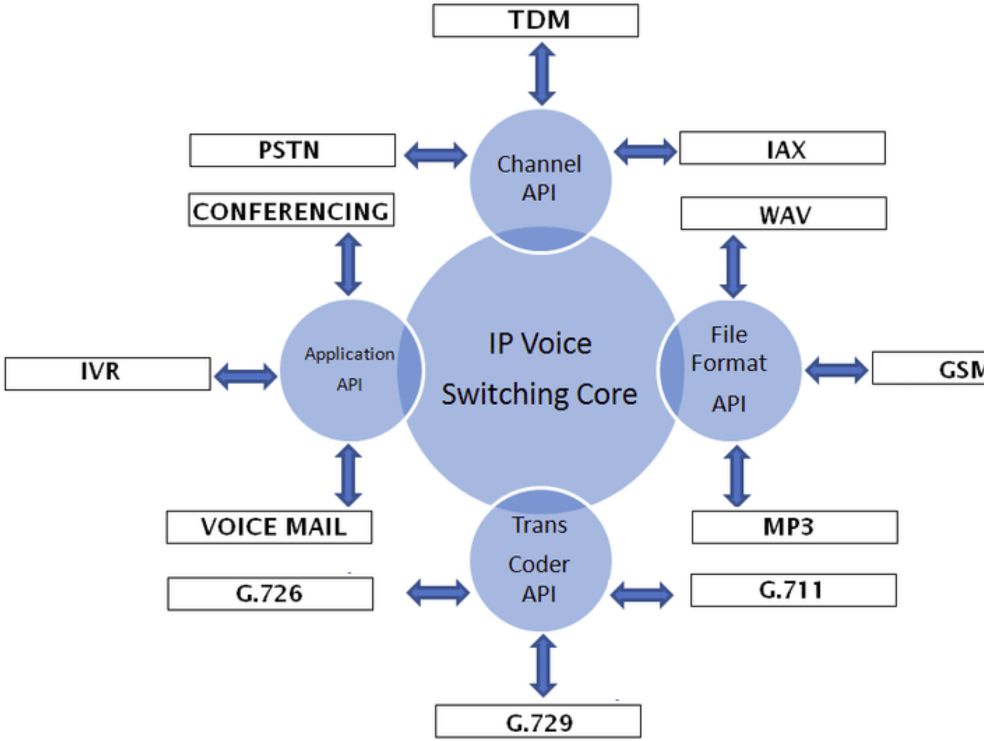
ASTERCONF
- 2020

Почему я использую и Astersisk, и Freeswitch

Asterisk

Удобные «инструменты»

- Офисная PBX
- Сервер регистрации
- AGI
- ARI



Офисная PBX

- Поддержка карт E1
- Множество примеров на форумах
- Хорошая документация (сообщество)
- Интеграция с множеством CRM
- API

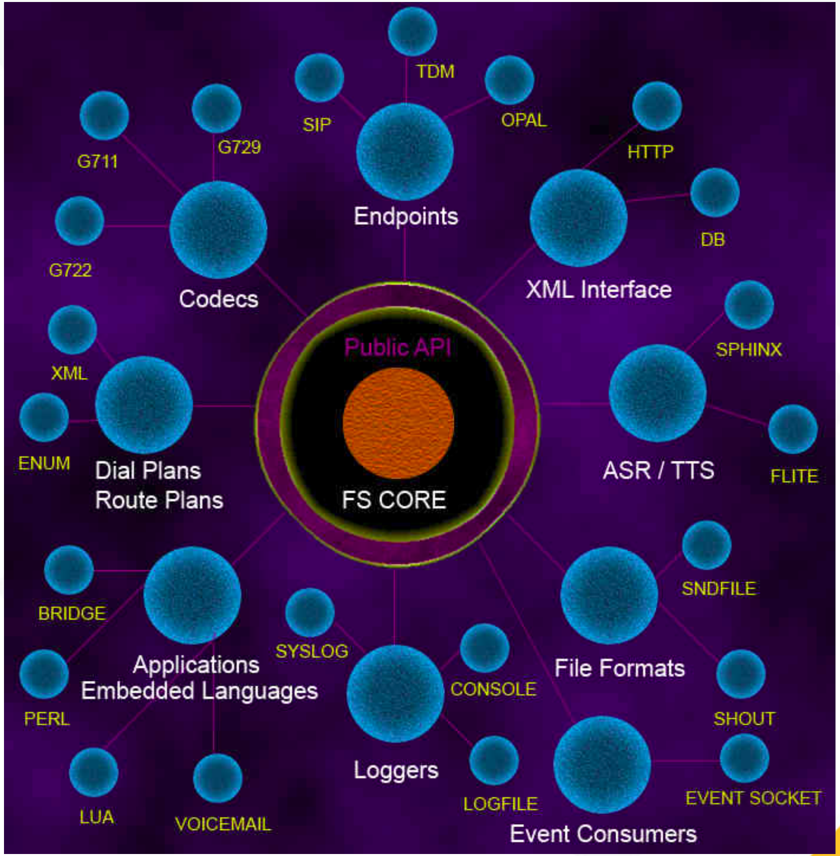
“Enterprise” PBX

- Нестабильность работы при «больших нагрузках»
- Масштабируемость
- «Общие ресурсы»
- Удаление(изменения) любого поля в SIP сообщениях

Freeswitch

Удобные «инструменты»

- Модификация(добавления) headers sip
- Модель состояний(большая нагрузка)
- ESL
- confluence



Freeswitch

Adding Request Headers

You can add arbitrary headers to outbound SIP calls by prefixing the string '**sip_h_**' to any channel variable, for example:

```
<action application="set" data="sip_h_X-Answer=42"/>
<action application="bridge" data="sofia/mydomain.com/1000@example.com"/>
```

Adding Response Headers

There are three types of response header prefixes that can be set:

- Response header
  sip_rh_
- Provisional response header
  sip_ph_
- Bye response header
  sip_bye_h_

Each prefix will exclusively add headers for their given types of requests - there is no "global" response header prefix that will add a header to all response messages.

For example:

```
<action application="set" data="sip_rh_X-Reason=Destination Number Not in Footprint"/>
<action application="set" data="sip_bye_h_X-Accounting=Some Accounting Data"/>
```

Freeswitch

Adding Custom Headers

For instance, you may need **P-Charge-Info** to append to your INVITE header, you may do as follows:

```
<action application="set"><![CDATA[sip_h_P-Charge-Info=<sip:${caller_id_number}@${domain_name}>;npi=0;noa=3]]></action>
```

Then, you would see it in SIP message:

Strip Individual SIP Headers

Sometimes a SIP provider will add extra header information. Most of the time they do that for their own use (tracking calls). But that extra information can cause a lot of problems. For examp (provider1). Since im not in the office the call gets bridged to my cell phone (provider2). Provider1 add's extra information to the sip packet like displayed below:

```
X-voipnow-did: 01234567890
X-voipnow-extension: 987654321
...
```

In some scenario, we bridge this call directly to provider2 the calls get dropped since provider2 doesnt accept the X-voipnow header, so we have to strip off those SIP headers.

To strip them off, use the application UNSET in the dialplan (the inverse of SET):

```
<action application="unset" data="sip_h_X-voipnow-did"/>
<action application="unset" data="sip_h_X-voipnow-extension"/>
...
```

Strip All custom SIP Headers

If you wish to strip all custom headers while keeping only those defined in dialplan:

```
<action application="set" data="sip_copy_custom_headers=false"/>
<action application="set" data="sip_h_X-myCustomHeader=${sip_h_X-myCustomHeader}"/>
...
```

API

- Asterisk
  - AGI
  - AMI
  - ARI
  - CLI

API

- Freswitch
  - CLI
  - API/Event interface
    - mod_event_socket
    - mod_erlang_event
    - mod_xml_rpc
  - Scripting interface
    - mod_perl
    - mod_lua
    - mod_python

ASTERCONF
- 2020

СПАСИБО
ЗА ВНИМАНИЕ!

Замятин Михаил

agic@agic.xyz