



La Plateforme Asterisk – From Scratch

On installe une VM avec Debian V12 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
```

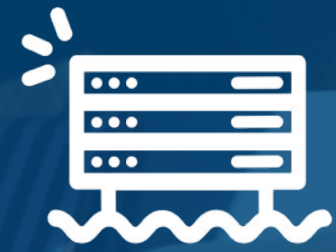
```
echo "IPv6 \6" >> /etc/issue
```

On autorise ssh sur le compte root :

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

On installe les paquets 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
```

La Plateforme Asterisk – From Scratch

On Télécharge la dernière version d'asterisk

```
cd /usr/src
```

```
apt install wget
```

```
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-20-current.tar.gz
```

```
tar xfv asterisk-20-current.tar.gz
```

```
rm asterisk-20-current.tar.gz
```

```
cd asterisk-20*
```

```
./configure
```

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

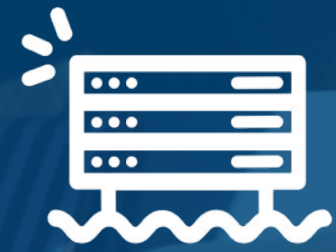



La Plateforme Asterisk – From Scratch

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
```

La Plateforme Asterisk – From Scratch

Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)

Applications

Bridging Modules
Call Detail Recording
Channel Event Logging
Channel Drivers
Codec Translators
Format Interpreters
Dialplan Functions
PBX Modules
Resource Modules
Test Modules
Compiler Flags
Utilities

--- 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

Asterisk Module and Build Option Selection

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Compiler Flags
Utilities

--- 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





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



La Plateforme Asterisk – From Scratch

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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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)

Applications

Bridging Modules

Call Detail Recording

Channel Event Logging

Channel Drivers

Codec Translators

Format Interpreters

Dialplan Functions

PBX Modules

Resource Modules

Test Modules

Compiler Flags

Utilities

--- Core ---

XXX test_abstract_jb

XXX test_acl

XXX test_aeap

XXX test_aeap_speech

XXX test_aeap_transaction

XXX test_aeap_transport

XXX test_amihooks

XXX test_aoc

XXX test_app

XXX test_ari

XXX test_ari_model

XXX test_ast_format_str_red

XXX test_astobj2

Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)

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Call Detail Recording

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Channel Drivers

Codec Translators

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Dialplan Functions

PBX Modules

Resource Modules

Test Modules

Compiler Flags

Utilities

--- Core ---

[] DONT_OPTIMIZE

< > COMPILE_DOUBLE

[] DEBUG_THREADS

[] DEBUG_FD_LEAKS

XXX BETTER_BACKTRACES

[] LOTS_OF_SPANS

[] MALLOC_DEBUG

[] DEBUG_CHAOS

[*] BUILD_NATIVE

--- Extended ---

[] REF_DEBUG

[] A02_DEBUG

XXX REBUILD_PARSERS



Thierry Rami



La Plateforme Asterisk – From Scratch

Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)

Applications

Bridging Modules

Call Detail Recording

Channel Event Logging

Channel Drivers

Codec Translators

Format Interpreters

Dialplan Functions

PBX Modules

Resource Modules

Test Modules

Compiler Flags

Utilities

--- Core ---

[*] astcanary

[*] astdb2sqlite3

[*] astdb2bdb

--- Extended ---

[] check_expr

[] check_expr2

XXX smsq

[] stereorize

[] streamplayer

XXX conf_bridge_binaural_hri

--- Deprecated ---

[] aelparse

[] astman

Asterisk Module and Build Option Selection

--- Extended ---

[] agi-test.agi

[] eagi-test

[] eagi-sphinx-test

[] jukebox.agi

Bridging Modules

Call Detail Recording

Channel Event Logging

Channel Drivers

Codec Translators

Format Interpreters

Dialplan Functions

PBX Modules

Resource Modules

Test Modules

Compiler Flags

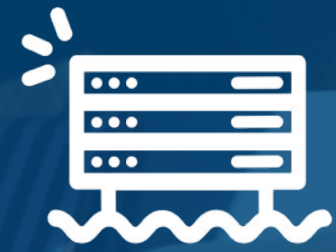
Utilities

AGI Samples

Core Sound Packages



Thierry Rami



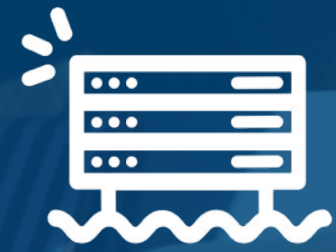
La Plateforme Asterisk – From Scratch

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	



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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



La Plateforme Asterisk – From Scratch

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.



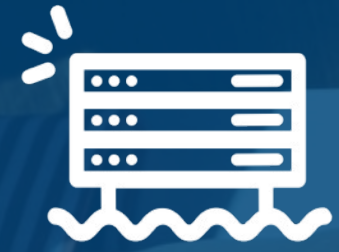
La Plateforme Asterisk – From Scratch

Une fois terminé, on lance Asterisk et on vérifie son statut :

```
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#
```





La Plateforme Asterisk – From Scratch

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



La Plateforme Asterisk – From Scratch

Les fichiers de config sont à récupérer sur : https://github.com/thierry-rami/Asterisk_config
Et a copier dans /etc/asterisk

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

et on vérifie que Asterisk Toune :

```
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 d'asterisk :

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

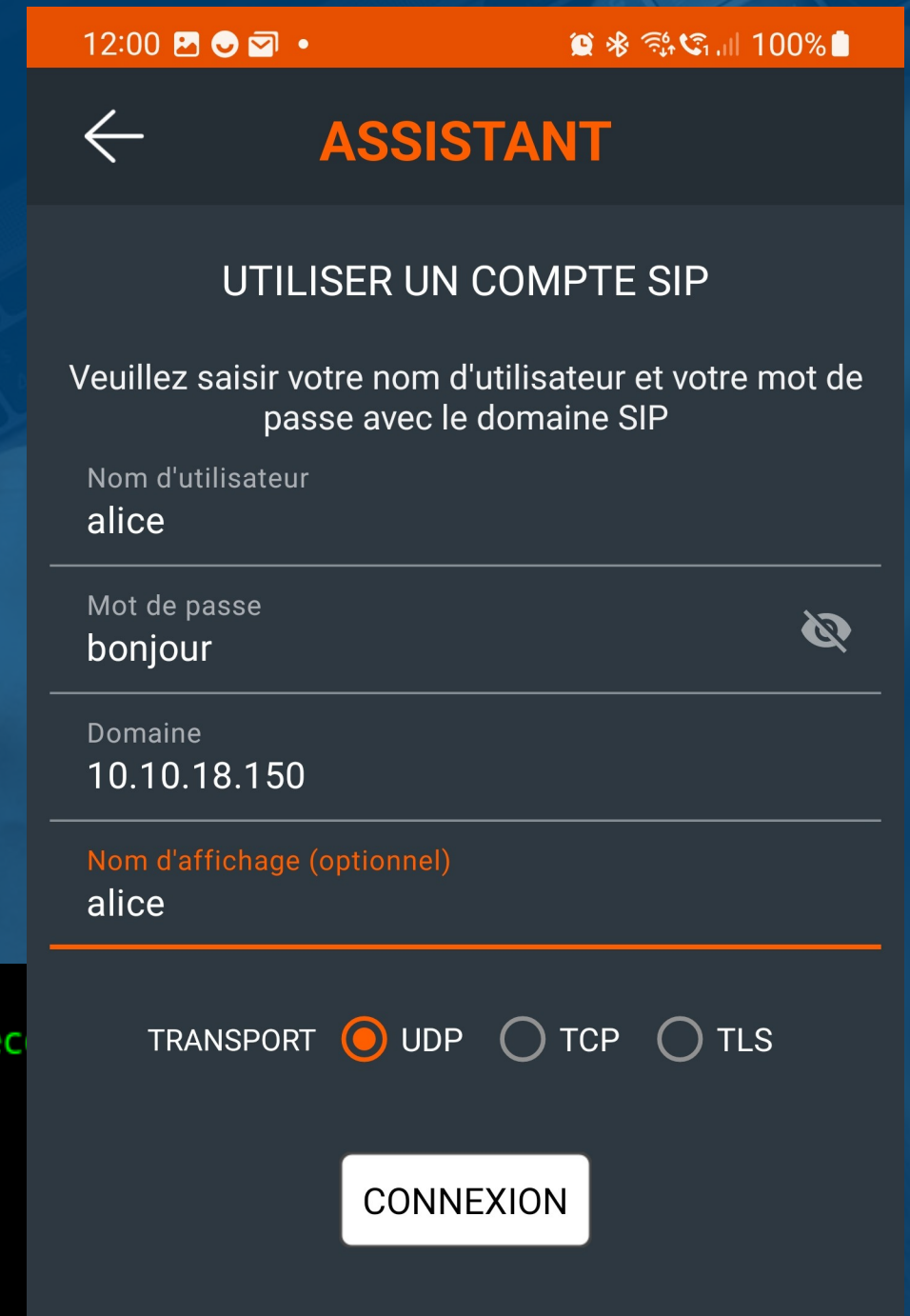
Normalement, on devrait voir les 3 utilisateurs : Alice , Bob , Martin

La Plateforme Asterisk – From Scratch

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 sec
== 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> █
```



The screenshot shows the 'ASSISTANT' screen of the Linphone app. At the top, there's a back arrow and the title 'ASSISTANT'. Below it, the section 'UTILISER UN COMPTE SIP' is displayed. A prompt asks the user to enter their SIP account details. The form includes fields for 'Nom d'utilisateur' (alice), 'Mot de passe' (bonjour), and 'Domaine' (10.10.18.150). There is also an optional field for 'Nom d'affichage' (alice). At the bottom, there are radio buttons for 'TRANSPORT' with options UDP (selected), TCP, and TLS. A 'CONNEXION' button is located at the very bottom.

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