SANOG 18

Building Robust IPTSP Based on Open Source Technology

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Session Goal

To provide you a understanding of Building IPTSP, Based on Open source technology includes Asterisk, Kamailio and some other things...

Agenda

- Prelude
- Is Asterisk not enough?
- Why Kamailio?
- Installing Asterisk.
- Installing Kamailio.
- Configuring them to work together.
- Expanding them for bigger game.

Prelude

- Where it all Started.
  - BTRC Issued IPTSP License by the end of 2009 in Bangladesh
  - SANOG 15 at Dhaka in January 2010
    - Jonny Martin and Kabindra Sreshtha took a workshop on IP Telephony and VOIP Deployment.
  - Asterisk
    - The Future of Telephony
    - BUT..........

Is Asterisk Not Enough?

- Asterisk covers everything but does it fit for all purpose?
    - Asterisk - A complete PBX in software
    - Voicemail, conferencing, IVR, queuing, as well as standard calling function
    - Highly extensible - can handle virtually any task imaginable
    - It has some sister apps that has similer feature such as FreePBX, Callweaver, Yate, etc.

Is Asterisk Not Enough? ....

Without a doubt Asterisk is the BEST ip pbx you can have right now.
Its best for small and medium scale office.
Contact Center.
And mostly its nice for developing any telephony application you can think of.

But Think of a IPTSP. number of users may reach millions. even more.
Number of concurrent call. 1000. 5000 even more..
Is asterisk itself the answer?

Is Asterisk Not Enough? ....

In a single server handling millions registration, more than thousand concurrent call is impossible.

Have to deploy an array of server.
Managing the servers and maintain the number plan and routing is not impossible but a nightmare.

So we can think of some thing else that can be a supporting hand for Asterisk.

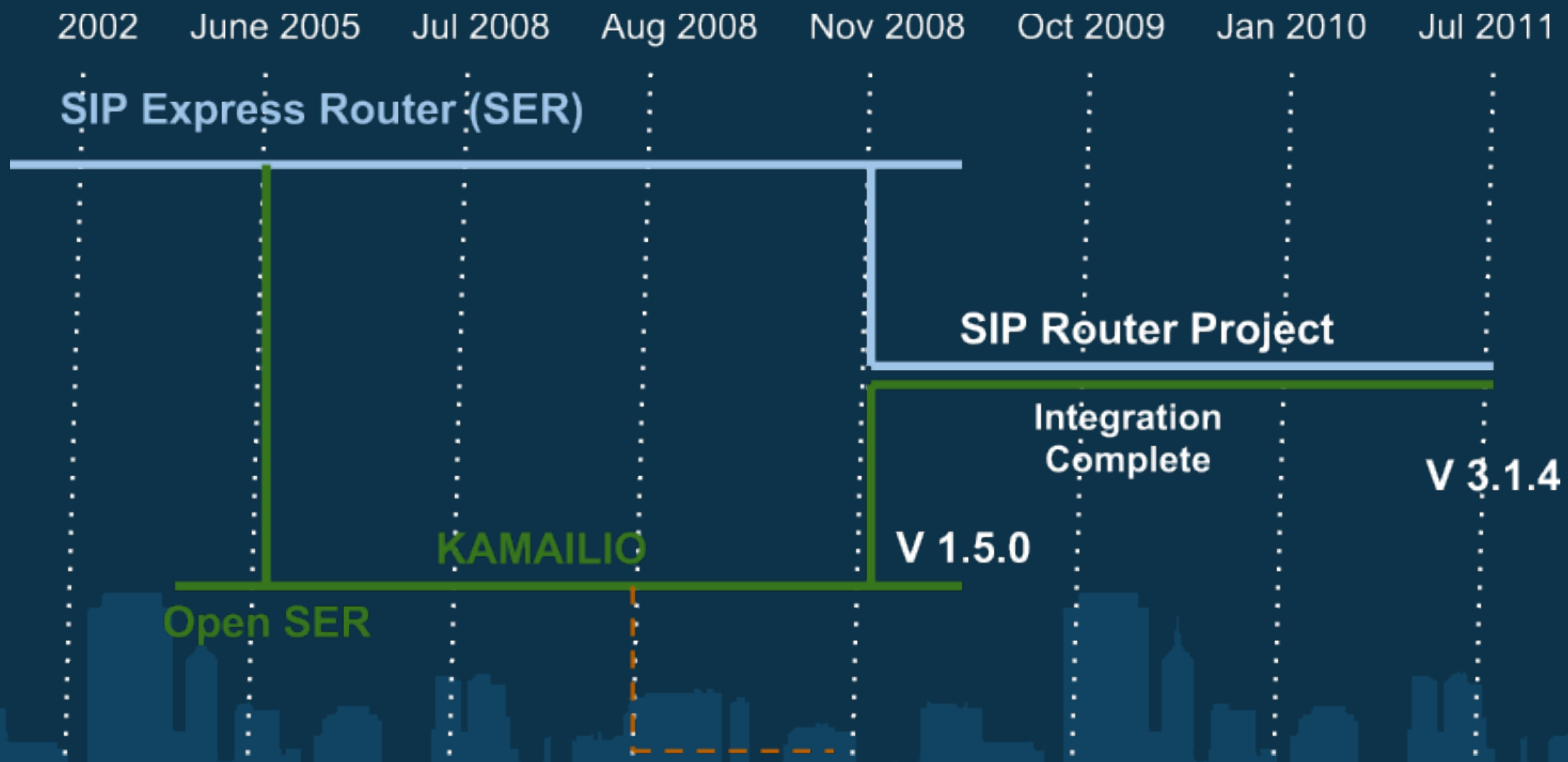
Why Kamailio?

Kamailio is a distribution of SER.
Sip Express Router. commonly known as SER can be the aid for Asterisk for broader perspective. because ..

- Kamailio can handle over 5000 call setups per second.
- On a systems with 4GB memory, Kamailio can serve a population over 300 000 online subscribers
- System can easily scale by adding more Kamailio servers

kamailio is simple first because it doesn't care about everything that done by asterisk. is a simple sip router.

Kamailio development.



Kamailio features

Robust and Performant SIP (RFC3261) Server flavours

- Registrar server
- Location server
- Proxy server
- SIP Application server
- Redirect server

Kamailio features

Flexibility

- small footprint - suitable for embedded devices - the binary file is small size, functionality can be stripped/added via modules
- plug&play module interface - ability to add new extensions, without touching the core, therefore assuring a great stability of core components
- modular architecture - core, internal libraries and module interface to extend the server's functionality
- impressive extension repository - overall 150 modules are included in the Kamailio source tree

Kamailio features.

SIP Routing Capabilities

- stateless and transitional stateful SIP Proxy processing
- serial and parallel forking
- NAT traversal support for SIP and RTP traffic
- load balancing with many distribution algorithms and failover support
- flexible least cost routing
- routing failover
- replication for High Availability (HA)

Kamailio features

Transport Layers

- support for communication via UDP, TCP, TLS and SCTP
- IPv4 and IPv6
- transport layer gatewaying (IPv4 to IPv6, UDP to TLS, a.s. o.)
- SCTP multi-homing and multi-streaming

Kamailio features

Secure Communication

- Digest SIP User authentication
- Authorization via ACL or group membership
- IP and Network authentication
- TLS support for SIP signaling
- transparent handling of SRTP for secure audio
- TLS domain name extension support
- authentication and authorization against database (MySQL, PostgreSQL, UnixODBC, BerkeleyDB, Oracle, text files), RADIUS and DIAMETER

Kamailio features

Accounting

- event based accounting
- configurable accounting data details
- multi-leg call accounting
- storage to database, Radius or Diameter

Kamailio features

Extensibility APIs

- Perl Programming Interface - embed your extensions written in Perl
- Java SIP Servlet Application Interface - write Java SIP Servlets to extent your VoIP services and integrate with web services
- Lua Programming Interface
- Python Programming Interface

for more visit http://www.kamailio.org/w/features/

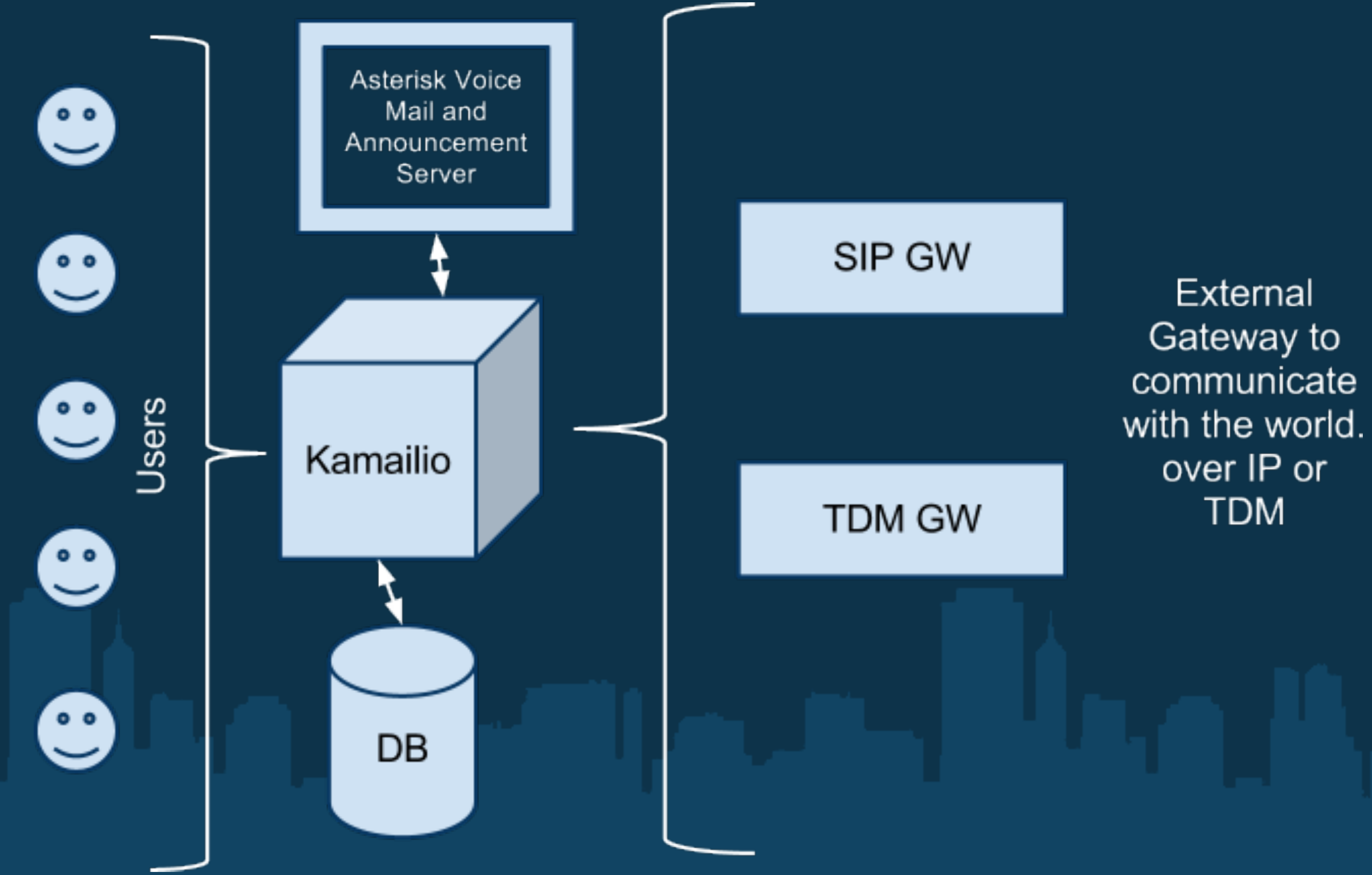
Why Kamailio+Asterisk?

If kamailio can do this all why do we need Asterisk with it?
because we can't do all in our telephony need in kamailio:
Like:

- We can't create IVR, Announcement in Kamailio.
- We can't do Voicemail in Kamailio.
- We can't translate protocol in Kamailio, like SIP->H323 or SIP->IAX2 etc.
- mainly we can't do TDM. like PRI,R2,SS7

So for complete telephony operation we go Kamailio+Asterisk

Kamailio+Asterisk


Asterisk Voice Mail and Announcement Server
Users
Kamailio
DB
SIP GW
TDM GW
External Gateway to communicate with the world. over IP or TDM

Installing Asterisk

On a typical system, you'll want to download three components:

- Asterisk
- DAHDI
- libpri

Additionally for SS7/R2 you may require libss7 or libopenr2.

for compiling you will require GCC and these system library's

- OpenSSL
- ncurses
- newt
- libxml2
- Kernel headers (for building DAHDI drivers)

Installing Asterisk

Download Source:

```
# cd /usr/src/
# wget http://downloads.asterisk.org/pub/telephony/asterisk/releases/asterisk-
1.8.5.0.tar.gz
# wget http://downloads.asterisk.org/pub/telephony/dahdi-linux-
complete/releases/dahdi-linux-complete-2.4.1.2+2.4.1.tar.gz
# wget http://downloads.asterisk.org/pub/telephony/libpri/releases/libpri-
1.4.12.tar.gz
# wget http://downloads.asterisk.org/pub/telephony/libss7/releases/libss7-
1.0.2.tar.gz
# wget http://downloads.asterisk.org/pub/telephony/sounds/asterisk-extra-
sounds-en-wav-current.tar.gz
```

Unter them all.

Building and Installing LibPRI

```
# cd libpri-1.X.Y
# make
# make install
```

Used by many manufacturers of PCI TDM cards

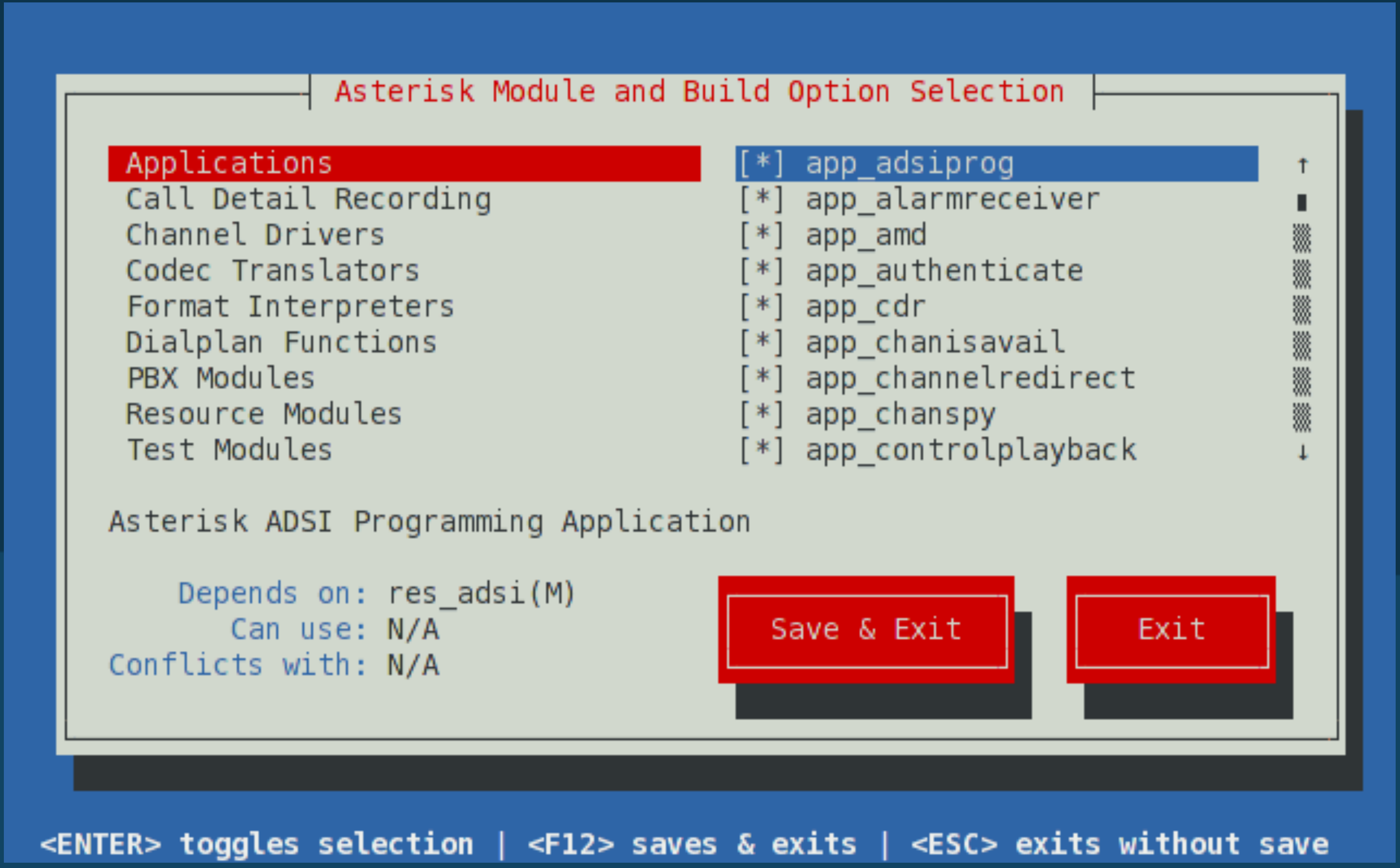
- Safe to compile even if a card is not installed/used

Building and Installing DAHDI

```
# cd dahdi-linux-complete-2.X.Y+2.X.Y
# make
# make install
# make config
```

Building and Installing Asterisk

```
# cd /usr/local/src/asterisk-1.8.X.Y
# ./configure
# make menuselect
```



```
Asterisk Module and Build Option Selection

Applications                    [*] app_adsiprog
Call Detail Recording           [*] app_alarmreceiver
Channel Drivers                 [*] app_amd
Codec Translators               [*] app_authenticate
Format Interpreters             [*] app_cdr
Dialplan Functions              [*] app_chanisavail
PBX Modules                     [*] app_channelredirect
Resource Modules                [*] app_chanspy
Test Modules                    [*] app_controlplayback

Asterisk ADSI Programming Application

   Depends on: res_adsi(M)
      Can use: N/A
Conflicts with: N/A
```

Save & Exit
Exit
<ENTER> toggles selection | <F12> saves & exits | <ESC> exits without save

Building and Installing Asterisk

```
# make
# make install
# make samples
```

if you look for easy way just use your package manager to install asterisk in your Linux system.

Building and Installing Kamailio

Install dependencies

- bison
- pcre-dev
- libpcap-dev

Download latest version of Kamailio

```
# cd /usr/src
# wget http://www.kamailio.
org/pub/kamailio/latest/src/kamailio-3.1.*_src.tar.
gz
# tar -xvzf kamailio-3.1.*_src.tar.gz
```

Building and Installing Kamailio

```
# cd kamailio-3.1.*
# make FLAVOUR=kamailio cfg
```

or if you want to include extra modules which are not installed by default you can type :

```
make FLAVOUR=kamailio include_modules="db_mysql
dialplan" cfg
# make all
# make install
```

Now you have kamailio installed at :

```
/usr/local/etc/kamailio
```

and you have executables at

```
/usr/local/sbin
```

Building and Installing Kamailio

Edit kamctlrc and set the database engine type

```
#vi /usr/local/etc/kamailio/kamctlrc
DBENGINE=MYSQL
```

Create the initial database for kamailio :

```
#kamdbctl create
```

copy the init scripts :

```
#cp /usr/local/src/kamailio-3.1.*/pkg/kamailio/rpm/kamailio.init /etc/init.d/kamailio
#chmod 755 /etc/init.d/kamailio
#cp /usr/local/src/kamailio-3.1.*/pkg/kamailio/rpm/kamailio.default
/etc/default/kamailio
```

`#vi /etc/default/kamailio` and set `RUN_KAMAILIO=yes`

`#vi /etc/init.d/kamailio` and update :

```
KAM=/usr/local/sbin/kamailio
```

/usr/local/etc/kamailio/kamailio.cfg edit this file to fit your requirements.

Configuring Asterisk

To work as a voicemail server for kamailio we have to create

- A SIP Trunk with Kamailio in Asterisk sip.conf.
- Add Voicemail Users in voicemail.conf. and
- Add specific dialplan in extension.conf

Configuring Asterisk

Add SIP Trunk in sip.conf

```
[voicemail-trunk]
type=friend
host=your.ser.domain.com
context=voice-mail-trunk
nat=no
qualify=yes
canreinvite=no
disallow=all
allow=alaw
allow=ulaw
insecure=port,invite
```

Configuring Asterisk

In voicemail.conf add your voice mail users.

```
[default]
extension_number => voicemail_password,user_name,
user_email_address,user_pager_email_address,user_option(s)
```

in extension.conf

```
[voice-mail-trunk]
exten => _XXXX,1,Voicemail(${EXTEN},u)
```

Configuring Kamailio

Before Configuring Kamailio for voicemail we have to create users for Kamailio. to create sip user we have to type ..

```
# kamctl add 1000 mysecret0
# kamctl add 1001 mysecret1
# kamctl add 1002 mysecret2
# kamctl add 1003 mysecret3
```

With this we just added 4 sip user in our kamailio server.

We can now configure our sip client and try dialing each other. you don't have to configure anything for calling your local user.

Basics of Kamailio configuration.

if you see the asterisk configuration folder folder you will find almost 20/30 files in a basic setup. All of them may not important for you.

In Kamailio there is only 2.
kamailio.cfg and
tls.cfg

Only things is important is kamailio.cfg

Basics of Kamailio configuration.

kamailio.cfg has 3 major part.

1. Where we load module

```
   loadmodule "dispatcher.so"
```

2. Where we define module parameter

```
   modparam("dispatcher","list_file","/path/dispatcher.
list")
```

3. Where we route call. the main part.

```
# - processing of any incoming SIP request starts with this
route
    route {
        # per request initial checks
        route(REQINIT);
        # NAT detection
        route(NAT);
```

Basics of Kamailio configuration.

Every Call Starts with

```
        route {
                    }
```

And end in this. there is some sub other function like

```
    failure_route[FAIL_ONE] {
                                              }
```

And

```
route[PSTN] {
                    }
```

this two is very important for us.

Basics of Kamailio configuration.

To use Asterisk voicemail we have to route failed call to asterisk. to do so just add this in you failure_route.

```
failure_route[FAIL_ONE] {
 ... ...... ..
        if (t_check_status("486|408|480")) {
             sethostport("asterisk server host:5060");
             append_branch();
             t_relay();
        }
}
```

if Asterisk and kamailio both installed in the same machine asterisk must be bind in a different port than 5060 and the port should be defined hare.

Basics of Kamailio configuration.

```
if (t_check_status("486|408|480")) {
```

In this clause 486 or 408 and 480 are SIP Response code.
this are

- 486 Busy Here
- 480 Temporarily Unavailable
- 408 Request Timeout

That's explain everything.

Basics of Kamailio configuration.

For Routing Calls to PSTN Gateways you just have to route calls that does't match in your local extension to another server which can handle PSTN or External SIP GW.

All number which doesn't match your local extension will be automatically routed to

```
    route[PSTN] {

        }
```

Basics of Kamailio configuration.

```
route[PSTN] {
# route to PSTN dialed numbers starting with '+' or '00' or
'0'
if(!($rU=~"^(\+|00|0)[1-9][0-9]{3,20}$"))
return;

$ru = "sip:" + $rU + "@your pstn gw ip;

route(RELAY);
exit;
}
```

This way your calls will be routed to PSTN Gateway. which is also a asterisk server with TDM card or an external sip provider.

Expanding for Bigger Game....

Now you have a better sip server that can locally handle lot more call than your only asterisk server can handle.

But the problem is if you have everything in a same machine or have a single Asterisk server as PSTN Gateway how many call it can handle.

And how can you ensure redundancy.

Kamailio has several solution for that. but what I frequently used is dispatcher.

Expanding for Bigger Game....

You already seen how to add dispatcher module and set dispatcher file parameter. here is a simple dispatcher file:

dispatcher.list

```
# gateways
1 sip:yourgatewayip1:5060
1 sip:yourgatewayip2:5060
```

the first column is gateway set id, next is server address and port.
you can add as many set as you wish.

Expanding for Bigger Game....

In-place of `$ru = "sip:" + $rU + "@your pstn gw ip;`
add
```
ds_select_dst("1", "4");
```

in ds_select_dst() method first parameter is the gateway set id. and the second parameter is the algorithm used to select the destination address. here 4 means round-robin (next destination).

Expanding for Bigger Game....

All the algorithm are like this.

- "0" - hash over callid
- "1" - hash over from uri.
- "2" - hash over to uri.
- "3" - hash over request-uri.
- "4" - round-robin (next destination).
- "5" - hash over authorization-username (Proxy-Authorization or "normal" authorization). If no username is found, round robin is used.
- "6" - random (using rand()).
- "7" - hash over the content of PVs string. Note: This works only when the parameter hash_pvar is set.
- "X" - if the algorithm is not implemented, the first entry in set is chosen.

The outcome....

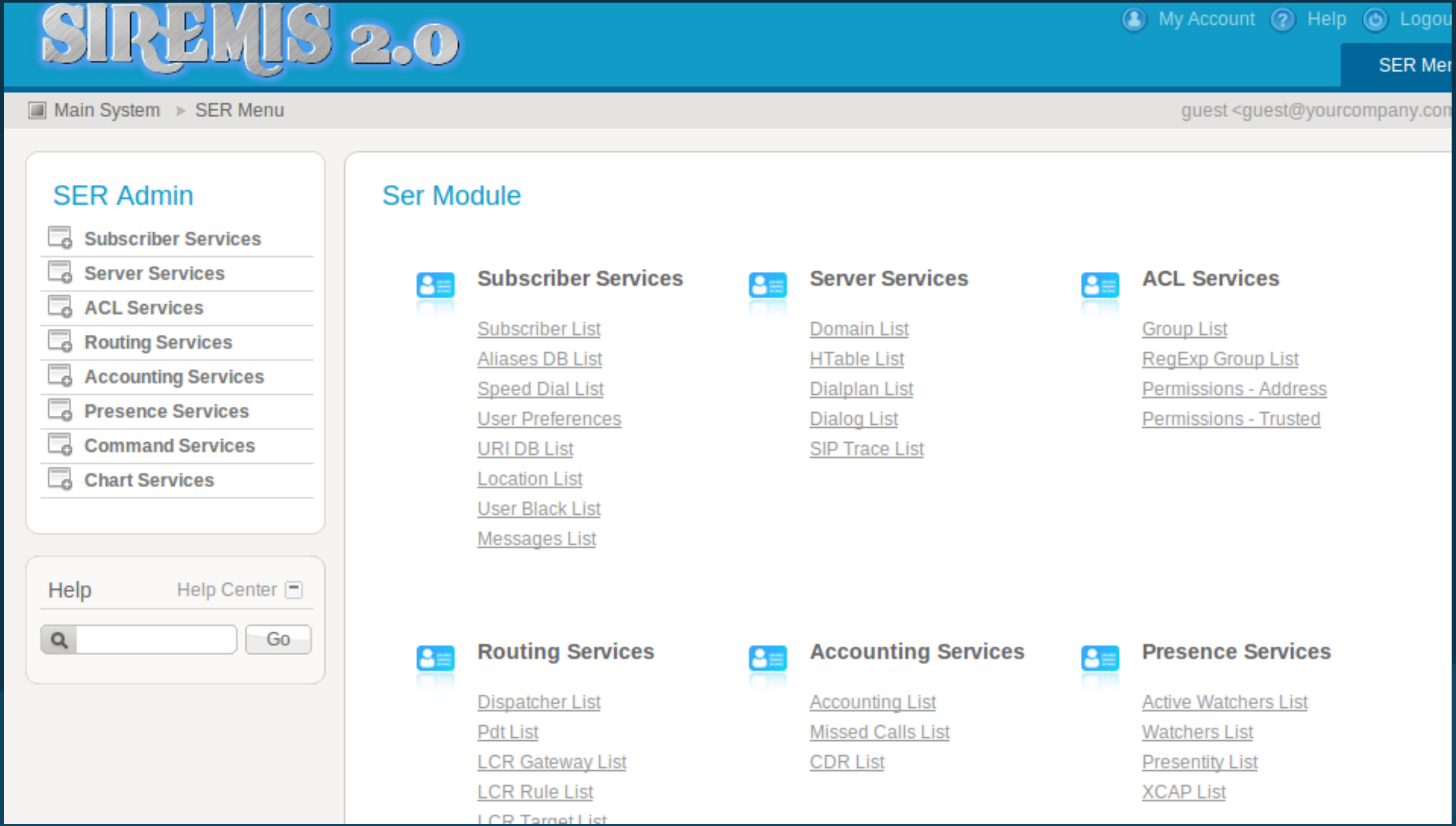
Now we know how to add multiple gateway.
And how we can distribute calls between them with balanced load.

By this way the gateways are not only expanded it also ensure redundancy and high availability.

kamailio server itself can be configured as HA in multiples server and extend its capacity.

Finally

Finally we can use graphical interface to manage kamailio by installing SIREMIS. from http://siremis.asipto.com/



SER Admin

- Subscriber Services
- Server Services
- ACL Services
- Routing Services
- Accounting Services
- Presence Services
- Command Services
- Chart Services

Ser Module

Subscriber Services
- Subscriber List
- Aliases DB List
- Speed Dial List
- User Preferences
- URI DB List
- Location List
- User Black List
- Messages List

Server Services
- Domain List
- HTable List
- Dialplan List
- Dialog List
- SIP Trace List

ACL Services
- Group List
- RegExp Group List
- Permissions - Address
- Permissions - Trusted

Routing Services
- Dispatcher List
- Pdt List
- LCR Gateway List
- LCR Rule List
- LCR Target List

Accounting Services
- Accounting List
- Missed Calls List
- CDR List

Presence Services
- Active Watchers List
- Watchers List
- Presentity List
- XCAP List

Any Questions?

For more reading ....

Asterisk.
https://wiki.asterisk.org/wiki/display/AST/Home

Kamailio
http://www.kamailio.org/w/documentation/

SIREMIS
http://siremis.asipto.com/

Thank You.

keep in touch in http://gplus.to/sujon