Q: ???? SIP Trunk ??????????

chan_sip.c: No compatible codecs, not accepting this offer

A??? codec ?????????? SIP Trunk ? codec?????????????? codec????????

1. ?? sip set debug

2. ???????????????? log

```
INVITE sip:07010178814@106.104.139.77:5060 SIP/2.0
Via: SIP/2.0/UDP
202.133.231.17:5060;rport;branch=z9hG4bK+2ba826d2cb9c28f55649158b2e42156c1+sip+1+a9d2c847
From: <sip:0975166435@chiefcall.com.tw>;tag=202.133.231.17+1+98d91835+cbefed37
To: <sip:07010178814@chiefcall.com.tw>
CSeq: 623980780 INVITE
Expires: 180
Content-Length: 175
Call-Info: <sip:202.133.231.17:5060>;method="NOTIFY;Event=telephone-event;Duration=2000"
Supported: resource-priority,siprec, 100rel
Contact: <sip:91a18931544b6920a8fa48f3fd1e7790@202.133.231.17:5060>
Content-Type: application/sdp
Allow-Events: message-summary, refer, dialog, line-seize, presence, call-info, as-feature-event,
calling-name
Call-ID: 0gQAAC8WAAACBAAALxYAAAPNgJ3t7scBqLgDdG2c1DM9/Dzz/Jtbhb8ykdqcuk3o@202.133.231.17
Organization: Metaswitch Networks
Max-Forwards: 69
Accept: application/sdp, application/dtmf-relay

v=0
o=- 51082140073993 51082140073993 IN IP4 202.133.231.17
s=-
c=IN IP4 202.133.231.17
t=0 0
m=audio 40866 RTP/AVP 8 101      <=== ???? codec
a=rtpmap:101 telephone-event/8000
a=ptime:20
<------------->
--- (16 headers 8 lines) ---
Sending to 202.133.231.17:5060 (NAT)
```

Codec ??????

3 : GSM
97 : iLBC
8 : PCMA
0 : PCMU
18 : G729

Q: ?? SIP Trunk ????? Follow Me ????????????????????

A: ?? Asterisk ?? LAN ????? sip_nat.conf ??? IP ????????????????????????

FreePBX > Tools > Asterisk SIP Settings

? Other SIP Setting ??

progressinband = yes

Q: ???? CDR ?? Master.csv ????

A????? logrotate ???????? SHELL

asterisk-cdr-rollover.sh?

reference to undefined name 'syntax' Exception of type 'MindTouch.Deki.Script.Runtime.DekiScriptUndefinedNameException' was thrown. (click for details)

Q: ???? xxx.so ????????

A??? asterisk ???? make menuselect ?????????**??????????**??????? https://wiki.asterisk.org/wiki/displ...Support+States

Q: ?? sendmail

A??: ???????

Edit /etc/aliases file and add a “root: username_to_forward_to” to forward all ‘root’ messages to your personal email address. Put in the full email address if it is not on the asterisk system itself.
Then run

/usr/bin/newaliases

to restart the service.

If emails are not received you must set up masquerading in sendmail. These still may be rejected if the email server requires the source of the email to also resolve to the same DNS that sendmail is masquerading as.
To enable this, add the following lines to the /etc/mail/sendmail.mc file:

```
MASQUERADE_AS(domain.com)dnl
FEATURE(masquerade_envelope)dnl
FEATURE(masquerade_entire_domain)dnl
MASQUERADE_DOMAIN(domain.com)dnl
```

Put a “dnl” in front of the line ”EXPOSED_USER (`root’) dnl”. This enables host masquerading for root as well which is disabled by default.
Update the Sendmail configuration files using the m4 macro processor to generate a new sendmail.cf file by executing the following command:

```
# m4 /etc/mail/sendmail.mc > /etc/mail/sendmail.cf
```

To get the Sendmail macro file, the sendmail-cf package must be installed on the system.
After creating a new /etc/mail/sendmail.cf file, restart Sendmail for the changes to take effect. To do this, use the following command:

```
# service sendmail restart
# nano /etc/asterisk/vm_general.inc
```

change serveremail=vm@asterisk to whom ever you want it to appear voicemail emails are coming from.

NOTE: If you are installing on a LAN or do not have a domain resolving to the IP of the VPS, Sendmail will hang for a couple minutes everytime you reboot. To prevent this your VPS hostname should end with .local or .localhost. So, for example, instead of naming the VPS hostname 'powerpbx' it should be named 'powerpbx.local'. The manual method is to edit your /etc/hosts file. There should be 2 lines.

127.0.0.1 localhost.localdomain localhost

yourIPaddress yourhostname.local yourhostname yourhostname

Q: ???? logrotate

A??? /etc/logrotate.d/asterisk

```
/var/log/asterisk/messages /var/log/asterisk/*log /var/log/asterisk/full {
  missingok
  notifempty
  sharedscripts
  create 0640 asterisk asterisk
  postrotate
  /usr/sbin/asterisk -rx 'logger reload' > /dev/null 2> /dev/null
  endscript
}
```

Q: ?????? 15 sec ?????????????:

[2012-02-29 16:41:42] WARNING[2713] chan_sip.c: Maximum retries exceeded on transmission 3c417d294417b1ff2139128123697520@sip2sip.info for seqno 103 (Critical Response) -- See https://wiki.asterisk.org/wiki/display/AST/SIP+Retransmissions
[2012-02-29 16:41:42] WARNING[2713] chan_sip.c: Hanging up call 3c417d294417b1ff2139128123697520@sip2sip.info - no reply to our critical packet (see https://wiki.asterisk.org/wiki/display/AST/SIP+Retransmissions).

A: ???????? Asterisk ???? NAT ??????????????

- ?? sip.conf ? externip, localnet ??????? firewall ?? 5060(UDP), 10000-20000(UDP) ? port forwarding?
- ?? canreinvite=no
- Asterisk ??????
- ????? firewall?pfsense 2.0.1 ????????????????????????????
    1. Firewall > NAT > Port Forward ?? 5060(UDP) ? 10000-20000(UDP) ? port forward ???
    2. Firewall > NAT > Outbound??? Manual Outbound NAT rule generation (AON) ??????????

- ? Interface = WAN
- ? Source = <Asterisk's IP>/32 ; netmask ? 32 ??? single IP
- ? Source Port = any
- ? Destination = any
- ? Destination Port = any
- ? NAT address = any
- ? NAT Port = any
- ? Static Port = yes ;??

3. NOTE:
   - ? ?? Auto NAT ??? Static NAT ????????? 3 ? rule????????? LAN ???????
   - ? Static NAT Rule for Asterisk ????????????????
4. pfsense ???????Asterisk ???????

- ???????? SIP Provider?? SIP Provider ??????????? CCNet ??????????? sip2sip.info ?????
- ??? pfsense ? Firewall???????? Automatic NAT & Static NAT ???????????? automatic NAT ????