La Plateforme Asterisk – From Scratch

On installe une VM avec Debian 12 en mode console (cocher SSH et outils systèmes seulement)

pour afficher l'adresse IP de la machine avant le login :
echo "IPv4 \4" >> /etc/issue
echo "IPv6 \6" >> /etc/issue

on autorise ssh sur le compte root :
echo "PermitRootLogin Yes" > /etc/ssh/sshd_config.d/ssh.conf

On installe les paquets necessaires pour compiler Asterisk :

apt install git curl build-essential libedit-dev git curl wget libnewt-dev libssl-dev libncurses5-dev subversion libsqlite3-dev build-essential libjansson-dev libxml2-dev uuid-dev avahi-daemon libsrtp2-dev

Thierry Rami

On Télécharge la derniere version d'asterisk

```
cd /usr/src
apt install wget
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-21-current.tar.gz
tar xfv asterisk-21-current.tar.gz
rm asterisk-21-current.tar.gz
cd asterisk-21*

./configure --with-srtp
contrib/scripts/get_mp3_source.sh
```

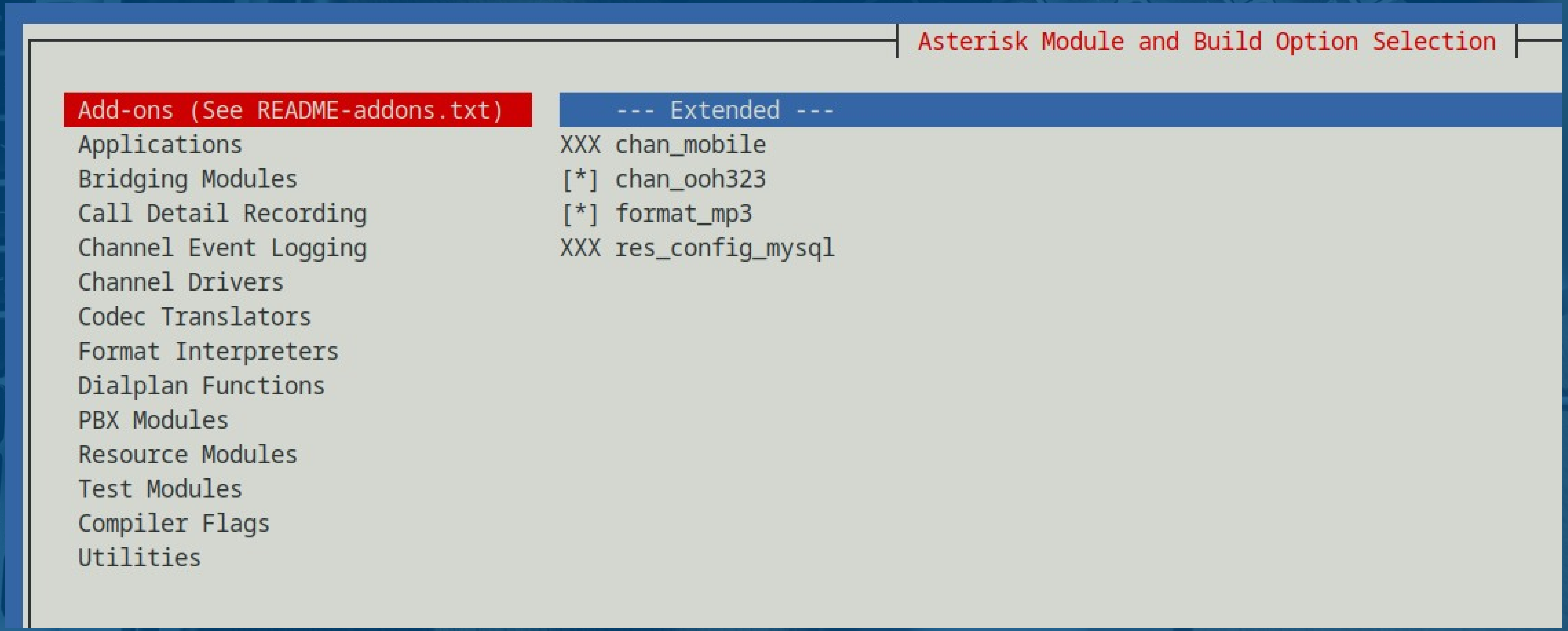
La Plateforme

Asterisk – From Scratch

On configure les options par :

make menuselect.makeopts

make menuselect



```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)     --- Core ---
Applications                        [*] app_agent_pool
Bridging Modules                    [*] app_authenticate
Call Detail Recording               [*] app_bridgeaddchan
Channel Event Logging               [*] app_bridgewait
Channel Drivers                     [*] app_cdr
Codec Translators                   [*] app_celgenuserevent
Format Interpreters                 [*] app_channelredirect
Dialplan Functions                  [*] app_chanspy
PBX Modules                         [*] app_confbridge
Resource Modules                    [*] app_controlplayback
Test Modules                        [*] app_db
Compiler Flags                      [*] app_dial
Utilities                           [*] app_directed_pickup
```

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)     --- Core ---
Applications                        [*] bridge_builtin_features
Bridging Modules                    [*] bridge_builtin_interval_features
Call Detail Recording               [*] bridge_holding
Channel Event Logging               [*] bridge_native_rtp
Channel Drivers                     [*] bridge_simple
Codec Translators                   [*] bridge_softmix
Format Interpreters                     --- Module Options ---
Dialplan Functions                  XXX binaural_rendering_in_bridge_softmix
PBX Modules
Resource Modules
Test Modules
Compiler Flags
Utilities
```

La Plateforme Asterisk – From Scratch

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)      --- Core ---
Applications                         XXX cdr_adaptive_odbc
Bridging Modules                     [*] cdr_custom
Call Detail Recording                [*] cdr_manager
Channel Event Logging                    --- Extended ---
Channel Drivers                      XXX cdr_beanstalkd
Codec Translators                    [*] cdr_csv
Format Interpreters                  XXX cdr_odbc
Dialplan Functions                   XXX cdr_pgsql
PBX Modules                          XXX cdr_radius
Resource Modules                     [*] cdr_sqlite3_custom
Test Modules                         XXX cdr_tds
Compiler Flags
Utilities
```

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)      --- Core ---
Applications                         [*] cel_custom
Bridging Modules                     [*] cel_manager
Call Detail Recording                XXX cel_odbc
Channel Event Logging                    --- Extended ---
Channel Drivers                      XXX cel_beanstalkd
Codec Translators                    XXX cel_pgsql
Format Interpreters                  XXX cel_radius
Dialplan Functions                   [*] cel_sqlite3_custom
PBX Modules                          XXX cel_tds
Resource Modules
Test Modules
Compiler Flags
Utilities
```

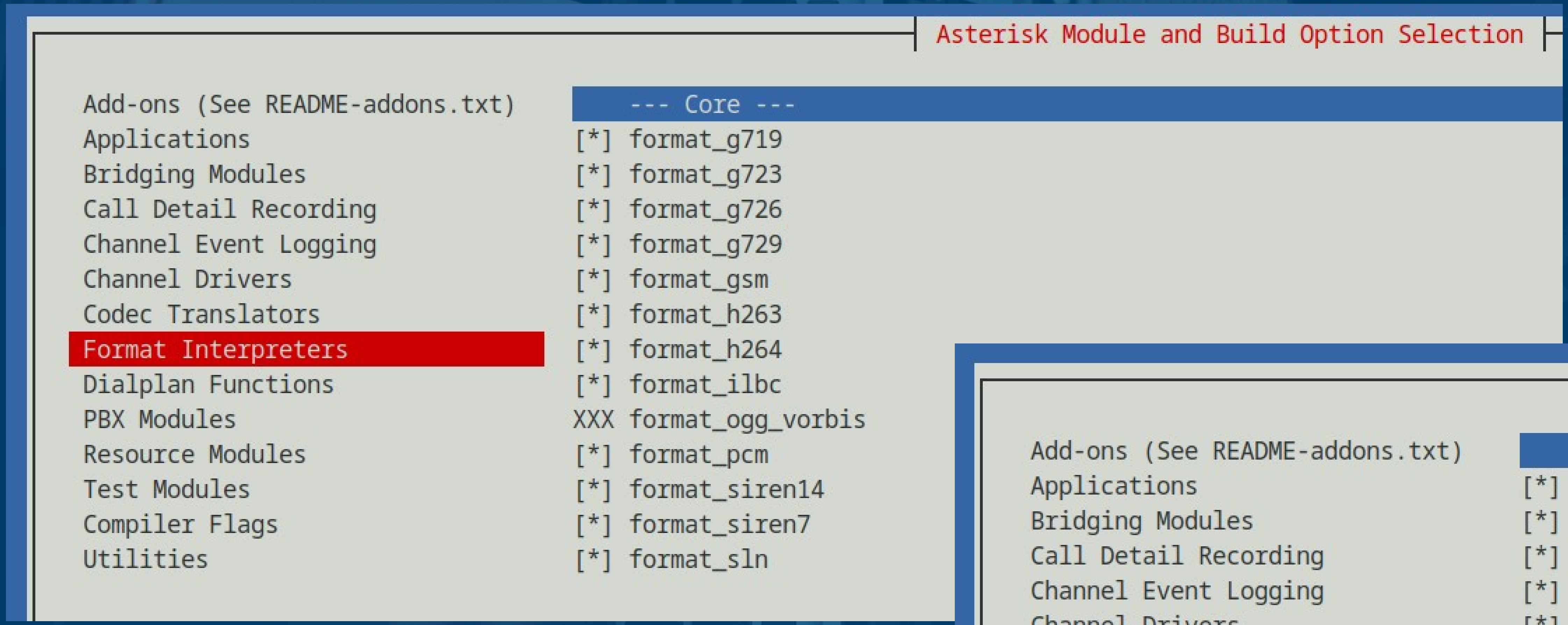
La Plateforme Asterisk – From Scratch

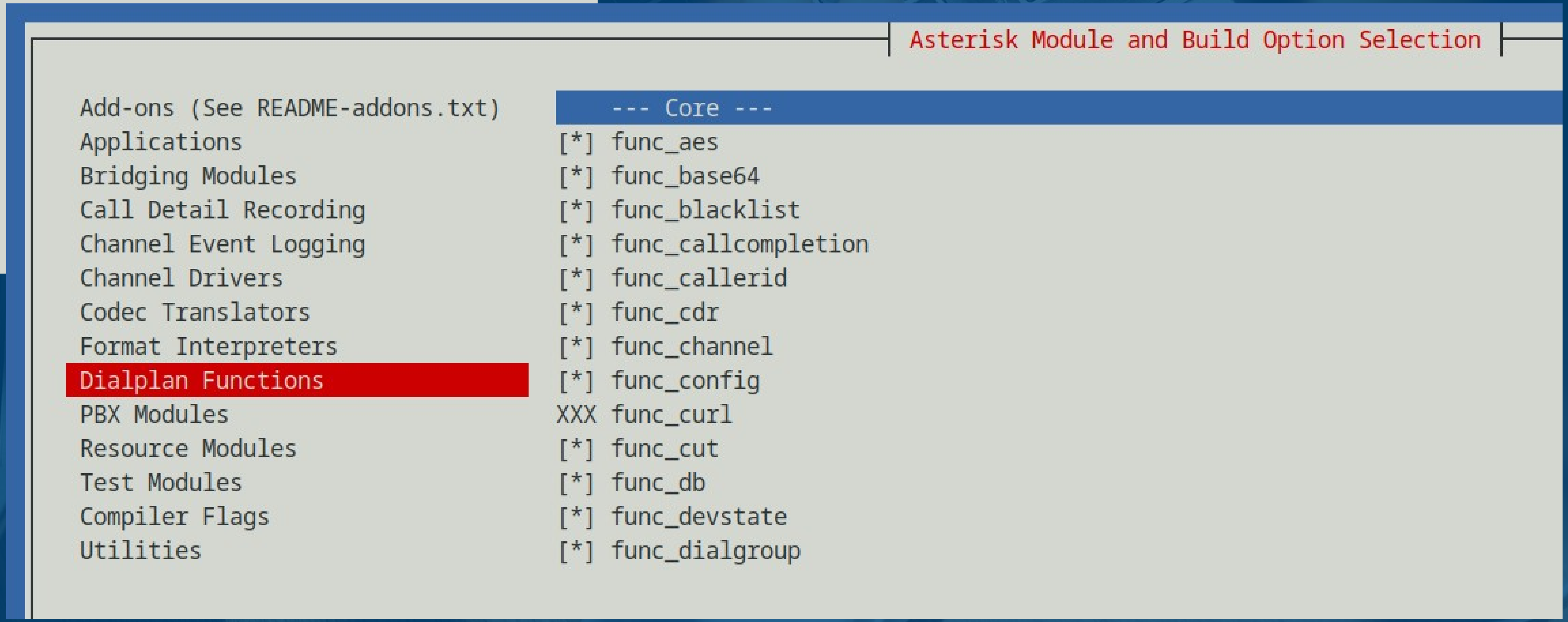
```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)     --- Core ---
Applications                        [*] chan_bridge_media
Bridging Modules                    XXX chan_dahdi
Call Detail Recording               [*] chan_iax2
Channel Event Logging               XXX chan_motif
Channel Drivers                     [*] chan_pjsip
Codec Translators                   [*] chan_rtp
Format Interpreters                     --- Extended ---
Dialplan Functions                  [*] chan_audiosocket
PBX Modules                         XXX chan_console
Resource Modules                    [*] chan_unistim
Test Modules                            --- Deprecated ---
Compiler Flags                      XXX chan_alsa
Utilities                           [ ] chan_mgcp
```

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)     --- Core ---
Applications                        [*] codec_a_mu
Bridging Modules                    [*] codec_adpcm
Call Detail Recording               [*] codec_alaw
Channel Event Logging               XXX codec_codec2
Channel Drivers                     XXX codec_dahdi
Codec Translators                   [*] codec_g722
Format Interpreters                 [*] codec_g726
Dialplan Functions                  [*] codec_gsm
PBX Modules                         [*] codec_ilbc
Resource Modules                    [*] codec_lpc10
Test Modules                        [*] codec_resample
Compiler Flags                      XXX codec_speex
Utilities                           [*] codec_ulaw
```

Thierry Rami

La Plateforme Asterisk – From Scratch





La Plateforme Asterisk – From Scratch

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)      --- Core ---
Applications                         [*] pbx_config
Bridging Modules                     [*] pbx_loopback
Call Detail Recording                [*] pbx_spool
Channel Event Logging                    --- Extended ---
Channel Drivers                      [*] pbx_ael
Codec Translators                    [*] pbx_dundi
Format Interpreters                  XXX pbx_lua
Dialplan Functions                   [*] pbx_realtime
PBX Modules
Resource Modules
Test Modules
Compiler Flags
Utilities
```

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)      --- Core ---
Applications                         [*] res_aeap
Bridging Modules                     [*] res_agi
Call Detail Recording                [*] res_ari
Channel Event Logging                [*] res_ari_applications
Channel Drivers                      [*] res_ari_asterisk
Codec Translators                    [*] res_ari_bridges
Format Interpreters                  [*] res_ari_channels
Dialplan Functions                   [*] res_ari_device_states
PBX Modules                          [*] res_ari_endpoints
Resource Modules                     [*] res_ari_events
Test Modules                         < > res_ari_mailboxes
Compiler Flags                       [*] res_ari_model
Utilities                            [*] res_ari_playbacks
```

La Plateforme

Asterisk – From Scratch

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)        --- Core ---
Applications                           XXX test_abstract_jb
Bridging Modules                       XXX test_acl
Call Detail Recording                  XXX test_aeap
Channel Event Logging                  XXX test_aeap_speech
Channel Drivers                        XXX test_aeap_transaction
Codec Translators                      XXX test_aeap_transport
Format Interpreters                    XXX test_amihooks
Dialplan Functions                     XXX test_aoc
PBX Modules                            XXX test_app
Resource Modules                       XXX test_ari
Test Modules                           XXX test_ari_model
Compiler Flags                         XXX test_ast_format_str_redu
Utilities                              XXX test_astobj2
```

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)        --- Core ---
Applications                           [ ] DONT_OPTIMIZE
Bridging Modules                       < > COMPILE_DOUBLE
Call Detail Recording                  [ ] DEBUG_THREADS
Channel Event Logging                  [ ] DEBUG_FD_LEAKS
Channel Drivers                        XXX BETTER_BACKTRACES
Codec Translators                      [ ] LOTS_OF_SPANS
Format Interpreters                    [ ] MALLOC_DEBUG
Dialplan Functions                     [ ] DEBUG_CHAOS
PBX Modules                            [*] BUILD_NATIVE
Resource Modules                           --- Extended ---
Test Modules                           [ ] REF_DEBUG
Compiler Flags                         [ ] AO2_DEBUG
Utilities                              XXX REBUILD_PARSERS
```

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)    --- Core ---
Applications                       [*] astcanary
Bridging Modules                   [*] astdb2sqlite3
Call Detail Recording              [*] astdb2bdb
Channel Event Logging                  --- Extended ---
Channel Drivers                    [ ] check_expr
Codec Translators                  [ ] check_expr2
Format Interpreters                XXX smsq
Dialplan Functions                 [ ] stereorize
PBX Modules                        [ ] streamplayer
Resource Modules                   XXX conf_bridge_binaural_hri
Test Modules                           --- Deprecated ---
Compiler Flags                     [ ] aelparse
Utilities                          [ ] astman
```

```
Asterisk Module and Build Option Selection

Bridging Modules                   --- Extended ---
Call Detail Recording              [ ] agi-test.agi
Channel Event Logging              [ ] eagi-test
Channel Drivers                    [ ] eagi-sphinx-test
Codec Translators                  [ ] jukebox.agi
Format Interpreters
Dialplan Functions
PBX Modules
Resource Modules
Test Modules
Compiler Flags
Utilities
AGI Samples
Core Sound Packages
```

La Plateforme Asterisk – From Scratch

```
Asterisk Module and Build Option Selection

Bridging Modules              [ ] CORE-SOUNDS-ES-GSM
Call Detail Recording         [ ] CORE-SOUNDS-ES-G729
Channel Event Logging         [ ] CORE-SOUNDS-ES-G722
Channel Drivers               [ ] CORE-SOUNDS-ES-SLN16
Codec Translators             [ ] CORE-SOUNDS-ES-SIREN7
Format Interpreters           [ ] CORE-SOUNDS-ES-SIREN14
Dialplan Functions            [ ] CORE-SOUNDS-FR-WAV
PBX Modules                   [*] CORE-SOUNDS-FR-ULAW
Resource Modules              [*] CORE-SOUNDS-FR-ALAW
Test Modules                  [*] CORE-SOUNDS-FR-GSM
Compiler Flags                [*] CORE-SOUNDS-FR-G729
Utilities                     [*] CORE-SOUNDS-FR-G722
AGI Samples                   [ ] CORE-SOUNDS-FR-SLN16
Core Sound Packages           [ ] CORE-SOUNDS-FR-SIREN7
```

```
Asterisk Module and Build Option Selection

Call Detail Recording             --- Core ---
Channel Event Logging         [*] MOH-OPSOUND-WAV
Channel Drivers               [ ] MOH-OPSOUND-ULAW
Codec Translators             [ ] MOH-OPSOUND-ALAW
Format Interpreters           [*] MOH-OPSOUND-GSM
Dialplan Functions            [*] MOH-OPSOUND-G729
PBX Modules                   [*] MOH-OPSOUND-G722
Resource Modules              [ ] MOH-OPSOUND-SLN16
Test Modules                  [ ] MOH-OPSOUND-SIREN7
Compiler Flags                [ ] MOH-OPSOUND-SIREN14
Utilities
AGI Samples
Core Sound Packages
Music On Hold File Packages
```

```
Asterisk Module and Build Option Selection

Channel Event Logging          [ ] EXTRA-SOUNDS-EN_GB-G729
Channel Drivers                [ ] EXTRA-SOUNDS-EN_GB-G722
Codec Translators              [ ] EXTRA-SOUNDS-EN_GB-SLN16
Format Interpreters            [ ] EXTRA-SOUNDS-EN_GB-SIREN7
Dialplan Functions             [ ] EXTRA-SOUNDS-EN_GB-SIREN14
PBX Modules                    [ ] EXTRA-SOUNDS-FR-WAV
Resource Modules               [ ] EXTRA-SOUNDS-FR-ULAW
Test Modules                   [ ] EXTRA-SOUNDS-FR-ALAW
Compiler Flags                 [*] EXTRA-SOUNDS-FR-GSM
Utilities                      [*] EXTRA-SOUNDS-FR-G729
AGI Samples                    [*] EXTRA-SOUNDS-FR-G722
Core Sound Packages            [ ] EXTRA-SOUNDS-FR-SLN16
Music On Hold File Packages    [ ] EXTRA-SOUNDS-FR-SIREN7
Extras Sound Packages          [ ] EXTRA-SOUNDS-FR-SIREN14

French, G.722.1C (Siren14) format

   Depends on: N/A
      Can use: N/A
Conflicts with: N/A
 Support Level: core

                                          Save & Exit     Exit
```

Une fois la configuration terminée on compile :
make
make install
make samples
make basic-pbx
make config

Selon la puissance de la machine , l'operation peux etre longue .
editer le fichier /etc/asterisk/modules.conf et ajouter : load = res_srtp.so

Une fois terminé on lance Asterisk et on verifie son status :

service asterisk start

service asterisk status

```
root@pbx:/usr/src/asterisk-20.3.0# service asterisk status
● asterisk.service - LSB: Asterisk PBX
     Loaded: loaded (/etc/init.d/asterisk; generated)
     Active: active (running) since Tue 2023-06-20 08:31:46 CEST; 2h 9min ago
       Docs: man:systemd-sysv-generator(8)
      Tasks: 39 (limit: 2265)
     Memory: 34.7M
        CPU: 1min 28.284s
     CGroup: /system.slice/asterisk.service
             └─2373 /usr/sbin/asterisk

juin 20 08:31:46 pbx systemd[1]: Starting asterisk.service - LSB: Asterisk PBX...
juin 20 08:31:46 pbx asterisk[2361]: Starting Asterisk PBX: asterisk.
juin 20 08:31:46 pbx systemd[1]: Started asterisk.service - LSB: Asterisk PBX.
root@pbx:/usr/src/asterisk-20.3.0#
```

On vas faire une sauvegarde des fichiers que l'on vas modifier :
mv /etc/asterisk/pjsip.conf /etc/asterisk/pjsip.conf.origin
mv /etc/asterisk/voicemail.conf /etc/asterisk/voicemail.conf.origin
cp /etc/asterisk/asterisk.conf /etc/asterisk/asterisk.conf.origin

Pour le test on va créer 3 Utilisateurs :
alice, numéro : 6001, mot de passe : bonjour
bob, numéro : 6002, mot de passe : bonjour
martin, numéro : 6003, mot de passe : bonjour

les fichiers de config sont a recuperer sur : https://github.com/thierry-rami/Asterisk_config

Une fois les fichiers de config installés faire :
asterisk -rx "core restart now"

et on vérifie que Astesrisk Tourne :

```
root@pbx:~# ss -nlut | grep -E '5060|State'
Netid   State    Recv-Q   Send-Q     Local Address:Port      Peer Address:Port

udp     UNCONN   0        0          0.0.0.0:5060            0.0.0.0:*
```

La Plateforme Asterisk – From Scratch

On lance la commande asterisk -rvvvvvvvvvv
on se trouve maintenant dans l’invite de commande de asterisk :

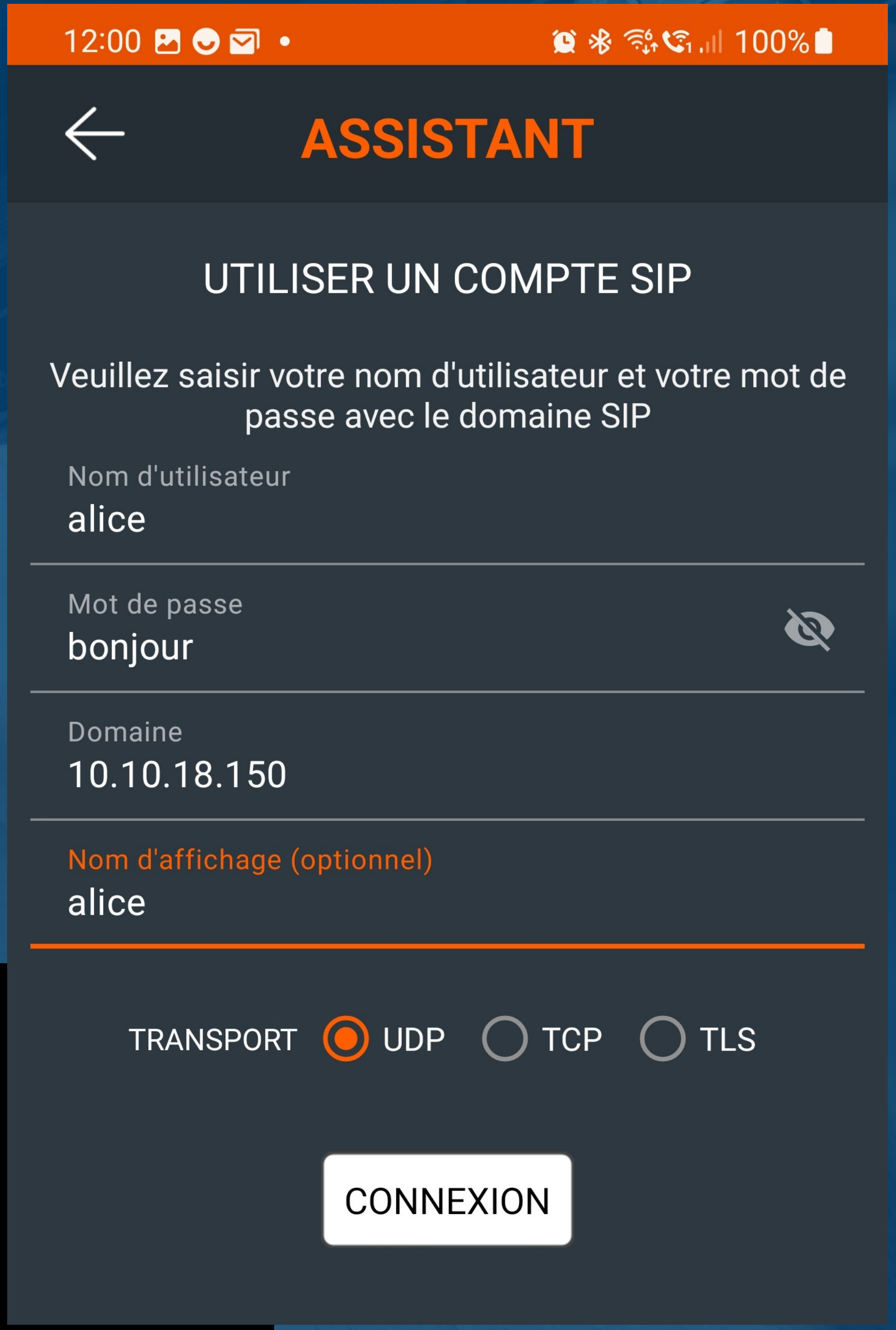
pjsip list aors
pjsip list auths

Normalement on devrait voir les 3 utilisateurs : Alice , Bob , Martin

Maintenant que le serveur Asterisk tourne , on va installer le client sur Windows , Linux , MacOs , Smartphone : Linphone

Dans ce cas je me connecte en tant que « Alice »
et l'adresse IP du serveur asterisk est:10.10.18.150
Une fois connecté dans l'interface CLI d'Asterisk on a :

```
  -- Added contact 'sip:alice@10.10.18.151:44678;transport=udp' to AOR 'alice' with expiration of 3600
== Endpoint alice is now Reachable
  -- Removed contact 'sip:alice@10.10.18.151:44678;transport=udp' from AOR 'alice' due to request
== Contact alice/sip:alice@10.10.18.151:44678;transport=udp has been deleted
== Endpoint alice is now Unreachable
pbx*CLI>
```



La Plateforme

Asterisk – From Scratch

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