

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/Jtbhb8ykdcuk3o@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
```


- ? 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 ?????