FusionPBX

A full-featured domain based multi-tenant PBX and voice switch for FreeSwitch.

Links

- Website: https://www.fusionpbx.com/
- Forum: https://www.pbxforums.com/
- Documentation: https://docs.fusionpbx.com/en/latest/index.html
- Github: https://github.com/fusionpbx/fusionpbx

Installation

- Install script: https://www.fusionpbx.com/download

Debian 11

```
wget -O - https://raw.githubusercontent.com/fusionpbx/fusionpbx-install.sh/master/debian/pre-install.sh | sh;

cd /usr/src/fusionpbx-install.sh/debian && ./install.sh
```

NAT Setting

Web Admin > Advanced > Variables > IP Addresses

- external_rtp_ip: <server-public-ip>
- external_sip_ip: <server-public-ip>

?? freeswitch

```
systemctl restart freeswitch
```

??

Web Admin > Status > SIP Status

- sofia status profile internal: ext-rtp-ip, ext-sip-ip
- sofia status profile external: ext-rtp-ip, ext-sip-ip

RTP Port

/etc/freeswitch/autoload_configs/switch.conf.xml:

```
<!-- RTP port range -->
<!-- If no definitation the port range would be 16384 - 32768 -->
<param name="rtp-start-port" value="16384"/>
<param name="rtp-end-port" value="17000"/>
```

Gateway to Asterisk

On FreePBX

1. Added a custom context 'from-ext-sip-server' with the module Custom Contexts.
2. FreePBX Admin > Connectivity > Custom Contexts > Add Context
   - Context: from-ext-sip-server
   - Description: Whatever
   - Outbound Routes: <allow-some-route>
3. Add Trunk
   - Trunk Name: fusionpbx
   - PEER Details:

```
host=sip.osslab.tw
type=peer
context=from-ext-sip-server
nat=yes
insecure=port,invite
```

On FusionPBX

Web Admin > Accounts > Gateways > Add

- Gateway: myasterisk
- Proxy: <my-asterisk-sip>
- Register: False
- Profile: external
- Enable: Checked

Web Admin > Dialplan > Outbound Routes > Add

- Gateway: myasterisk
- Dialplan Expression: 9 Digits
- Prefix: <blank>
- Enable: True

??????? Bridge ?? Gateway

Applications > Bridge

- Name: <???>
- Destination: `sofia/Internal/$1@<your-freePBX-IP>:5060`

Advanced > Access Controls > Add

- Name: FreePBX
- Default: deny
- Nodes
  - Type: allow
  - CIDR: `<your-freePBX-IP>/32`
  - Domain: <CIDR ? Domain ???>
  - Description: <??>

Outbound Route ???? Bridge?

Voicemail to Email

Web Admin > Accounts > Extensions > Select extension and Edit

- Voicemail Mail to: <your-email-addr>

Web Admin > Advanced > Default Settings > Email

- address_type: add_address
- method: smtp
- smtp_auth: True
- smtp_from: <sender-from-addr>
- smtp_from_name: <sender-from-name>
- smtp_host: smtp-relay.sendinblue.com
- smtp_hostname: False
- smtp_username: <smtp-user>
- smtp_password: <smtp-pass>
- smtp_port: 587
- smtp_secure: tls

- smtp_validate_certificate: True

Send Test Email

Web Admin > Status > Email Logs > TEST

Bug Fixed:

> [ERR] switch_cpp.cpp:1465 [database] can not bind parameter: undefined parameter: email_from

Edit /usr/share/freeswitch/scripts/resources/functions/send_mail.lua

```
if (email_from == nil or email_from == "") then
        email_from = settings:get('email', 'smtp_from', 'text');
        from_name = settings:get('email', 'smtp_from_name', 'text');
end
-- added by Alang
-- fixed: [ERR] switch_cpp.cpp:1465 [database] can not bind parameter: undefined parameter: email_from
email_from = 'noreply@your.domain';

if (email_from == nil or email_from == "") then
        email_from = address;
elseif (from_name ~= nil and from_name ~= "") then
        email_from = from_name .. "<" .. email_from .. ">";
end
```

Voicemail Transcription

IBM Watson API

IBM Cloud > Watson > STT

- API Key: `SQCKJOwC_4VoRozrhw-zm2vYFcxgztFlb2LskGr`
- API URL: `https://api.au-syd.speech-to-text.watson.cloud.ibm.com/instances/{GUID}`

FusionPBX > Advanced > Default Settings > ??????

Category	Subcategory	Type	Value	Enabled
voicemail	transcribe_provider	text	watson	True
voicemail	watson_key	text	{your watson key}	True
voicemail	watson_url	text	{watson url}	True
voicemail	transcribe_language	text	en-US	True
voicemail	transcribe_enabled	boolean	true	True
voicemail	json_enabled	boolean	true	True

- watson_url = `https://api.au-syd.speech-to-text.watson.cloud.ibm.com/instances/{GUID}/v1/recognize?model=en-US_NarrowbandModel&smart_formatting=true`
- ???????https://cloud.ibm.com/docs/speech-to-text?topic=speech-to-text-models

?? "Reload" ?????

FusionPBX > Status > SIP Status

?? "Flush Cache"?"Reload XML" ? "Rescan"?

?? STT API

```
# Audio file: audio-file.flac
curl -X POST -u "apikey:{API_KET}" \
--header "Content-Type: audio/flac" \
--data-binary @audio-file.flac \
"https://api.au-syd.speech-to-text.watson.cloud.ibm.com/instances/{GUID}/v1/recognize"
```

Auto Provisioning

Provision (Linksys PAP2T)

Web Admin > Advanced > Default Settings > Provision

- enabled: `True`, Enabled: True
- admin_password: <????,?????????>, Enabled: True
- http_auth_username: <??>, Enable: False
  NOTE: PAP2T ??? http ??? Auto Provisioning?
- ntp_server_primary: `tw.pool.ntp.org`, Enable: True

- spa_dial_plan: `(*xx|*0xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|*xxxx|xxxxxxxxxxxx.)`, Enabled: True
- spa_time_zone: `GMT+08:00`, Enabled: True

Device

Web Admin > Accounts > Devices > ADD

- MAC Address: <PAP2T-MAC-Addr>
- Domain: <???????>
- Enabled: Checked
- ???????????

Extension

Web Admin > Accounts > Extensions > Add

- Extension: <?????>
- Device Provisioning:
  - Line: 1
  - MAC Address: <?? PAP2T ? MAC>
  - Template: cisco/pap2t
  - Domain: <???????>
  - Enabled: Checked

?? Provision Configuration

????? `http://<fusionpbx-ip-addr>/app/provision/?mac=<pap2t-mac-addr>`

> ????? XML ????????????????????

Linksys PAP2T

PAP2T Web Admin > Provisioning

- Provision Enable: `yes`
- Profile Rule: `http://<fusionpbx-ip-addr>/app/provision/?mac=$MA`

> TIP: ??????????????? NAT ???????? SIP ????????? PAP2T ? Local SIP port ??? port?

?? PAP2T ? Local Port (?? FusionPBX)

Web Admin > Accounts > Devices > Select: <PAP2T-Mac-Addr> > Lines > Line 1

- Port: 1001 (??? 5060)

?? PAP2T ??????????

Cisco IP Phone 8800/7800 Series

- Cisco 8841 User Manual
- Cisco IP Phone 8800 Series :Deployment and Provisioning
- Cisco IP Phone 8800 Series : Provisioning Examples
- Cisco IP Phone 7800 Series and Cisco IP Conference Phone 7832 Multiplatform Phones Provisioning Guide

Let's Encrypt

- Doc - Lent's Encrypt

WebRTC

- Doc - WebRTC
- Github
- SaraPhone

Revision #37
Created 2023-04-09 12:28:51 CST by A-Lang (Admin)
Updated 2024-03-29 15:04:33 CST by A-Lang (Admin)