rport;branch=z9hG4bK1942891727
    From: <sip:10.128.236.68>;tag=897616244
    To: <sip:10.128.230.147>;tag=1791049864
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    CSeq: 6 ACK
    Max-Forwards: 20
    Contact: <sip:10.128.236.68:5060>
    User-Agent: YATE/6.0.0
    Content-Length: 0
```

- Request-Line: ACK sip:10.128.230.147:5060 SIP/2.0

主叫最后使用一条 ACK 消息进行确认

5.2.6 RTP 连接

- 通过端口 20812 和 31170 使用 SDP 消息中选择的方法建立连接

5.2.7 BYE

```
Session Initiation Protocol (BYE)
  Request-Line: BYE sip:10.128.236.68:5060 SIP/2.0
  Message Header
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    From: <sip:10.128.230.147>;tag=1791049864
      SIP from address: sip:10.128.230.147
      SIP from tag: 1791049864
    To: <sip:10.128.236.68>;tag=897616244
      SIP to address: sip:10.128.236.68
      SIP to tag: 897616244
    P-RTP-Stat: PS=2853,OS=456480,PR=2829,OR=452640,PL=0
    Via: SIP/2.0/UDP 10.128.230.147:5060;rport;branch=z9hG4bK391249970
    CSeq: 20 BYE
    User-Agent: YATE/6.0.0
    Max-Forwards: 70
    Allow: ACK, INVITE, BYE, CANCEL, OPTIONS, INFO
    Content-Length: 0
```

- Request-Line: BYE sip:10.128.236.68:5060 SIP/2.0 //会话结束，此处由被叫挂断挂断此时发送一条新的请求 BYE,可由任一方发送；此处由被叫发送

5.2.8 TRY

```
Session Initiation Protocol (100)
  Status-Line: SIP/2.0 100 Trying
  Message Header
    Via: SIP/2.0/UDP 10.128.230.147:5060;rport=5060;branch=z9hG4bK391249970;received=10.12
    From: <sip:10.128.230.147>;tag=1791049864
      SIP from address: sip:10.128.230.147
      SIP from tag: 1791049864
    To: <sip:10.128.236.68>;tag=897616244
      SIP to address: sip:10.128.236.68
      SIP to tag: 897616244
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    CSeq: 20 BYE
    Server: YATE/6.0.0
    Content-Length: 0
```

- 主叫读取被叫的挂断请求，但尚未对其进行处理，请被叫等待

5.2.9 OK

```
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 OK
  Message Header
    Via: SIP/2.0/UDP 10.128.230.147:5060;rport=5060;branch=z9hG4bK391249970;received=10.12
    From: <sip:10.128.230.147>;tag=1791049864
      SIP from address: sip:10.128.230.147
      SIP from tag: 1791049864
    To: <sip:10.128.236.68>;tag=897616244
      SIP to address: sip:10.128.236.68
      SIP to tag: 897616244
    Call-ID: 1040351045@10.128.236.68
    [Generated Call-ID: 1040351045@10.128.236.68]
    CSeq: 20 BYE
    P-RTP-Stat: PS=2870,OS=459200,PR=2809,OR=449440,PL=5
    Server: YATE/6.0.0
    Allow: ACK, INVITE, BYE, CANCEL, OPTIONS, INFO
    Content-Length: 0
```

- 主叫接受 BYE 请求，使用 OK 回复，呼叫断开

6. 实验结果

实验后，我们可以通过抓包分析看到一次 SIP 的完整过程；对不同包的每个字段进行分析，将一次呼叫过程拆分为客户端间的多次消息交流

7. 实验小结

此次实验结合课堂所讲的 SIP 信令工作流程，对软电话呼叫的信令进行抓包分析，理解了 VoIP 呼叫中会话信令、媒体协商信令 SDP 的作用，加深了对 VoIP 的理解