

FusionPBX

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

Links

- Website: <https://www.fusionpbx.com/>
- Forum: <https://www.pbxfors.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 definition 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

Category	Subcategory	Type	Value	Enabled
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_KEY}" \
--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|xxxxxxxxxxxxx.)`, 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:
- Profile Rule:

“ 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 - Let's Encrypt](#)

WebRTC

- [Doc - WebRTC](#)
- [Github](#)
- [SaraPhone](#)

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