

La Plateforme Asterisk – From Scratch

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

on autorise ssh sur le compte root :

```
echo "PermitRootLogin Yes" > /etc/ssh/sshd_config.d/ssh.conf
```

On installe les paquets nécessaires 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
```

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On Télécharge la dernière version d'Asterisk :

```
cd /usr/src  
apt install wget  
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-22-current.tar.gz  
tar xfv asterisk-22-current.tar.gz  
rm asterisk-22-current.tar.gz  
cd asterisk-22*
```

```
./configure  
contrib/scripts/get_mp3_source.sh
```

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On configure les options par : `make menuselect`

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)  --- Extended ---
Applications                      XXX chan_mobile
Bridging Modules                  [*] chan_ooh323
Call Detail Recording             [*] format_mp3
Channel Event Logging            XXX res_config_mysql
Channel Drivers
Codec Translators
Format Interpreters
Dialplan Functions
PBX Modules
Resource Modules
Test Modules
Compiler Flags
Utilities
```

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```
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--- Core ---
[*] app_agent_pool
[*] app_authenticate
[*] app_bridgeaddchan
[*] app_bridgewart
[*] app_cdr
[*] app_celgenuserevent
[*] app_channelredirect
[*] app_chanspy
[*] app_confbridge
[*] app_controlplayback
[*] app_db
[*] app_dial
[*] app_directed_pickup
```

```
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--- Core ---
[*] bridge_builtin_features
[*] bridge_builtin_interval_features
[*] bridge_holding
[*] bridge_native_rtp
[*] bridge_simple
[*] bridge_softmix
--- Module Options ---
XXX binaural_rendering_in_bridge_softmix
```



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Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)

Applications

Bridging Modules

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Utilities

--- Core ---

XXX cdr_adaptive_odbc

[*] cdr_custom

[*] cdr_manager

--- Extended ---

XXX cdr_beanstalkd

[*] cdr_csv

XXX cdr_odbc

XXX cdr_pgsql

XXX cdr_radius

[*] cdr_sqlite3_custom

XXX cdr_tds

Asterisk Module and Build Option Selection

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Channel Drivers

Codec Translators

Format Interpreters

Dialplan Functions

PBX Modules

Resource Modules

Test Modules

Compiler Flags

Utilities

--- Core ---

[*] cel_custom

[*] cel_manager

XXX cel_odbc

--- Extended ---

XXX cel_beanstalkd

XXX cel_pgsql

XXX cel_radius

[*] cel_sqlite3_custom

XXX cel_tds

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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

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Asterisk Module and Build Option Selection	
Add-ons (See README-addons.txt)	--- Core ---
Applications	[*] format_g719
Bridging Modules	[*] format_g723
Call Detail Recording	[*] format_g726
Channel Event Logging	[*] format_g729
Channel Drivers	[*] format_gsm
Codec Translators	[*] format_h263
Format Interpreters	[*] format_h264
Dialplan Functions	[*] format_ilbc
PBX Modules	XXX format_ogg_vorbis
Resource Modules	[*] format_pcm
Test Modules	[*] format_siren14
Compiler Flags	[*] format_siren7
Utilities	[*] format_sln

Asterisk Module and Build Option Selection	
Add-ons (See README-addons.txt)	--- Core ---
Applications	[*] func_aes
Bridging Modules	[*] func_base64
Call Detail Recording	[*] func_blacklist
Channel Event Logging	[*] func_callcompletion
Channel Drivers	[*] func_callerid
Codec Translators	[*] func_cdr
Format Interpreters	[*] func_channel
Dialplan Functions	[*] func_config
PBX Modules	XXX func_curl
Resource Modules	[*] func_cut
Test Modules	[*] func_db
Compiler Flags	[*] func_devstate
Utilities	[*] func_dialgroup

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```
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Utilities

--- Core ---
[*] pbx_config
[*] pbx_loopback
[*] pbx_spool
--- Extended ---
[*] pbx_ael
[*] pbx_dundi
XXX pbx_lua
[*] pbx_realtime
```

```
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Resource Modules
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Utilities

--- Core ---
[*] res_aeap
[*] res_agi
[*] res_ari
[*] res_ari_applications
[*] res_ari_asterisk
[*] res_ari_bridges
[*] res_ari_channels
[*] res_ari_device_states
[*] res_ari_endpoints
[*] res_ari_events
< > res_ari_mailboxes
[*] res_ari_model
[*] res_ari_playbacks
```


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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	[] A02_DEBUG
Utilities	XXX REBUILD_PARSERS



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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			

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Asterisk Module and Build Option Selection

Channel Event Logging	<input type="checkbox"/>	EXTRA-SOUNDS-EN_GB-G729
Channel Drivers	<input type="checkbox"/>	EXTRA-SOUNDS-EN_GB-G722
Codec Translators	<input type="checkbox"/>	EXTRA-SOUNDS-EN_GB-SLN16
Format Interpreters	<input type="checkbox"/>	EXTRA-SOUNDS-EN_GB-SIREN7
Dialplan Functions	<input type="checkbox"/>	EXTRA-SOUNDS-EN_GB-SIREN14
PBX Modules	<input type="checkbox"/>	EXTRA-SOUNDS-FR-WAV
Resource Modules	<input type="checkbox"/>	EXTRA-SOUNDS-FR-ULAW
Test Modules	<input type="checkbox"/>	EXTRA-SOUNDS-FR-ALAW
Compiler Flags	<input checked="" type="checkbox"/>	EXTRA-SOUNDS-FR-GSM
Utilities	<input checked="" type="checkbox"/>	EXTRA-SOUNDS-FR-G729
AGI Samples	<input checked="" type="checkbox"/>	EXTRA-SOUNDS-FR-G722
Core Sound Packages	<input type="checkbox"/>	EXTRA-SOUNDS-FR-SLN16
Music On Hold File Packages	<input type="checkbox"/>	EXTRA-SOUNDS-FR-SIREN7
Extras Sound Packages	<input type="checkbox"/>	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

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Une fois la configuration terminée on compile :

`make`

`make install`

`make samples`

`make basic-pbx`

`make config`

Selon la puissance de la machine , l'opération peut être longue .

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Une fois terminé on lance Asterisk et on vérifie 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#
```

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On va faire une sauvegarde des fichiers que l'on va 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

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Les fichiers de config sont à récupérer 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 Asterisk 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:*

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

```
pjsip list aors  
pjsip list auths
```

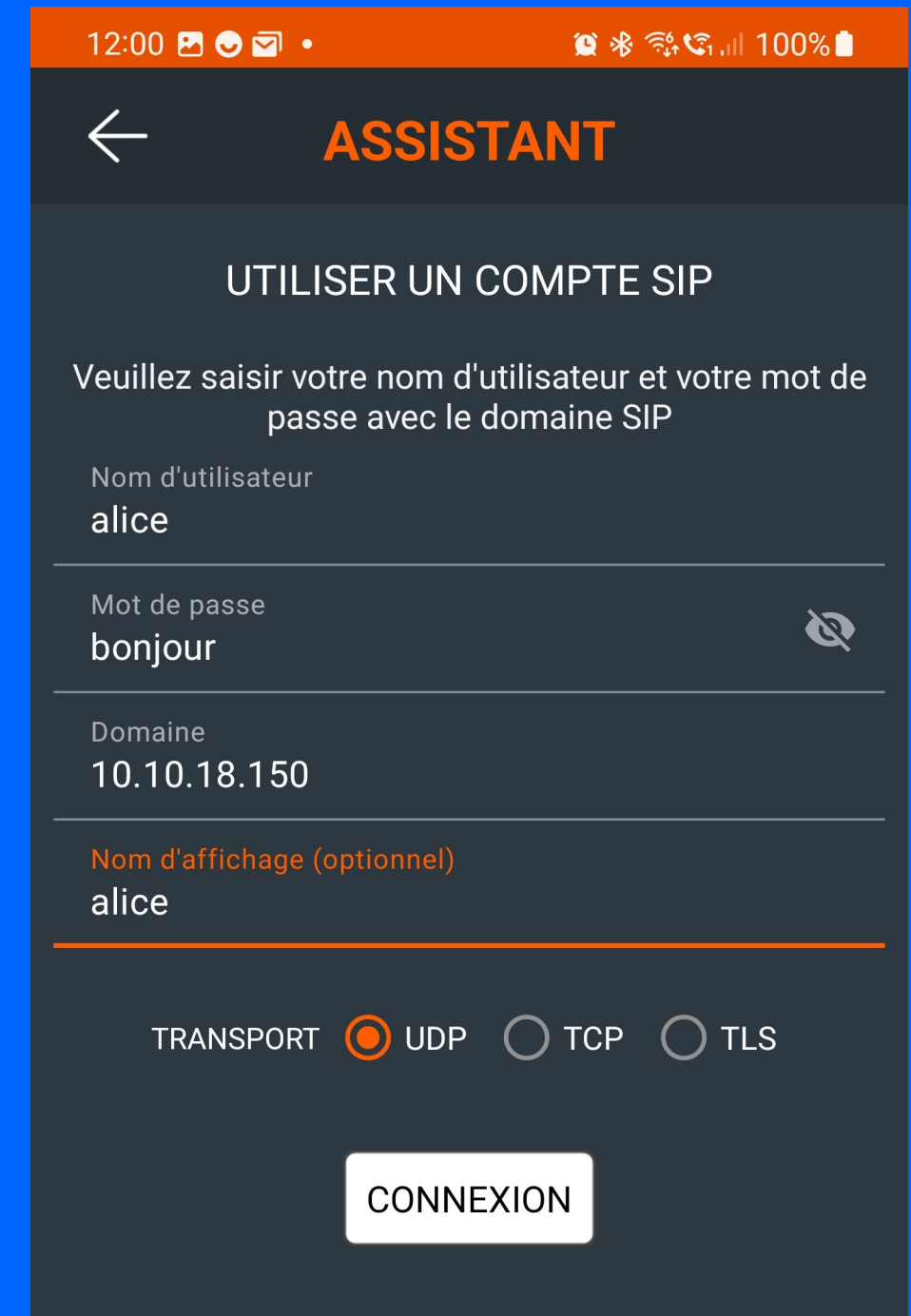
Normalement on devrait voir les 3 utilisateurs : Alice ,
Bob , Martin

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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 seconds
== 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> █
```



12:00 100%

← ASSISTANT

UTILISER UN COMPTE SIP

Veuillez saisir votre nom d'utilisateur et votre mot de passe avec le domaine SIP

Nom d'utilisateur
alice

Mot de passe
bonjour

Domaine
10.10.18.150

Nom d'affichage (optionnel)
alice

TRANSPORT ☒ UDP ☐ TCP ☐ TLS

CONNEXION