如何在 asterisk 收发传真

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前言

本文档将就如何使用 OpenVox 的 E1 卡在 asterisk 上收发传真的配置和调试进行说明：如何使用 t.38 在 asterisk 收发传真。

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1 如何使用 t.38 在 asterisk 收发传真

1.1：T.38 协议概述

T.38 全名是 T.38: Procedures for Real-Time Group 3 Facsimile Communication Over IP Networks，是一种透过 IP 的传真编码协定！该协议通过 IP 网进行实时三类传真的通信规程。它对在 IP 网上进行实时三类传真所采用的通信方式，报文格式，纠错方式以及部分通信流程均作了一定的描述和规定。简单的说，T.38 是专门用于 IP 传真使用而定制的规程！

1.2： T.38 协议的安装

1）查看服务器是否支持 T.38 协议，如下图所示：

```
*CLI> module show like fax
Module                          Description                          Use Count
res_fax.so                      Generic FAX Applications             0
1 modules loaded
```

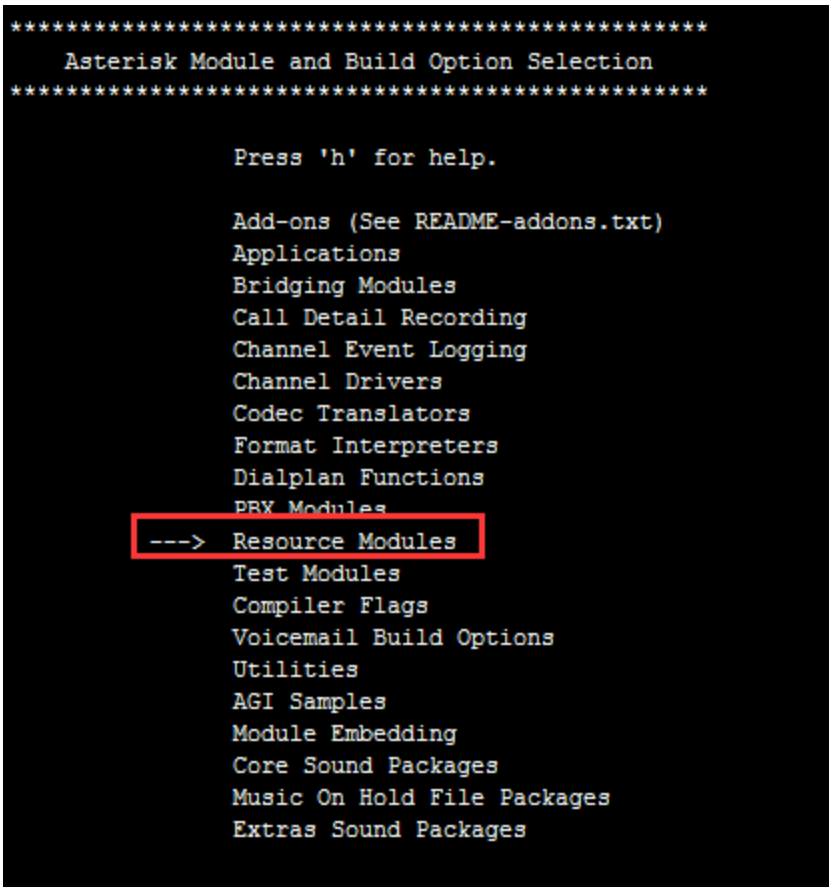
如果出现上图显示结果，则表明不支持 T.38 协议，需要安装 T.38 协议
2）安装 spandsp-0.0.6,安装过程如下:

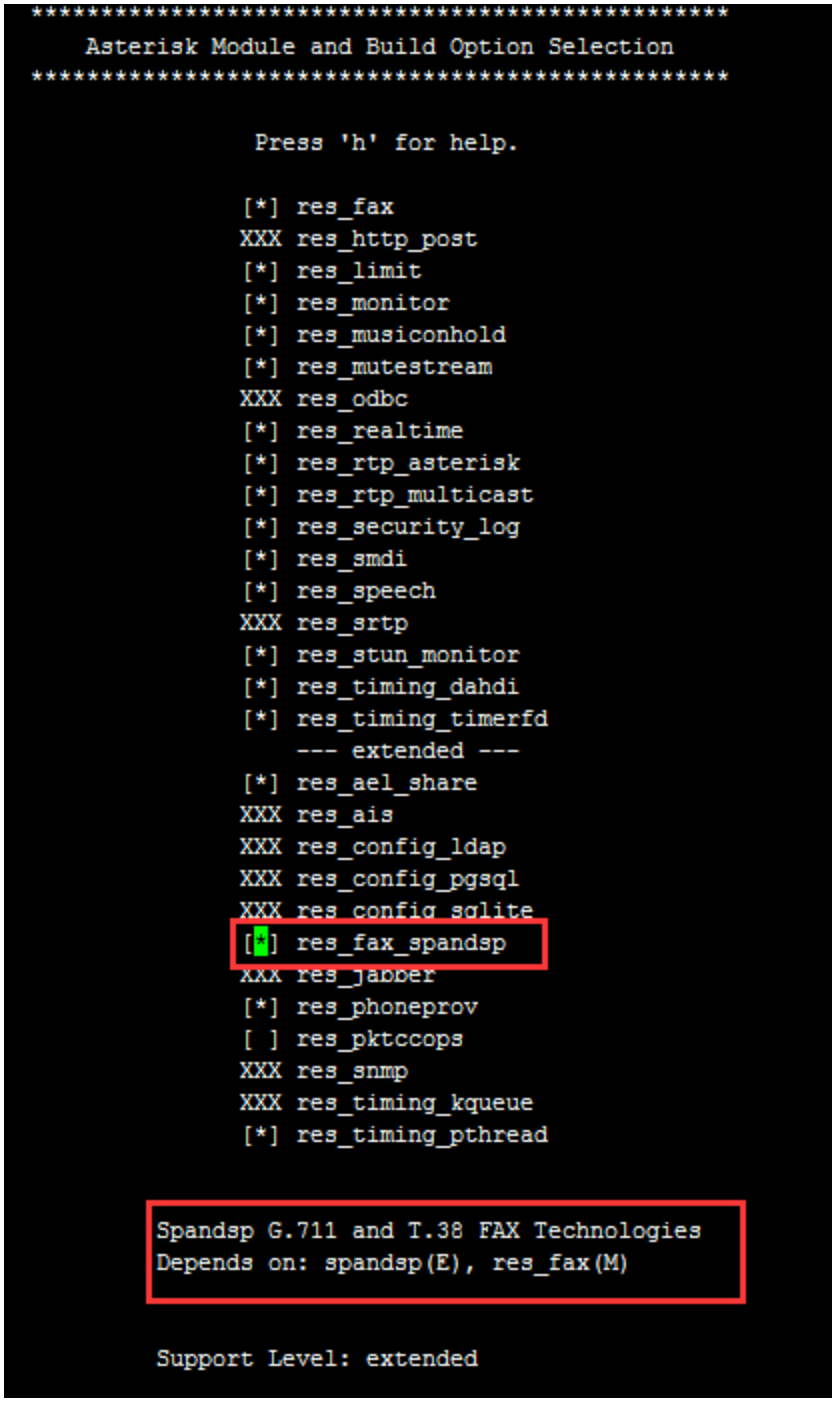
- # wget http://www.soft-switch.org/downloads/spandsp/spandsp-0.0.6pre21.tgz
- # tar -zxvf spandsp-0.0.6pre21.tgz
- # cd spandsp-0.0.6
- # ./configure
- # make
- # make install

3）重新编译安装 asterisk，步骤如下：

- ./configure

- make menuselect

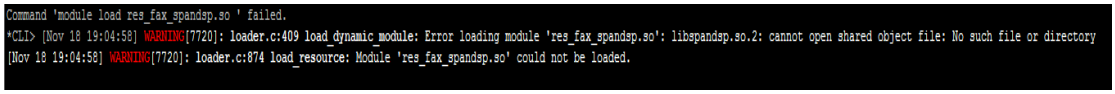



Asterisk Module and Build Option Selection
Press 'h' for help.
[*] res_fax
XXX res_http_post
[*] res_limit
[*] res_monitor
[*] res_musiconhold
[*] res_mutestream
XXX res_odbc
[*] res_realtime
[*] res_rtp_asterisk
[*] res_rtp_multicast
[*] res_security_log
[*] res_smdi
[*] res_speech
XXX res_srtp
[*] res_stun_monitor
[*] res_timing_dahdi
[*] res_timing_timerfd
--- extended ---
[*] res_ael_share
XXX res_ais
XXX res_config_ldap
XXX res_config_pgsql
XXX res_config_sqlite
[*] res_fax_spandsp
XXX res_jabber
[*] res_phoneprov
[ ] res_pktccops
XXX res_snmp
XXX res_timing_kqueue
[*] res_timing_pthread
Spandsp G.711 and T.38 FAX Technologies
Depends on: spandsp(E), res_fax(M)
Support Level: extended

- make
- make install

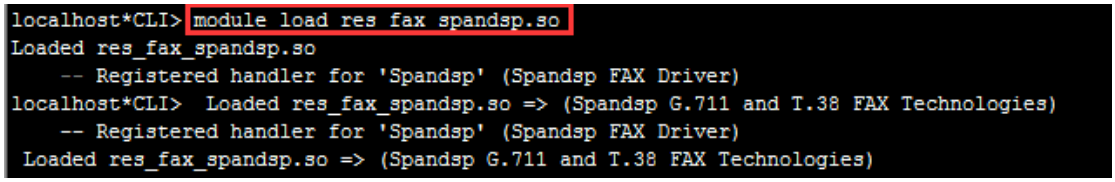
4）安装完成之后检查是否能正常加载 res_fax_spandsp.so 模块

执行命令：module load res_fax_spandsp.so，如出现下图所示警告

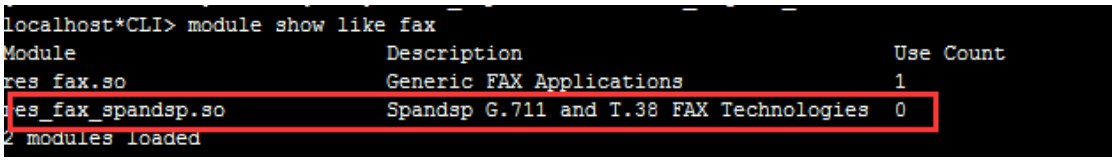


则执行命令：[root@localhost ~]# ln /usr/local/lib/libspandsp.so.2.0.0 /lib/libspandsp.so.2

再次加载模块，如下图所示：



查看是否加载成功：



出现上图显示结果表明加载成功！

1.3：Asterisk T.38 传真测试

1.测试环境

Asterisk version：1.8.20

Server IP：192.168.2.120

2.测试拓扑图 1：

VoIP Plug-In fax(windows A)----→Asterisk Server(192.168.2.120)--→VoIP Plug-In fax(Windows B)

1）Asterisk Server SIP 的配置（sip.conf）

1.3.2.1.1：启动 T.38 支持，在[general] section 下增加如下内容

[general]

directmedia=no

t38pt_udptl = yes,fec,maxdatagram=400

pedantic=no

如下图所示：

```
[general]
directmedia=no
t38pt_udptl = yes,fec,maxdatagram=400
pedantic=no
```

2）配置 fax 分机，如下图所示：

```
[8888]
type=friend
context=from-8888
qualify=yes
qualifyfreq=300
host=dynamic
defaultexpiry=300
defaultuser=8888
callerid="Rick Zhu" <8888>
secret=8888


[9999]
type=friend
context=from-9999
qualify=yes
qualifyfreq=300
host=dynamic
defaultexpiry=300
defaultuser=9999
callerid="Rick Zhu" <9999>
secret=9999
```

注意：该分机的配置方式采用注册方式配置，即 fax 设备注册到 asterisk 服务器上，如需要其他的对接方式，可参考下图进行配置：

```
[t38fax]
type=peer
context=Power-T.38-in
qualify=no
defaultexpiry=900
insecure=port,invite
host=sip.t38fax.com
realm=sip.t38fax.com
defaultuser=YOUR_DID
secret=YOUR_PWORD
```

3）拨号规则的配置(/etc/asterisk/extensions.conf)，如下图所示：

```
[from-9999]
exten => _X.,1,Dial(SIP/8888)
exten => _X.,n,Hangup()
```

以上部分完成了所有的配置过程，下面开始测试过程，尝试从 windows A 发送传真，检查 Windows B 是否能收到传真。

4）简单的调试方法以及步骤

A：检查 fax 分机的对接状态，如下图所示：

命令如下：sip show peers

```
centos58*CLI> sip show peers
Name/username              Host                                  Dyn Forcerport ACL Port     Status
8888/8888                  192.168.2.104                          D   N            5060     OK (7 ms)
9999/9999                  192.168.2.122                          D   N            5060     OK (80 ms)
2 sip peers [Monitored: 2 online, 0 offline Unmonitored: 0 online, 0 offline]
```

注意：如检测到 SIP 对接状态不正确，请检查 fax 设备的设置情况，以及 sip.conf 的配置情况，可通过 sip 日志进行查看确定原因！

B：正常发送接收的完整 SIP 日志可参考该日志进行排错（附录 1）！

3.测试拓扑图 2，如下图所示：

VoIP Plug-In fax(windows A)----→Asterisk Server(192.168.2.120)—>save

local

1）sip 设置可按照如测试 1 进行配置

2）拨号规则配置如下（/etc/asterisk/extensions.conf）

```
[from-9999]
;exten => _X.,1,Dial(SIP/8888)
;exten => _X.,n,Hangup()
exten => _X.,1,Set(FAXFILE=/var/spool/asterisk/fax/test.tif)
;exten => _X.,n,Set(FAXFILENOEXT=/var/spool/asterisk/fax/${CALLERIDNUM})
exten => _X.,n,ReceiveFAX(${FAXFILE})
```

打开存储的目录，可看到保存的 tif 文件，查看结果如下：

```
[root@centos58 fax]# pwd
/var/spool/asterisk/fax
[root@centos58 fax]# ls
test.tif
[root@centos58 fax]#
```

4.测试拓扑图 3，如下图所示：

VoIP Plug-In fax(windows A)----→Asterisk Server(192.168.2.120)—>Send to Email

可参考如下代码，注意：AGI 脚本的存放目录为：/var/lib/asterisk/agi-bin

拨号规则如下：

/etc/asterisk/extensions.conf

```
[macro-faxreceive]
  exten => s,1,Verbose(3,Macro: faxreceive)
    same => n,ReceiveFAX(${FAXFILE}.tiff)

[fax]
  exten => s,1,NoOp("Enter Fax")
    same => n,Macro(faxreceive)

  exten => h,1,NoOp("Fax Hang Up")
    same => Set(DSTEMAIL=test@abc.com)
    same => n,ExecIf($[ "${FAXSTATUS}" == "SUCCESS" ]?AGI(ast-sendmail.sh,${CALLERID(num)},${CALLEDFAX},${DSTEMAIL},${FAXFILE}))


[from-9999]
exten => _1234XXXX,1,NoOp("Incoming fax")
  same => n,Set(CALLEDFAX=${EXTEN})
  same => n,Set(FAXTIME=${STRFTIME(${EPOCH},,%C%y%m%d%H%M)})
  same => n,Set(FAXPATH=/var/spool/asterisk/fax/${CALLEDFAX}/)
  same => n,Set(FAXFILE=${FAXPATH}${CALLEDFAX}-${CALLERID(num)}-${UNIQUEID}-${FAXTIME})
  same => n,Answer()
  same => n,Wait(5)
  same => n,Goto(fax,s,1)
  same => n,Hangup()
```

AGI 脚本如下：

```
#!/bin/bash
echo Received parameters $1 $2 $3 $4
DATETIME=`date +"%A %d %b %Y %H:%M"`
TXTPATH=/var/spool/asterisk/fax/txt/
TXTFILE=$1_$2_$DATETIME.txt

if [ -a $4.tiff ]; then

PAGES=$(tiffinfo $4.tiff |grep "Page Number"|grep -c "P")
DT=$(tiffinfo $4.tiff | grep "Date"|awk '{print $2 " " $3;exit;}')
        echo Dear Sir/Madam, >$TXTPATH$TXTFILE
        echo >>$TXTPATH$TXTFILE
        echo You have just received a new fax document. Details as follow:>>$TXTPATH$TXTFILE
        echo >>$TXTPATH$TXTFILE
        echo "From : "$1>>$TXTPATH$TXTFILE
        echo "To   : "$2>>$TXTPATH$TXTFILE
        echo "Start : "$DT>>$TXTPATH$TXTFILE
        echo "End : "$DATETIME>>$TXTPATH$TXTFILE
        echo "Pages : "$PAGES>>$TXTPATH$TXTFILE
        echo >>$TXTPATH$TXTFILE
        echo >>$TXTPATH$TXTFILE
        /usr/bin/mutt -s "New Fax Received" $3 -a $4.tiff < $TXTPATH$TXTFILE
else
        echo "Not found"

fi
```

附录 1：

<--- SIP read from UDP:192.168.2.122:5060 --->
INVITE sip:12345@192.168.2.120 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103A
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:9999@192.168.2.122:5060>
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Allow: INVITE, ACK, BYE, CANCEL, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: 166

v=0
o=IPFax 0 0 IN IP4 192.168.2.122
s=SIP Fax Call
i=IPFax
c=IN IP4 192.168.2.122
t=0 0
m=audio 49156 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv

<------------>
--- (12 headers 10 lines) ---
Sending to 192.168.2.122:5060 (NAT)
Using INVITE request as basis request - 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
Found peer '9999' for '9999' from 192.168.2.122:5060

<--- Reliably Transmitting (NAT) to 192.168.2.122:5060 --->
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103A;received=192.168.2.122;rport=5060
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>;tag=as1b2ffbb3
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 1 INVITE
Server: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="61718f5c"
Content-Length: 0

<------------>
Scheduling destruction of SIP dialog '21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122' in 6400 ms (Method: INVITE)

<--- SIP read from UDP:192.168.2.122:5060 --->
ACK sip:12345@192.168.2.120 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103A;received=192.168.2.122;rport=5060
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>;tag=as1b2ffbb3
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 1 ACK
Max-Forwards: 70
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Content-Length: 0

<------------>
--- (9 headers 0 lines) ---

<--- SIP read from UDP:192.168.2.122:5060 --->

```
INVITE sip:12345@192.168.2.120 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103B
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 2 INVITE
Max-Forwards: 70
Contact: <sip:9999@192.168.2.122:5060>
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Authorization: Digest
        username="9999",realm="asterisk",nonce="61718f5c",opaque="",uri="sip:12345@
        192.168.2.120",response="0adbec0fcd891ec00f71ebd3a573d78d"
Authorization: Digest
        username="9999",realm="asterisk",nonce="61718f5c",opaque="",uri="sip:12345@
        192.168.2.120",response="0adbec0fcd891ec00f71ebd3a573d78d"
Allow: INVITE, ACK, BYE, CANCEL, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: 166

v=0
o=IPFax 0 0 IN IP4 192.168.2.122
s=SIP Fax Call
i=IPFax
c=IN IP4 192.168.2.122
t=0 0
m=audio 49156 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv

<------------>
--- (14 headers 10 lines) ---
Sending to 192.168.2.122:5060 (NAT)
Using INVITE request as basis request - 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
```

```
Found peer '9999' for '9999' from 192.168.2.122:5060
   == Using SIP RTP CoS mark 5
Found RTP audio format 0
Found audio description format PCMU for ID 0
Capabilities: us - 0x80000008000e (gsm|ulaw|alaw|h263|testlaw), peer - audio=0x4
                (ulaw)/video=0x0 (nothing)/text=0x0 (nothing), combined - 0x4 (ulaw)
Non-codec capabilities (dtmf): us - 0x1 (telephone-event|), peer - 0x0 (nothing), combined - 0x0
                (nothing)
Peer audio RTP is at port 192.168.2.122:49156
Looking for 12345 in from-9999 (domain 192.168.2.120)
list_route: hop: <sip:9999@192.168.2.122:5060>

<--- Transmitting (NAT) to 192.168.2.122:5060 --->
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103B;received=192.168.2.122;rport=5060
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 2 INVITE
Server: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:12345@192.168.2.120:5060>
Content-Length: 0


<------------>
      -- Executing [12345@from-9999:1] Dial("SIP/9999-00000000", "SIP/8888") in new stack
   == Using SIP RTP CoS mark 5
Audio is at 17234
Adding codec 0x4 (ulaw) to SDP
Adding codec 0x2 (gsm) to SDP
Adding codec 0x8 (alaw) to SDP
Adding codec 0x800000000000 (testlaw) to SDP
```

```
Adding non-codec 0x1 (telephone-event) to SDP
Reliably Transmitting (NAT) to 192.168.2.104:5060:
INVITE sip:8888@192.168.2.104:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK44c72049;rport
Max-Forwards: 70
From: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
To: <sip:8888@192.168.2.104:5060>
Contact: <sip:9999@192.168.2.120:5060>
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX 1.8.20.0
Date: Sun, 01 Jan 2012 05:10:31 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 286

v=0
o=root 1161657757 1161657757 IN IP4 192.168.2.120
s=Asterisk PBX 1.8.20.0
c=IN IP4 192.168.2.120
t=0 0
m=audio 17234 RTP/AVP 0 3 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:3 GSM/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=sendrecv

---
    -- Called SIP/8888
```

<--- SIP read from UDP:192.168.2.104:5060 --->
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK44c72049;rport
From: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
To: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 102 INVITE
Contact: <sip:8888@192.168.2.104:5060>
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Content-Length: 0

<------------>
--- (9 headers 0 lines) ---
list_route: hop: <sip:8888@192.168.2.104:5060>
    -- SIP/8888-00000001 is ringing

<--- Transmitting (NAT) to 192.168.2.122:5060 --->
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103B;received=192.168.2.122;rport=5060
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>;tag=as18b7c96f
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 2 INVITE
Server: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:12345@192.168.2.120:5060>
Content-Length: 0

<------------>

<--- SIP read from UDP:192.168.2.104:5060 --->

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK44c72049;rport
From: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
To: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 102 INVITE
Contact: <sip:8888@192.168.2.104:5060>
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Allow: INVITE, ACK, BYE, CANCEL, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: 166

v=0
o=IPFax 0 0 IN IP4 192.168.2.104
s=SIP Fax Call
i=IPFax
c=IN IP4 192.168.2.104
t=0 0
m=audio 49158 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv

<---------->
--- (11 headers 10 lines) ---
Found RTP audio format 0
Found audio description format PCMU for ID 0
Capabilities: us - 0x80000008000e (gsm|ulaw|alaw|h263|testlaw), peer - audio=0x4 (ulaw)/video=0x0 (nothing)/text=0x0 (nothing), combined - 0x4 (ulaw)
Non-codec capabilities (dtmf): us - 0x1 (telephone-event|), peer - 0x0 (nothing), combined - 0x0 (nothing)
Peer audio RTP is at port 192.168.2.104:49158
list_route: hop: <sip:8888@192.168.2.104:5060>
set_destination: Parsing <sip:8888@192.168.2.104:5060> for address/port to send to

set_destination: set destination to 192.168.2.104:5060
Transmitting (NAT) to 192.168.2.104:5060:
ACK sip:8888@192.168.2.104:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK288cfa0c;rport
Max-Forwards: 70
From: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
To: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
Contact: <sip:9999@192.168.2.120:5060>
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX 1.8.20.0
Content-Length: 0

-- SIP/8888-00000001 answered SIP/9999-00000000
Audio is at 19842
Adding codec 0x4 (ulaw) to SDP

<--- Reliably Transmitting (NAT) to 192.168.2.122:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103B;received=192.168.2.122;rport=5060
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>;tag=as18b7c96f
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 2 INVITE
Server: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:12345@192.168.2.120:5060>
Content-Type: application/sdp
Content-Length: 181

v=0

o=root 658791135 658791135 IN IP4 192.168.2.120
s=Asterisk PBX 1.8.20.0
c=IN IP4 192.168.2.120
t=0 0
m=audio 19842 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv

<------------>
-- Locally bridging SIP/9999-00000000 and SIP/8888-00000001

<--- SIP read from UDP:192.168.2.122:5060 --->
ACK sip:12345@192.168.2.120:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.122:5060;branch=z9hG4bK103C
From: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
To: <sip:12345@192.168.2.120>;tag=as18b7c96f
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 2 ACK
Max-Forwards: 70
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Authorization: Digest
username="9999",realm="asterisk",nonce="61718f5c",opaque="",uri="sip:12345@
192.168.2.120",response="0adbec0fcd891ec00f71ebd3a573d78d"
Content-Length: 0

<------------>
--- (10 headers 0 lines) ---

<--- SIP read from UDP:192.168.2.104:5060 --->
INVITE sip:9999@192.168.2.120:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.104:5060;branch=z9hG4bK1012
From: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011

To: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 103 INVITE
Max-Forwards: 70
Contact: <sip:8888@192.168.2.104:5060>
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Supported: timer,replaces,billing,presence,*
Allow: INVITE, ACK, BYE, CANCEL, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: 359

v=0
o=IPFax 0 1 IN IP4 192.168.2.104
s=SIP Fax Call
i=IPFax
c=IN IP4 192.168.2.104
t=0 0
m=image 49154 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:72
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxUdpEC:t38UDPRedundancy

<---------->
--- (13 headers 16 lines) ---
Sending to 192.168.2.104:5060 (NAT)
  == Using UDPTL CoS mark 5
Got T.38 offer in SDP in dialog 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
Capabilities: us - 0x80000008000e (gsm|ulaw|alaw|h263|testlaw), peer - audio=0x0

(nothing)/video=0x0 (nothing)/text=0x0 (nothing), combined - 0x0 (nothing)
Non-codec capabilities (dtmf): us - 0x1 (telephone-event|), peer - 0x0 (nothing), combined - 0x0 (nothing)
Got T.38 Re-invite without audio. Keeping RTP active during T.38 session.

<--- Transmitting (NAT) to 192.168.2.104:5060 --->
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.2.104:5060;branch=z9hG4bK1012;received=192.168.2.104;rport=5060
From: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
To: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 103 INVITE
Server: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:9999@192.168.2.120:5060>
Content-Length: 0

<------------>
== Using UDPTL CoS mark 5
set_destination: Parsing <sip:9999@192.168.2.122:5060> for address/port to send to
set_destination: set destination to 192.168.2.122:5060
Reliably Transmitting (NAT) to 192.168.2.122:5060:
INVITE sip:9999@192.168.2.122:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK28f3d8ca;rport
Max-Forwards: 70
From: <sip:12345@192.168.2.120>;tag=as18b7c96f
To: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
Contact: <sip:12345@192.168.2.120:5060>
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 102 INVITE
User-Agent: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH

Supported: replaces, timer
X-asterisk-Info: SIP re-invite (External RTP bridge)
Content-Type: application/sdp
Content-Length: 265

v=0
o=root 658791135 658791136 IN IP4 192.168.2.120
s=Asterisk PBX 1.8.20.0
c=IN IP4 192.168.2.120
t=0 0
m=image 4737 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxDatagram:204
a=T38FaxUdpEC:t38UDPFEC

<--- SIP read from UDP:192.168.2.122:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK28f3d8ca;rport
From: <sip:12345@192.168.2.120>;tag=as18b7c96f
To: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 102 INVITE
Contact: <sip:9999@192.168.2.122:5060>
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Content-Type: application/sdp
Content-Length: 359

v=0
o=IPFax 0 1 IN IP4 192.168.2.122
s=SIP Fax Call

i=IPFax
c=IN IP4 192.168.2.122
t=0 0
m=image 49152 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:72
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxUdpEC:t38UDPRedundancy

<------------>
--- (10 headers 16 lines) ---
Got T.38 offer in SDP in dialog 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
Capabilities: us - 0x80000008000e (gsm|ulaw|alaw|h263|testlaw), peer - audio=0x0 (nothing)/video=0x0 (nothing)/text=0x0 (nothing), combined - 0x0 (nothing)
Non-codec capabilities (dtmf): us - 0x1 (telephone-event|), peer - 0x0 (nothing), combined - 0x0 (nothing)
Got T.38 Re-invite without audio. Keeping RTP active during T.38 session.
set_destination: Parsing <sip:9999@192.168.2.122:5060> for address/port to send to
set_destination: set destination to 192.168.2.122:5060
Transmitting (NAT) to 192.168.2.122:5060:
ACK sip:9999@192.168.2.122:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK1506d721;rport
Max-Forwards: 70
From: <sip:12345@192.168.2.120>;tag=as18b7c96f
To: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
Contact: <sip:12345@192.168.2.120:5060>
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 102 ACK
User-Agent: Asterisk PBX 1.8.20.0

Content-Length: 0

<--- Reliably Transmitting (NAT) to 192.168.2.104:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.104:5060;branch=z9hG4bK1012;received=192.168.2.104;rport=5060
From: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
To: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 103 INVITE
Server: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:9999@192.168.2.120:5060>
Content-Type: application/sdp
Content-Length: 274

v=0
o=root 1161657757 1161657758 IN IP4 192.168.2.120
s=Asterisk PBX 1.8.20.0
c=IN IP4 192.168.2.120
t=0 0
m=image 4118 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxDatagram:397
a=T38FaxUdpEC:t38UDPRedundancy

<---------->

<--- SIP read from UDP:192.168.2.104:5060 --->

```
ACK sip:9999@192.168.2.120:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.104:5060;branch=z9hG4bK1013
From: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
To: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 103 ACK
Max-Forwards: 70
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Content-Length: 0


<------------>
--- (9 headers 0 lines) ---

<--- SIP read from UDP:192.168.2.104:5060 --->
BYE sip:9999@192.168.2.120:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.104:5060;branch=z9hG4bK1013
From: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
To: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 104 BYE
Max-Forwards: 70
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Content-Length: 0


<------------>
--- (9 headers 0 lines) ---
Sending to 192.168.2.104:5060 (NAT)
Scheduling destruction of SIP dialog '4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060' in
                    6400 ms (Method: BYE)

<--- Transmitting (NAT) to 192.168.2.104:5060 --->
SIP/2.0 200 OK
```

Via: SIP/2.0/UDP 192.168.2.104:5060;branch=z9hG4bK1013;received=192.168.2.104;rport=5060
From: <sip:8888@192.168.2.104:5060>;tag=IPF_PORT_0002_1011
To: "John Doe" <sip:9999@192.168.2.120>;tag=as3486268c
Call-ID: 4a0ac7fa0857b4c2249f84bd5c4a6cc9@192.168.2.120:5060
CSeq: 104 BYE
Server: Asterisk PBX 1.8.20.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0

<------------>
== Spawn extension (from-9999, 12345, 1) exited non-zero on 'SIP/9999-00000000'
Scheduling destruction of SIP dialog '21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122' in 6400 ms (Method: ACK)
set_destination: Parsing <sip:9999@192.168.2.122:5060> for address/port to send to
set_destination: set destination to 192.168.2.122:5060
Reliably Transmitting (NAT) to 192.168.2.122:5060:
BYE sip:9999@192.168.2.122:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK09510fcd;rport
Max-Forwards: 70
From: <sip:12345@192.168.2.120>;tag=as18b7c96f
To: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 103 BYE
User-Agent: Asterisk PBX 1.8.20.0
Proxy-Authorization: Digest username="9999", realm="asterisk", algorithm=MD5, uri="sip:192.168.2.120", nonce="", response="c2f1c35e72aaf8310a8b1692c8c44099"
X-Asterisk-HangupCause: Normal Clearing
X-Asterisk-HangupCauseCode: 16
Content-Length: 0

<--- SIP read from UDP:192.168.2.122:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.120:5060;branch=z9hG4bK09510fcd;rport
From: <sip:12345@192.168.2.120>;tag=as18b7c96f
To: IPFax <sip:9999@192.168.2.122>;tag=IPF_PORT_0001_1039
Call-ID: 21ab6b53-9ac2-4a30-83d5-15178d8a6e54@192.168.2.122
CSeq: 103 BYE
Contact: <sip:IPFax@192.168.2.122:5060>
User-Agent: Net Satisfaxtion/IP_FAX-9.0.6194.732
Content-Length: 0

<------------>