《现代交换原理》实验报告

实验名称 ______拨打 SIP 电话 ______

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一、 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。

二、 实验内容和实验步骤

1. SIP服务端软件的安装

本次我们选用的SIP服务端为Flexisip，其提供了较为简单的搭建SIP服务器的方式，而且其配套客户端UI比较好看。

1) 准备工作

本次实验的服务端我们选择在阿里云主机上进行搭建，操作系统为Ubuntu 18.04 LTS。

2) 软件安装

首先增加对应的软件源。

在/etc/apt/source.list上增加如下代码：

```
# For Ubuntu 18.04 LTS
deb [arch=amd64] http://linphone.org/snapshots/ubuntu bionic stable beta # alpha
```

之后需要信任软件源，添加PGP Key

```
wget https://www.linphone.org/snapshots/ubuntu/pubkey.gpg -O - | sudo apt-key add -
```

之后即可直接安装

```
sudo apt update

sudo apt install bc-flexisip
```

3) 软件配置

生成默认配置文件

使用如下指令生成默认配置文件，以 root 权限执行如下指令：

```
/opt/belledonne-communications/bin/flexisip --dump-default all >
/etc/flexisip/flexisip.conf
```

编写配置文件

之后我们打开 /etc/flexisip/flexisip.conf，删除全部内容，把如下配置复制到配置文件中：

```
[global]
debug=true

aliases=sip.name1e5s.fun


[module::Registrar]
enabled=true
reg-domains=sip.name1e5s.fun
db-implementation=internal

[module::Authentication]
enabled=true
auth-domains=sip.name1e5s.fun
db-implementation=file
datasource=/etc/flexisip/users.db.txt
```

这里面 sip.name1e5s.fun 为个人域名。

之后创建 /etc/flexisip/users.db.txt，输入账户配置：

```
version:1

test@sip.name1e5s.fun clrtxt:test ;
test2@sip.name1e5s.fun clrtxt:test ;
```

其中 test 为用户 ID， clrtxt: 后，空格前的为密码。

4) 运行

以 root 权限执行如下指令：

```
/opt/belledonne-communications/bin/flexisip
```

至此，服务端配置成功，可以开始进行通话。

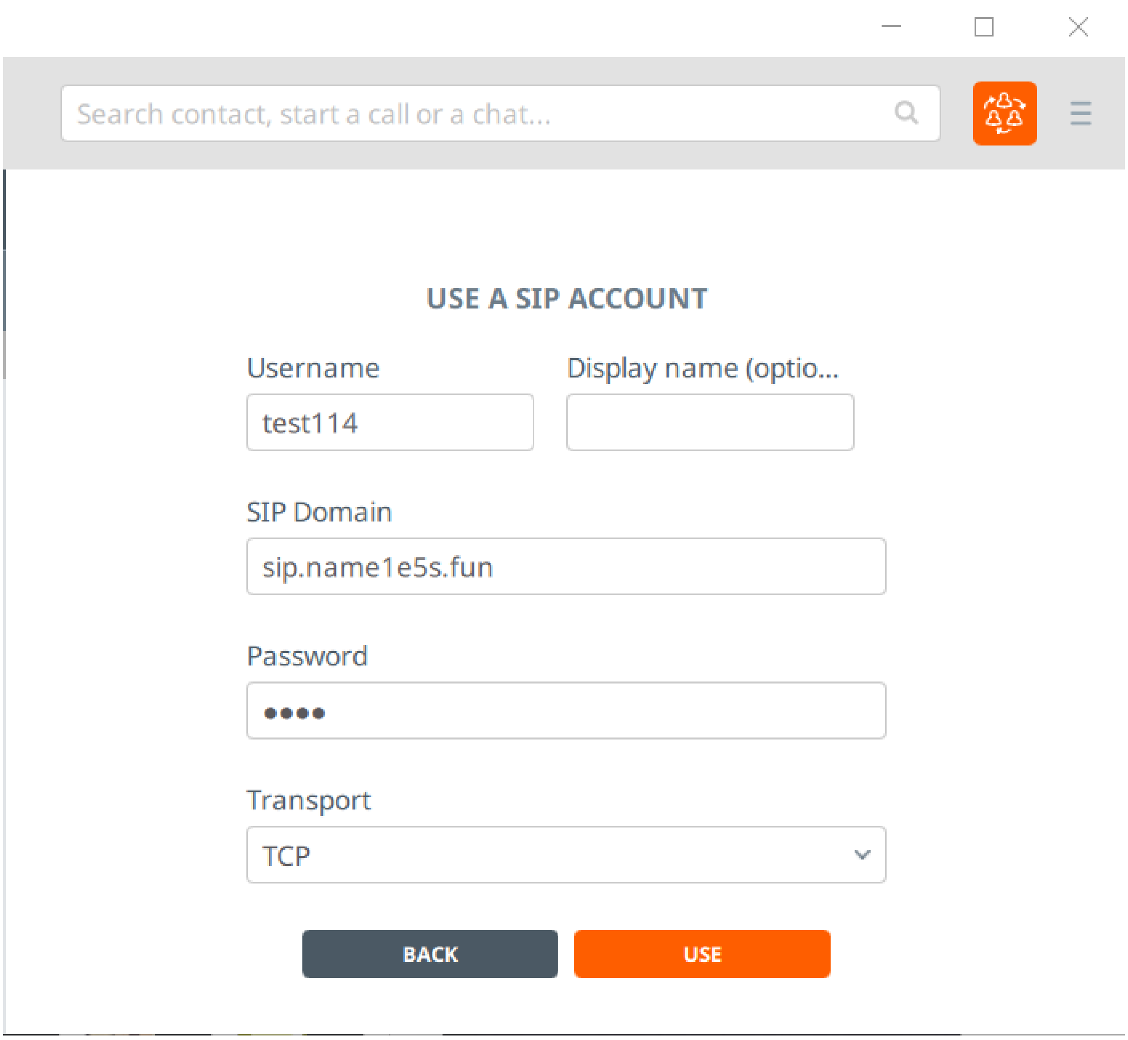
2. SIP 客户端软件的安装

1) 软件安装

这里我们选用的 SIP 客户端为 Linphone，其特点为 UI 比较现代，而且为 GPLv3 授权下 的 Free Software。其安装过程十分简单，下载后一路下一步即可在此不再过多介绍。

2) 用户注册

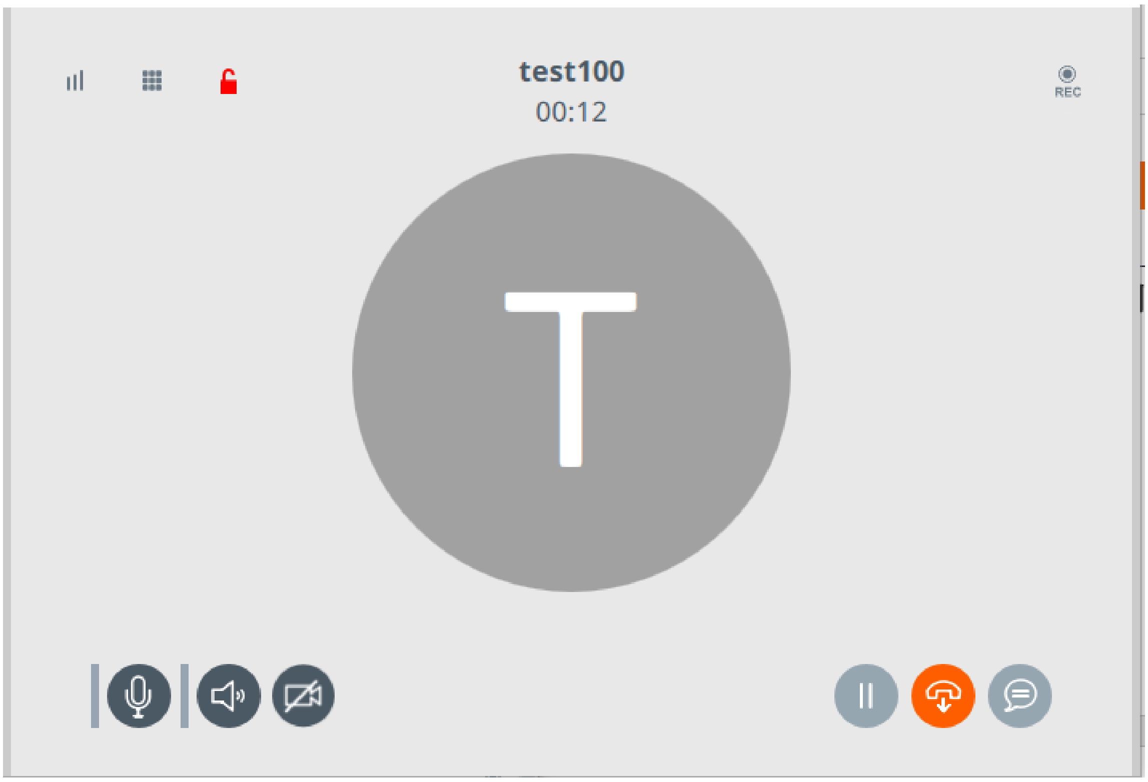
打开软件后，我们创建新的 SIP 账号，按照如下方式输入账号信息即可：



之后选择 “USE”，并在左上角选择这一用户。当注意到用户名前出现绿点时，表示与服务端链接成功，可以开始通话。

3) 通话测试

在最上方输入想要拨打的账号，即可开始拨打电话。通话页面如下：



Linphone 支持视频通话，但是碍于带宽等原因，我们没有进行测试。

我们也使用 MicroSIP 以确认服务器的兼容性，其安装使用在此不表。

3. 抓包分析 SIP 流程

1) 注册

注册流程如下:

1. 终端向服务器发起 REGISTER 请求

```
Session Initiation Protocol (REGISTER)
  Request-Line: REGISTER sip:sip.name1e5s.fun;transport=tcp SIP/2.0
    Method: REGISTER
    Request-URI: sip:sip.name1e5s.fun;transport=tcp
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP 192.168.101.13:51539;rport;branch=z9hG4bKPjdf6b976868974a9890739952a8ab77d7;alias
    Max-Forwards: 70
    From: <sip:test0@sip.name1e5s.fun>;tag=9dbbdc28b76c437dbad5f569ec4652ff
    To: <sip:test0@sip.name1e5s.fun>
    Call-ID: 610a33e7ad4c48e09b78ddbffef967ae
    [Generated Call-ID: 610a33e7ad4c48e09b78ddbffef967ae]
    CSeq: 14263 REGISTER
    User-Agent: MicroSIP/3.19.30
    Supported: outbound, path
```

2. 服务器返回 401，要求进行安全认证

```
Session Initiation Protocol (401)
  Status-Line: SIP/2.0 401 Unauthorized
    Status-Code: 401
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP 192.168.101.13:51539;rport=51539;branch=z9hG4bKPjdf6b976868974a9890739952a8ab7
    From: <sip:test0@sip.name1e5s.fun>;tag=9dbbdc28b76c437dbad5f569ec4652ff
    To: <sip:test0@sip.name1e5s.fun>;tag=bf8638324618dc61059d4c604476fea1.abf45b37
    Call-ID: 610a33e7ad4c48e09b78ddbffef967ae
    [Generated Call-ID: 610a33e7ad4c48e09b78ddbffef967ae]
    CSeq: 14263 REGISTER
    WWW-Authenticate: Digest realm="sip.name1e5s.fun", nonce="Xszd2V7M3K3NVuvcObmcUxWU+wGbVZ95"
    Server: kamailio (5.3.4 (x86_64/linux))
    Content-Length: 0
```

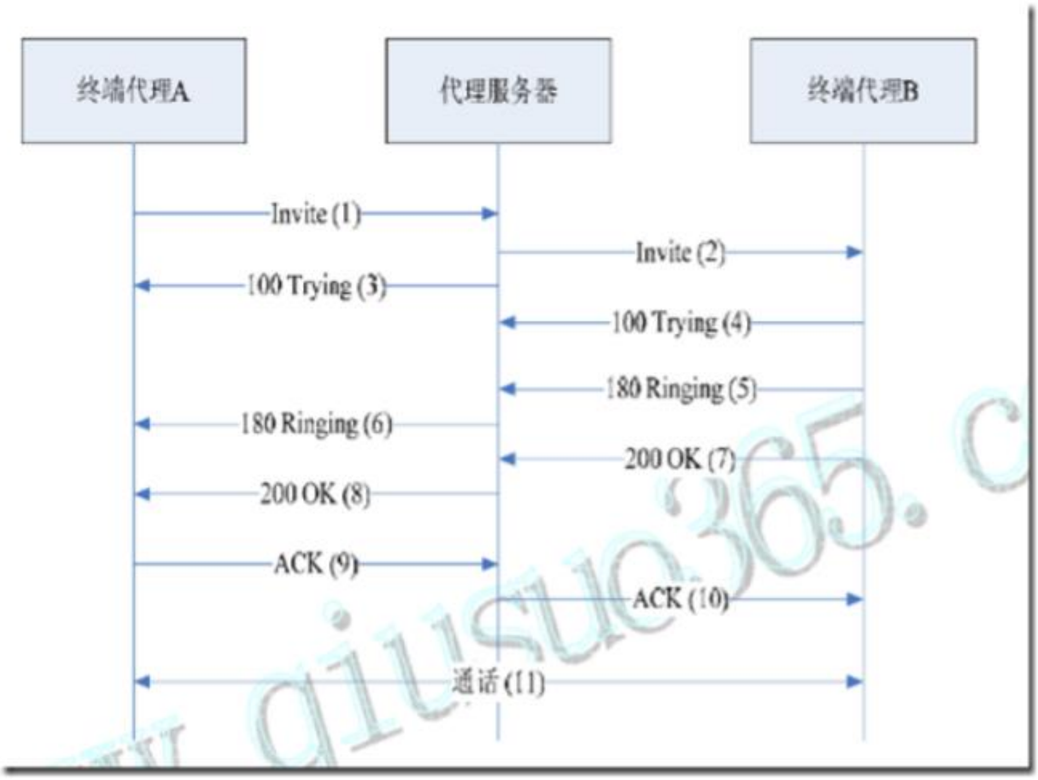
3. 服务端按照要求加密用户信息，重新 REGISTER

```
Session Initiation Protocol (REGISTER)
  Request-Line: REGISTER sip:sip.name1e5s.fun;transport=tcp SIP/2.0
    Method: REGISTER
    Request-URI: sip:sip.name1e5s.fun;transport=tcp
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP 192.168.101.13:51539;rport;branch=z9hG4bKPj38c219834a0d482fba776bb29a09bd23;alias
    Max-Forwards: 70
    From: <sip:test0@sip.name1e5s.fun>;tag=9dbbdc28b76c437dbad5f569ec4652ff
    To: <sip:test0@sip.name1e5s.fun>
    Call-ID: 610a33e7ad4c48e09b78ddbffef967ae
    [Generated Call-ID: 610a33e7ad4c48e09b78ddbffef967ae]
    CSeq: 14264 REGISTER
    User-Agent: MicroSIP/3.19.30
    Supported: outbound, path
```

4. 服务器进行认证，返回 200

```
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 OK
    Status-Code: 200
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP 192.168.101.13:51539;rport=51539;branch=z9hG4bKPj38c219834a0d482fba776bb29a09bd23;alias;received=192.168.101.13
    From: <sip:test0@sip.name1e5s.fun>;tag=9dbbdc28b76c437dbad5f569ec4652ff
    To: <sip:test0@sip.name1e5s.fun>;tag=bf8638324618dc61059d4c604476fea1.abf45b37
    Call-ID: 610a33e7ad4c48e09b78ddbffef967ae
    [Generated Call-ID: 610a33e7ad4c48e09b78ddbffef967ae]
    CSeq: 14264 REGISTER
    Contact: <sip:test0@192.168.101.13:51539;transport=TCP;ob>;expires=300;+sip.instance="<urn:uuid:00000000-0000-0000-0000-000027dd7554>";reg-
    Server: kamailio (5.3.4 (x86_64/linux))
    Content-Length: 0
```

2) 通话



A 用户视角

5111	176.189085	192.168.101.13	39.106.171.163	SIP/SDP	49	Request: INVITE sip:TianTian@sip.name1e5s.fun |
5112	176.214172	39.106.171.163	192.168.101.13	SIP	507	Status: 407 Proxy Authentication Required |
5113	176.222995	192.168.101.13	39.106.171.163	SIP	453	Request: ACK sip:TianTian@sip.name1e5s.fun |
5115	176.223207	192.168.101.13	39.106.171.163	SIP/SDP	245	Request: INVITE sip:TianTian@sip.name1e5s.fun |
5116	176.248816	39.106.171.163	192.168.101.13	SIP	375	Status: 100 trying -- your call is important to us |
5122	176.806151	39.106.171.163	192.168.101.13	SIP	515	Status: 180 Ringing |
5744	183.337112	39.106.171.163	192.168.101.13	SIP/SDP	1258	Status: 200 Ok |
5750	183.481740	192.168.101.13	39.106.171.163	SIP	673	Request: ACK sip:TianTian@223.104.103.76:54536;transport=tcp |

1. 发起 INVITE 信息

```
v Message Body
  v Session Description Protocol
      Session Description Protocol Version (v): 0
    > Owner/Creator, Session Id (o): yhx 3284 4004 IN IP4 192.168.101.13
      Session Name (s): Talk
    > Connection Information (c): IN IP4 192.168.101.13
    > Time Description, active time (t): 0 0
    > Session Attribute (a): ice-pwd:9977ebde0f31238471766a86
    > Session Attribute (a): ice-ufrag:008ffcec
    > Session Attribute (a): rtcp-xr:rcvr-rtt=all:10000 stat-summary=loss,dup,jitt,TTL voip-m
    > Media Description, name and address (m): audio 7078 RTP/AVPF 96 97 98 0 8 101 99 100
    > Connection Information (c): IN IP4 122.137.129.21
    > Media Attribute (a): rtpmap:96 opus/48000/2
    > Media Attribute (a): fmtp:96 useinbandfec=1
    > Media Attribute (a): rtpmap:97 speex/16000
    > Media Attribute (a): fmtp:97 vbr=on
    > Media Attribute (a): rtpmap:98 speex/8000
    > Media Attribute (a): fmtp:98 vbr=on
    > Media Attribute (a): rtpmap:101 telephone-event/48000
    > Media Attribute (a): rtpmap:99 telephone-event/16000
    > Media Attribute (a): rtpmap:100 telephone-event/8000
    > Media Attribute (a): candidate:1 1 UDP 2130706431 192.168.101.13 7078 typ host
    > Media Attribute (a): candidate:1 2 UDP 2130706430 192.168.101.13 7079 typ host
    > Media Attribute (a): candidate:2 1 UDP 1694498815 122.137.129.21 7078 typ srflx raddr 1
    > Media Attribute (a): candidate:2 2 UDP 1694498814 122.137.129.21 7079 typ srflx raddr 1
    > Media Attribute (a): rtcp-fb:* trr-int 5000
    > Media Attribute (a): rtcp-fb:* ccm tmmbr
      [Generated Call-ID: xga-M2AafP]
```

INVITE 信息中包含了本次通话的详细信息，如上图所示。由此可知我们的通话为语音通话，协议为 RTP。

2. 服务端回复 TRYING

```
Session Initiation Protocol (100)
  Status-Line: SIP/2.0 100 trying -- your call is important to us
    Status-Code: 100
    [Resent Packet: False]
    [Request Frame: 5115]
    [Response Time (ms): 25]
  Message Header
    Via: SIP/2.0/UDP 192.168.101.13:5060;branch=z9hG4bK.kx0LUNoPU;rport=5060;received=192.168.101.13
    From: <sip:yhx@sip.name1e5s.fun>;tag=-rU0F3h1R
    To: sip:TianTian@sip.name1e5s.fun
    CSeq: 21 INVITE
    Call-ID: r~bgZLDT~b
```

3. 服务端回复 RINGING

```
    [Resent Packet: False]
    [Request Frame: 5115]
    [Response Time (ms): 583]
  Message Header
    Via: SIP/2.0/UDP 192.168.101.13:5060;received=192.168.101.13;branch=z9hG4bK.kx0LUNoPU;rport=5060
    From: <sip:yhx@sip.name1e5s.fun>;tag=-rU0F3h1R
    To: <sip:TianTian@sip.name1e5s.fun>;tag=0~-NuF-
    Call-ID: r~bgZLDT~b
    [Generated Call-ID: r~bgZLDT~b]
    CSeq: 21 INVITE
    User-Agent: Linphone Desktop/4.1.1 (belle-sip/1.6.3)
    Supported: replaces, outbound
    Record-route: <sip:sip.name1e5s.fun:5060;transport=tcp;r2=on;lr>
    Record-route: <sip:sip.name1e5s.fun;r2=on;lr>
    Content-Length: 0
```

此时，我们得知对面的客户端开始响铃。

4. 服务端回复 OK

此时，对面接起电话。

5. 回应 ACK

```
Session Initiation Protocol (ACK)
  Request-Line: ACK sip:TianTian@223.104.103.76:54536;transport=tcp SIP/2.0
    Method: ACK
    Request-URI: sip:TianTian@223.104.103.76:54536;transport=tcp
      Request-URI User Part: TianTian
      Request-URI Host Part: 223.104.103.76
      Request-URI Host Port: 54536
    [Resent Packet: False]
    [Request Frame: 5115]
    [Response Time (ms): 7258]
  Message Header
    Via: SIP/2.0/UDP 192.168.101.13:5060;rport;branch=z9hG4bK.R87UV35la
    From: <sip:yhx@sip.name1e5s.fun>;tag=-rU0F3h1R
```

此时，我们确认了我们已经得知消息。

B 用户视角

```
6538 382.793176   39.106.171.163   192.168.43.54    SIP/SDP   428 Request: INVITE sip:TianTian@223.104.103.76:54536;transport=tcp |
6540 382.805115   192.168.43.54    39.106.171.163   SIP       432 Status: 100 Trying |
6545 382.927039   192.168.43.54    39.106.171.163   SIP       645 Status: 180 Ringing |
6960 386.613332   192.168.43.54    39.106.171.163   SIP/SDP  1388 Status: 200 Ok |
6995 386.872775   39.106.171.163   192.168.43.54    SIP       728 Request: ACK sip:TianTian@223.104.103.76:54536;transport=tcp |
```

1. 收到 INVITE 信息

```
⌄ Message Body
  ⌄ Session Description Protocol
      Session Description Protocol Version (v): 0
    › Owner/Creator, Session Id (o): yhx 72 2003 IN IP4 122.137.129.21
      Session Name (s): Talk
    › Connection Information (c): IN IP4 122.137.129.21
    › Time Description, active time (t): 0 0
    › Session Attribute (a): ice-pwd:37b5e26bf62167561345f81d
    › Session Attribute (a): ice-ufrag:6d0e9b26
    › Session Attribute (a): rtcp-xr:rcvr-rtt=all:10000 stat-summary=loss,dup,jitt,TTL voip-metrics
    › Media Description, name and address (m): audio 7078 RTP/AVPF 96 97 98 0 8 101 99 100
    › Connection Information (c): IN IP4 122.137.129.21
    › Media Attribute (a): rtpmap:96 opus/48000/2
    › Media Attribute (a): fmtp:96 useinbandfec=1
    › Media Attribute (a): rtpmap:97 speex/16000
    › Media Attribute (a): fmtp:97 vbr=on
    › Media Attribute (a): rtpmap:98 speex/8000
    › Media Attribute (a): fmtp:98 vbr=on
    › Media Attribute (a): rtpmap:101 telephone-event/48000
    › Media Attribute (a): rtpmap:99 telephone-event/16000
    › Media Attribute (a): rtpmap:100 telephone-event/8000
    › Media Attribute (a): candidate:1 1 UDP 2130706431 192.168.101.13 7078 typ host
    › Media Attribute (a): candidate:1 2 UDP 2130706430 192.168.101.13 7079 typ host
    › Media Attribute (a): candidate:2 1 UDP 1694498815 122.137.129.21 7078 typ srflx raddr 192.168.101.13 rport 7078
    › Media Attribute (a): candidate:2 2 UDP 1694498814 122.137.129.21 7079 typ srflx raddr 192.168.101.13 rport 7079
    › Media Attribute (a): rtcp-fb:* trr-int 5000
    › Media Attribute (a): rtcp-fb:* ccm tmmbr
      [Generated Call-ID: IgEDpEj3~D]
      [Generated Call-ID: r~bgZLDT~b]
      [Generated Call-ID: 04m52Wu349]
      [Generated Call-ID: 36b6b9dd7d614da0972a1eddc56b7a22]
```

如图，我们收到了 A 发来的请求，包含了与 A 相同的通话信息。

2. 服务端回复 TRYING

```
Session Initiation Protocol (100)
⌄ Status-Line: SIP/2.0 100 Trying
    Status-Code: 100
    [Resent Packet: False]
⌄ Message Header
  › Via: SIP/2.0/TCP sip.name1e5s.fun:5060;received=39.106.171.163;branch=z9hG4bK5f91.08bd20a928319f65fe994b71ee7c2865.0
  › Via: SIP/2.0/UDP 122.137.129.21:5060;received=122.137.129.21;branch=z9hG4bK.bfPGUIS3y;rport=5060
  › From: <sip:yhx@sip.name1e5s.fun>;tag=8uOtpPqP2
  › To: sip:TianTian@sip.name1e5s.fun
    Call-ID: 04m52Wu349
    [Generated Call-ID: 04m52Wu349]
  › CSeq: 21 INVITE
    Content-Length: 0
```

3. 服务端回复 RINGING

```
Session Initiation Protocol (180)
  Status-Line: SIP/2.0 180 Ringing
    Status-Code: 180
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP sip.name1e5s.fun:5060;received=39.106.171.163;branch=z9hG4bK5f91.08bd20a928319f65fe994b71ee7c2865.0
    Via: SIP/2.0/UDP 122.137.129.21:5060;received=122.137.129.21;branch=z9hG4bK.bfPGUIS3y;rport=5060
    From: <sip:yhx@sip.name1e5s.fun>;tag=8uOtpPqP2
    To: <sip:TianTian@sip.name1e5s.fun>;tag=Ep0Iofz
    Call-ID: O4m52Wu349
    [Generated Call-ID: O4m52Wu349]
    CSeq: 21 INVITE
    User-Agent: Linphone Desktop/4.1.1 (belle-sip/1.6.3)
    Supported: replaces, outbound
    Record-route: <sip:sip.name1e5s.fun:5060;transport=tcp;r2=on;lr>
    Record-route: <sip:sip.name1e5s.fun;r2=on;lr>
    Content-Length: 0
```

开始响铃。

4. 服务端回复 OK

```
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 Ok
    Status-Code: 200
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP sip.name1e5s.fun:5060;received=39.106.171.163;branch=z9hG4bK5f91.08bd20a928319f65fe994b71ee7c2865.0
    Via: SIP/2.0/UDP 122.137.129.21:5060;received=122.137.129.21;branch=z9hG4bK.bfPGUIS3y;rport=5060
    From: <sip:yhx@sip.name1e5s.fun>;tag=8uOtpPqP2
    To: <sip:TianTian@sip.name1e5s.fun>;tag=Ep0Iofz
    Call-ID: O4m52Wu349
    [Generated Call-ID: O4m52Wu349]
    CSeq: 21 INVITE
    User-Agent: Linphone Desktop/4.1.1 (belle-sip/1.6.3)
    Supported: replaces, outbound
    Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO, UPDATE
    Contact: <sip:TianTian@223.104.103.76:54536;transport=tcp>;+sip.instance="<urn:uuid:cf5cf527-8b45-424d-8441-a05346171653>"
    Content-Type: application/sdp
    Content-Length: 500
    Record-route: <sip:sip.name1e5s.fun:5060;transport=tcp;r2=on;lr>
    Record-route: <sip:sip.name1e5s.fun;r2=on;lr>
  Message Body
```

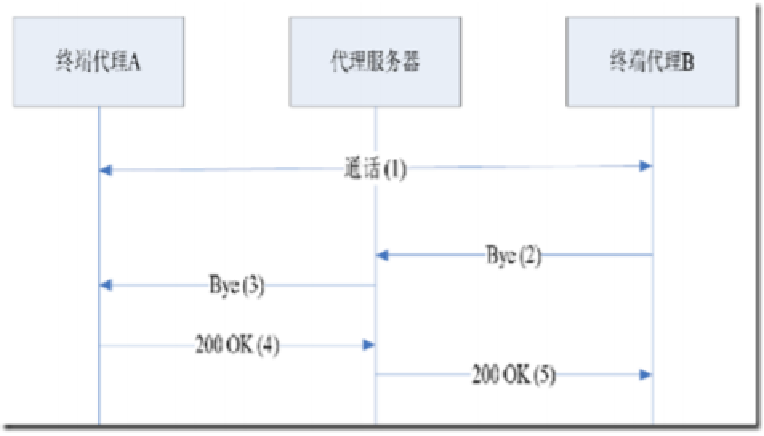
接起电话。

5. 回应 ACK

```
Session Initiation Protocol (ACK)
  Request-Line: ACK sip:TianTian@223.104.103.76:54536;transport=tcp SIP/2.0
    Method: ACK
    Request-URI: sip:TianTian@223.104.103.76:54536;transport=tcp
      Request-URI User Part: TianTian
      Request-URI Host Part: 223.104.103.76
      Request-URI Host Port: 54536
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP sip.name1e5s.fun:5060;branch=z9hG4bK5f91.3b56793100b4eb72a640d6dbc5ba4deb.0
    Via: SIP/2.0/UDP 122.137.129.21:5060;received=122.137.129.21;rport=5060;branch=z9hG4bK.ZS30EvRay
    From: <sip:yhx@sip.name1e5s.fun>;tag=8uOtpPqP2
    To: <sip:TianTian@sip.name1e5s.fun>;tag=Ep0Iofz
    CSeq: 21 ACK
    Call-ID: O4m52Wu349
    [Generated Call-ID: O4m52Wu349]
    Max-Forwards: 69
    Proxy-Authorization:  Digest realm="sip.name1e5s.fun", nonce="XszdWF7M3CzvhB1jlpS+3OFxLcKtjNSU", username="yhx",  uri="sip:TianTian@sip.name1e5s.fun", response="3bcf150333e8a793b5d3665
    User-Agent: Linphone Desktop/4.1.1 (belle-sip/1.6.3)
    Content-Length: 0
```

此时，我们收到 ACK,开始通话。

3) 挂断



A 用户视角

1) 收到挂断要求

```
Session Initiation Protocol (BYE)
  Request-Line: BYE sip:yhx@192.168.101.13:5060;transport=udp SIP/2.0
    Method: BYE
    Request-URI: sip:yhx@192.168.101.13:5060;transport=udp
      Request-URI User Part: yhx
      Request-URI Host Part: 192.168.101.13
      Request-URI Host Port: 5060
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/UDP sip.name1e5s.fun:5060;branch=z9hG4bK0ce1.859dd04d0a139ee3eb02bb807a01b36d.0;i=3
    Via: SIP/2.0/TCP 192.168.43.54:53797;received=223.104.103.76;branch=z9hG4bK.Z1z-5omzY;rport=54536
    From: <sip:TianTian@sip.name1e5s.fun>;tag=0~-NuF-
```

2) 回复 OK

```
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 Ok
    Status-Code: 200
    [Resent Packet: False]
    [Request Frame: 9304]
    [Response Time (ms): 8]
    [Release Time (ms): 8]
  Message Header
    Via: SIP/2.0/UDP sip.name1e5s.fun:5060;received=39.106.171.163;branch=z9hG4bK0ce1.859dd04d0a139ee3eb02bb807a01b3
    Via: SIP/2.0/TCP 192.168.43.54:53797;received=223.104.103.76;branch=z9hG4bK.Z1z-5omzY;rport=54536
    From: <sip:TianTian@sip.name1e5s.fun>;tag=0~-NuF-
    To: <sip:yhx@sip.name1e5s.fun>;tag=-rU0E3h1R
```

B 用户视角

1) 发起挂断要求

```
Session Initiation Protocol (BYE)
  Request-Line: BYE sip:TianTian@223.104.103.76:54536;transport=tcp SIP/2.0
    Method: BYE
    Request-URI: sip:TianTian@223.104.103.76:54536;transport=tcp
      Request-URI User Part: TianTian
      Request-URI Host Part: 223.104.103.76
      Request-URI Host Port: 54536
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP sip.name1e5s.fun:5060;branch=z9hG4bK2f91.33940cdd5e01b2d3b340f0682c5d294f.0
    Via: SIP/2.0/UDP 122.137.129.21:5060;received=122.137.129.21;branch=z9hG4bK.LVkuB3wHu;rport=5060
    From: <sip:yhx@sip.name1e5s.fun>;tag=8uOtpPqP2
    To: <sip:TianTian@sip.name1e5s.fun>;tag=Ep0Iofz
    CSeq: 22 BYE
    Call-ID: O4m52Wu349
    [Generated Call-ID: O4m52Wu349]
    Max-Forwards: 69
    User-Agent: Linphone Desktop/4.1.1 (belle-sip/1.6.3)
    Proxy-Authorization:  Digest realm="sip.name1e5s.fun", nonce="XszdWF7M3CzvhB1jlpS+3OFxLcKtjNSU", username="yhx",  uri="sip:TianTian@223.104.103.76:54536;transport=tcp", response="f350c
    Content-Length: 0
```

2) 收到 OK

```
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 Ok
    Status-Code: 200
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/TCP sip.name1e5s.fun:5060;received=39.106.171.163;branch=z9hG4bK2f91.33940cdd5e01b2d3b340f0682c5d294f.0
    Via: SIP/2.0/UDP 122.137.129.21:5060;received=122.137.129.21;branch=z9hG4bK.LVkuB3wHu;rport=5060
    From: <sip:yhx@sip.name1e5s.fun>;tag=8uOtpPqP2
    To: <sip:TianTian@sip.name1e5s.fun>;tag=Ep0Iofz
    Call-ID: O4m52Wu349
    [Generated Call-ID: O4m52Wu349]
    CSeq: 22 BYE
    User-Agent: Linphone Desktop/4.1.1 (belle-sip/1.6.3)
    Supported: replaces, outbound
    Content-Length: 0
```

此时，挂断成功。

4. 通话过程中

通话使用 RTP 协议，其中一个包如下：

```
> Internet Protocol Version 4, Src: 223.104.103.76, Dst: 192.168.101.13
> User Datagram Protocol, Src Port: 30601, Dst Port: 7078
v Real-Time Transport Protocol
  > [Stream setup by SDP (frame 1594)]
    10.. .... = Version: RFC 1889 Version (2)
    ..0. .... = Padding: False
    ...0 .... = Extension: False
    .... 0000 = Contributing source identifiers count: 0
    0... .... = Marker: False
    Payload type: opus (96)
    Sequence number: 1542
    [Extended sequence number: 67078]
    Timestamp: 817675027
    Synchronization Source identifier: 0xc6225b3f (3324140351)
    Payload: 78184d31e358f444351de3a90dc142a079254ee2a1b8ac87…
```

可见其为客户端间的直接通信，与服务端没有关系。

三、 实验总结与心得

于海鑫

本次实验我们把大部分时间花在了搭建 SIP 服务器上，因为之前分别尝试了 GNU SIP Witch 等软件，但是经过测试都存在各种各样的问题，最后选择了一个比较完善的 Flexisip。我们在实际部署时候仍然发现了一些心得问题，后续有时间将尝试看能否帮助开源社区修复掉我们遇到的问题。

通过本次实验，我对 SIP 协议的通信方式有了较为深入的了解。

田静悦

本次实验过程中，遇到了拨打电话但是听不到声音的情况，经过排查，我们怀疑是协议不统一的原因，我使用了 TCP,对端使用了 UDP,在统一协议之后，通话正常进行。此次实验通过抓包分析，我深入理解了拨打 SIP 电话的过程，看到了过程中使用的各类数据包，明白了包的内部参数及其意义。实验过程中也体会到了小组协作的快乐。通过本次实验对 SIP 协议有了更深入的认识。