How to set FAX on asterisk

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Introduction

This article explains how to use the OpenVox E1 card sends and receives configuration and debugging of faxes on asterisk, and explains how to use the t38 sends and receives faxes on asterisk.

Content

1 How to use the t38 sends and receives faxes on asterisk

1.1 T.38 Protocol

The full name of T.38: Procedures for Real-Time Group 3 Facsimile Communication Over IP Networks. It is a protocol through IP fax coding, which for communication regulation of real-time facsimile by IP network, and explains Communication, message format, correction, and part of the communication process of real-time facsimile. In a word, T.38 is a regulation specifically make for IP fax .

1.2 Installation T.38 protocol

1）Check the server to confirm whether it supports T.38 protocol:

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

If the results appears as above, it means that the protocol does not support T.38, T.38 protocol needs to be installed.

2）The installation process of spandsp-0.0.6 as shown below:

- # 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）Recompile and install asterisk，please step by the followings：

- ./configure

- make menuselect

```
*************************************************
    Asterisk Module and Build Option Selection
*************************************************

                Press 'h' for help.

                Add-ons (See README-addons.txt)
                Applications
                Bridging Modules
                Call Detail Recording
                Channel Event Logging
                Channel Drivers
                Codec Translators
                Format Interpreters
                Dialplan Functions
                PBX Modules
          --->  Resource Modules
                Test Modules
                Compiler Flags
                Voicemail Build Options
                Utilities
                AGI Samples
                Module Embedding
                Core Sound Packages
                Music On Hold File Packages
                Extras Sound Packages
```

```
 ************************************************
     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）After the installation is completed, please check whether the res_fax_spandsp.so module can load normally.

Excuting the order：module load res_fax_spandsp.so，if warnings appears as shown in the figure below,

```
Command 'module load res_fax_spandsp.so ' failed.
*CLI> [Nov 18 19:04:58] WARNING[7720]: loader.c:409 load_dynamic_module: Error loading module 'res_fax_spandsp.so': libspandsp.so.2: cannot open shared object file: No such file or directory
[Nov 18 19:04:58] WARNING[7720]: loader.c:874 load_resource: Module 'res_fax_spandsp.so' could not be loaded.
```

Next excuting the order：[root@localhost ~]# ln /usr/local/lib/libspandsp.so.2.0.0 /lib/libspandsp.so.2

Loading module again :

```
localhost*CLI> module load res_fax_spandsp.so
Loaded res_fax_spandsp.so
    -- Registered handler for 'Spandsp' (Spandsp FAX Driver)
localhost*CLI>  Loaded res_fax_spandsp.so => (Spandsp G.711 and T.38 FAX Technologies)
    -- Registered handler for 'Spandsp' (Spandsp FAX Driver)
 Loaded res_fax_spandsp.so => (Spandsp G.711 and T.38 FAX Technologies)
```

To check if it can load successfully：

```
localhost*CLI> module show like fax
Module                          Description                               Use Count
res_fax.so                      Generic FAX Applications                  1
res_fax_spandsp.so              Spandsp G.711 and T.38 FAX Technologies   0
2 modules loaded
```

If the above information appears, it means load successfully.

1.3：Asterisk T.38 fax test

1. Test environment:

   Asterisk version：1.8.20

   Server IP：192.168.2.120

2. Test topology 1：

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

1）Configuration of asterisk Server SIP（sip.conf）

Start T38 support， add the following information to [general] section

[general]

directmedia=no

t38pt_udptl = yes,fec,maxdatagram=400

pedantic=no

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

2）The follow is Configuring fax extension:

```
[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
```

Notice： The method of extension configuration is register mode configuration, that means fax device is registered to the asterisk server, if need other docking mode, refer to the configuration diagram :

```
[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）Configuring of dialing rules (/etc/asterisk/extensions.conf):

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

The above shows all the configuration process is completed, you can start to test try to send a fax from windows A, check Windows B to confirm whether it can receive the fax.

4） Simple debugging methods and steps

A： Check the status of fax extension butt

As follow： 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]
```

Notice： If detecting SIP docked incorrect, please check the setting of fax equipment and sip.conf configuration, you can view sip log to make sure what is the cause.

B: The SIP log be shown on Appedix 1（Appendix 1）

3. Test topology 2:

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

1） Sip settings can be configured according to Test 1

2） dialing rules configuration（/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})
```

Enter the storage directory, you can see the saved tif file, please check the results as follows：

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

4. Test topology 3:

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

Please refer to the code.

notice：Store directory of AGI script is：/var/lib/asterisk/agi-bin

dialing rules：

/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 script：

```bash
#!/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
```

Appendix 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:1234
        5@192.168.2.120",response="0adbec0fcd891ec00f71ebd3a573d78d"
```

```
Authorization: Digest
        username="9999",realm="asterisk",nonce="61718f5c",opaque="",uri="sip:1234
        5@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

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