

Serveur_Voip

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Etape1 : Configurer le fichier sources.list : `nano /etc/apt/sources.list`

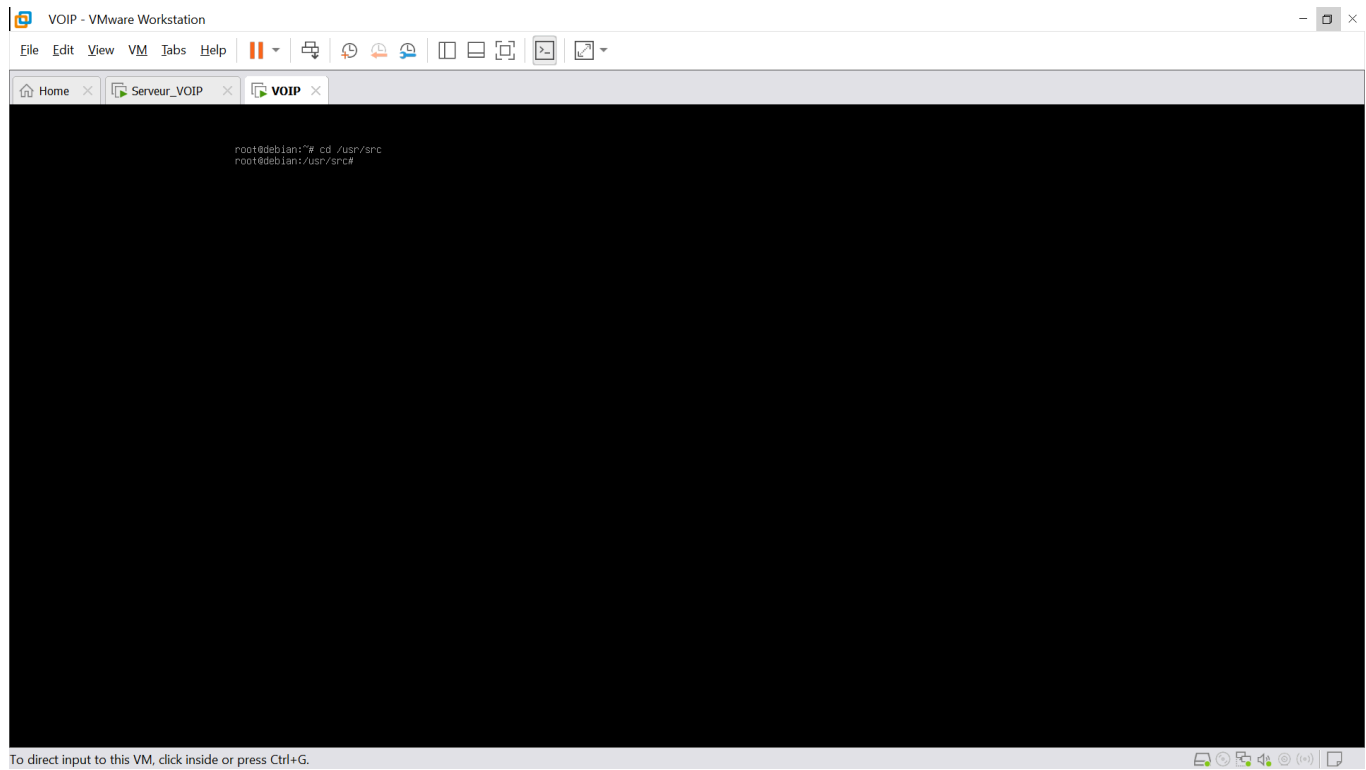
```

GNU nano 7.2 /etc/apt/sources.list
#deb cdrom:[Debian GNU/Linux 12.9.0 "Bookworm" - Official amd64 DVD Binary-1 with Firmware 20250111-10:55]/bookworm contrib main non-free-firmware
deb https://deb.debian.org/debian/ bookworm main contrib non-free non-free-firmware
deb-src https://deb.debian.org/debian/ bookworm main non-free-firmware
deb https://security.debian.org/debian-security bookworm-security main contrib non-free non-free-firmware
deb-src https://security.debian.org/debian-security bookworm-security main non-free-firmware
deb https://deb.debian.org/debian/ bookworm-updates main contrib non-free non-free-firmware
deb-src https://deb.debian.org/debian/ bookworm-updates main non-free-firmware
deb https://deb.debian.org/debian bookworm-backports main contrib non-free non-free-firmware
  
```

Etape 2 : Mettre à jour le système : `sudo apt update && sudo apt upgrade -y`

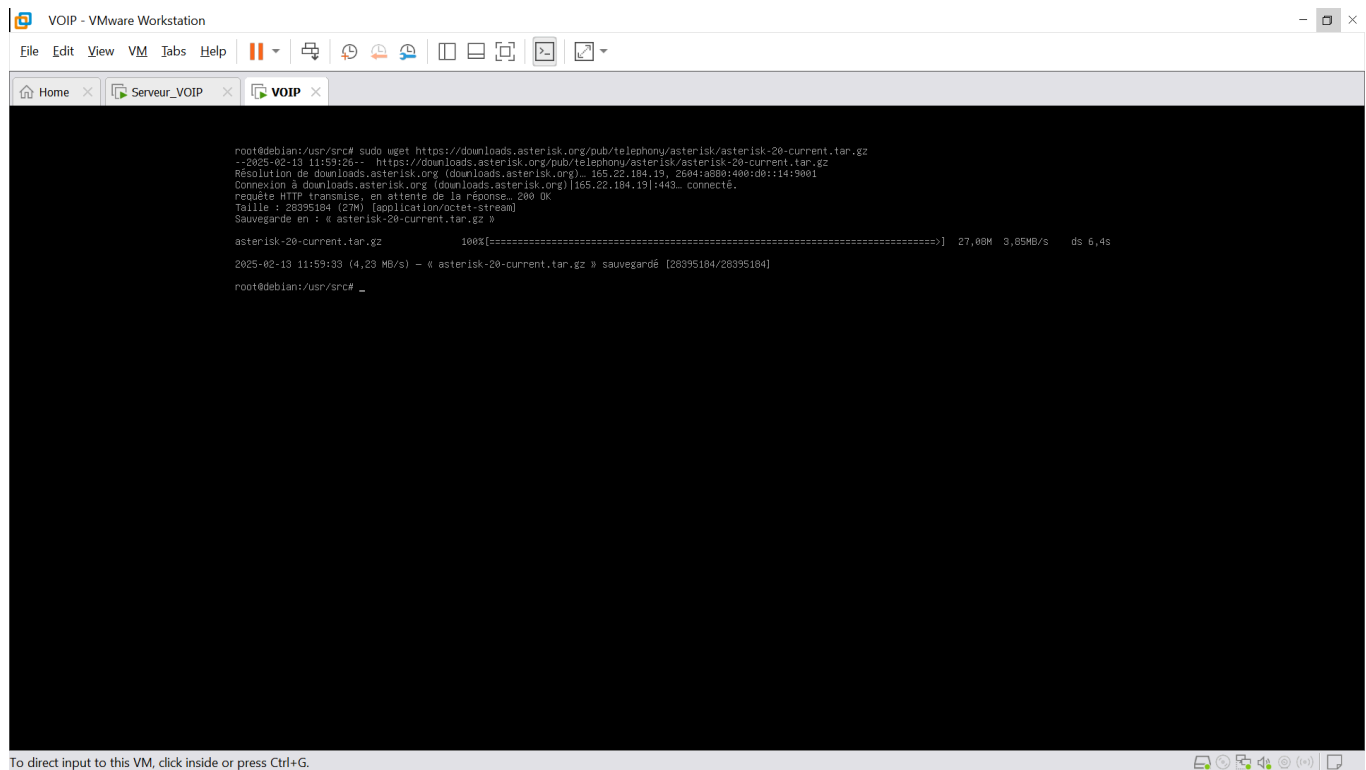
Etape 3 : Installer les dépendances nécessaires : `sudo apt install build-essential wget libncurses5-dev libssl-dev libxml2-dev libsqlite3-dev uuid-dev -y`

Etape 4 : Se déplacer dans le répertoire /usr/src : `cd /usr/src`

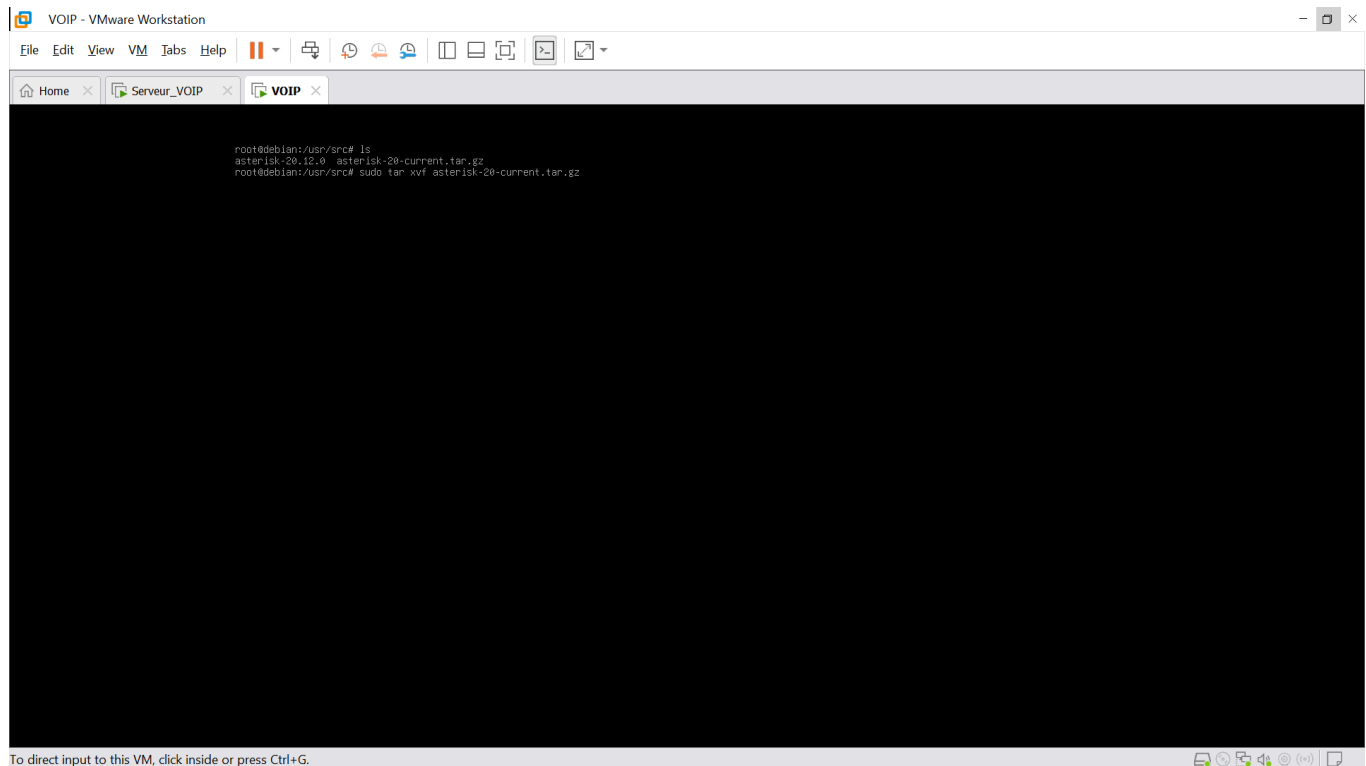


Etape 5 : Télécharger l'archive contenant les fichiers source d'Asterisk : `wget`

<http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-20.tar.gz>



Etape 6 : Décompresser l'archive téléchargée : tar -xzf asterisk-20.tar.gz



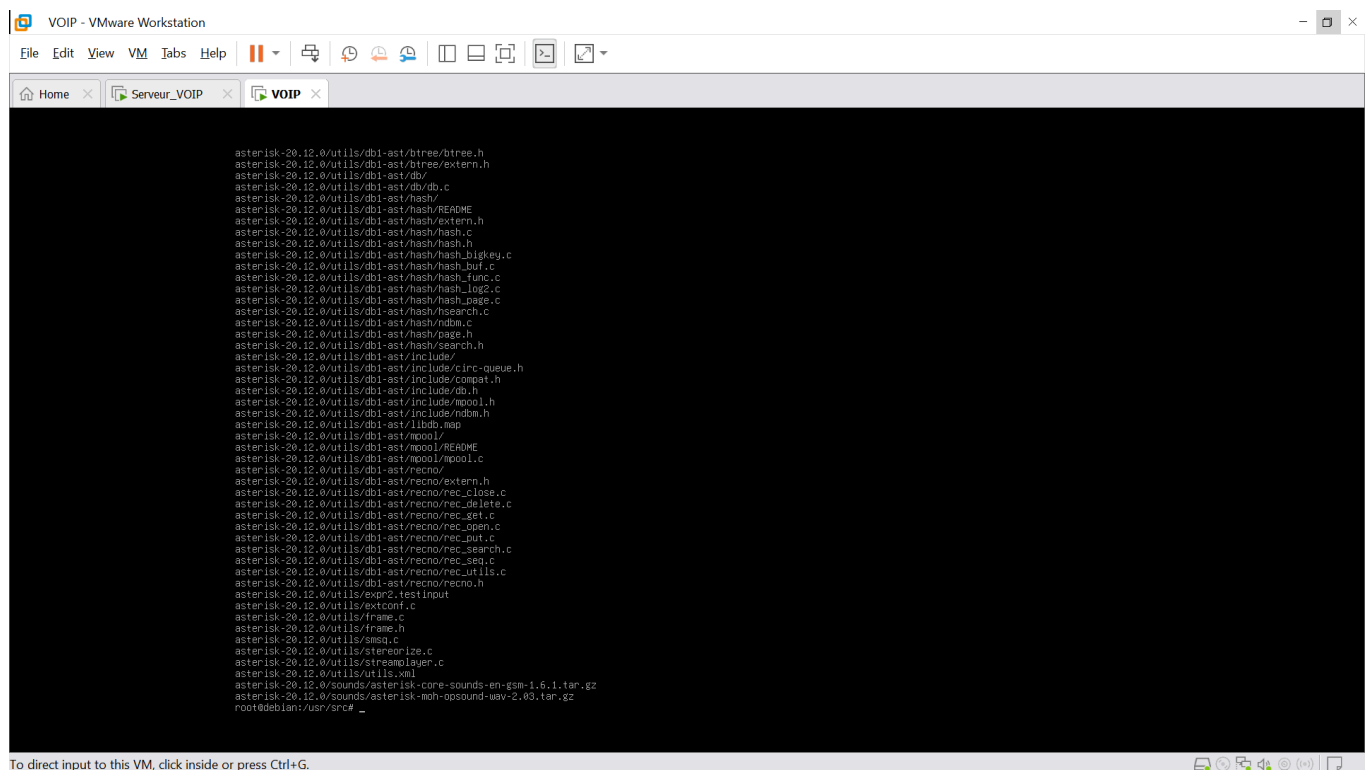
VOIP - VMware Workstation

File Edit View VM Tabs Help

Home x Serveur_VOIP x VOIP x

```
root@debian:/usr/src# ls
asterisk-20.12.0 asterisk-20-current.tar.gz
root@debian:/usr/src# sudo tar -xvf asterisk-20-current.tar.gz
```

To direct input to this VM, click inside or press Ctrl+G.



VOIP - VMware Workstation

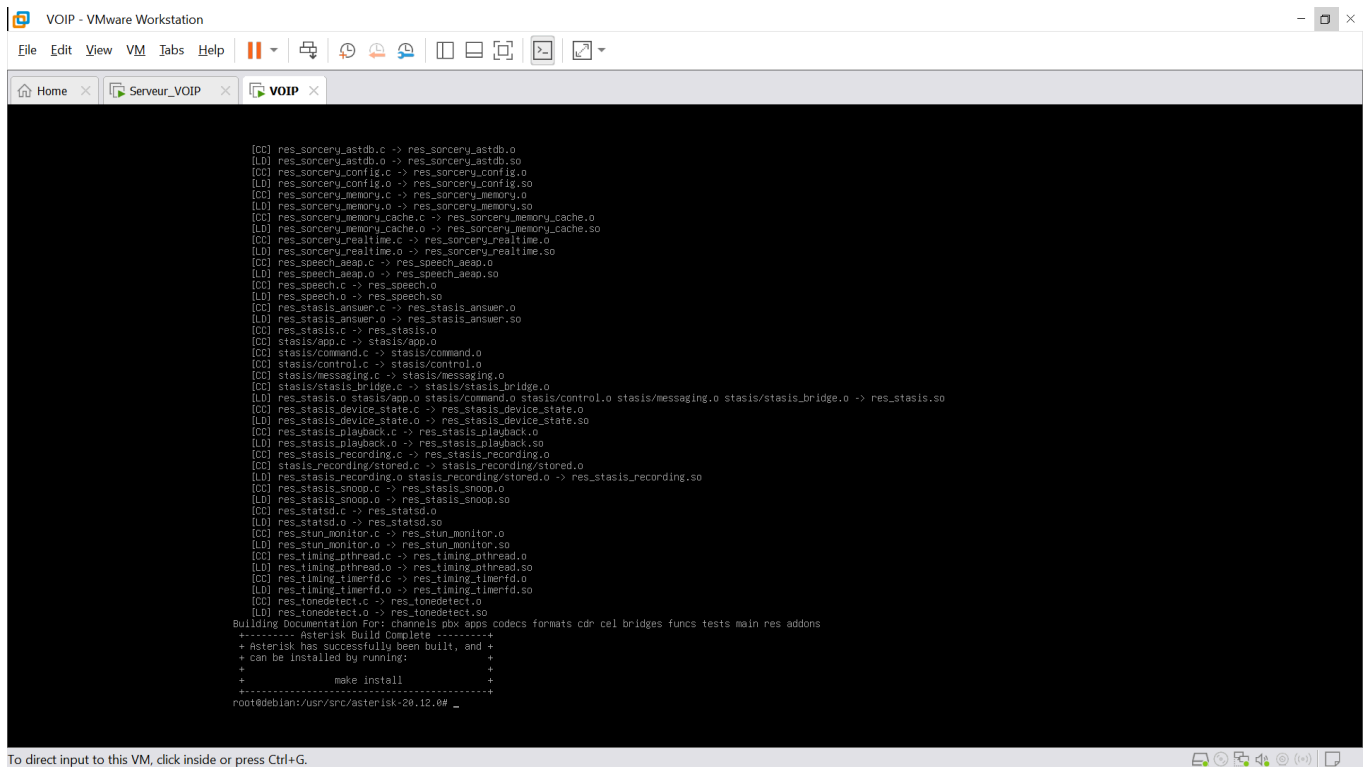
File Edit View VM Tabs Help

Home x Serveur_VOIP x VOIP x

```
asterisk-20.12.0/utills/db1-ast/btree/btree.h
asterisk-20.12.0/utills/db1-ast/btree/extern.h
asterisk-20.12.0/utills/db1-ast/db/
asterisk-20.12.0/utills/db1-ast/db/db.c
asterisk-20.12.0/utills/db1-ast/hash/
asterisk-20.12.0/utills/db1-ast/hash/README
asterisk-20.12.0/utills/db1-ast/hash/extern.h
asterisk-20.12.0/utills/db1-ast/hash/hash.c
asterisk-20.12.0/utills/db1-ast/hash/hash.h
asterisk-20.12.0/utills/db1-ast/hash/hash_bigkey.c
asterisk-20.12.0/utills/db1-ast/hash/hash_buf.c
asterisk-20.12.0/utills/db1-ast/hash/hash_func.c
asterisk-20.12.0/utills/db1-ast/hash/hash_log2.c
asterisk-20.12.0/utills/db1-ast/hash/hash_page.c
asterisk-20.12.0/utills/db1-ast/hash/hash_search.c
asterisk-20.12.0/utills/db1-ast/hash/ndbm.c
asterisk-20.12.0/utills/db1-ast/hash/page.h
asterisk-20.12.0/utills/db1-ast/hash/search.h
asterisk-20.12.0/utills/db1-ast/include/
asterisk-20.12.0/utills/db1-ast/include/circ-queue.h
asterisk-20.12.0/utills/db1-ast/include/compat.h
asterisk-20.12.0/utills/db1-ast/include/db.h
asterisk-20.12.0/utills/db1-ast/include/mpool.h
asterisk-20.12.0/utills/db1-ast/include/ndbm.h
asterisk-20.12.0/utills/db1-ast/libdb.m4
asterisk-20.12.0/utills/db1-ast/mpool/
asterisk-20.12.0/utills/db1-ast/mpool/README
asterisk-20.12.0/utills/db1-ast/mpool/mpool.c
asterisk-20.12.0/utills/db1-ast/recno/
asterisk-20.12.0/utills/db1-ast/recno/extern.h
asterisk-20.12.0/utills/db1-ast/recno/rec_close.c
asterisk-20.12.0/utills/db1-ast/recno/rec_delete.c
asterisk-20.12.0/utills/db1-ast/recno/rec_get.c
asterisk-20.12.0/utills/db1-ast/recno/rec_open.c
asterisk-20.12.0/utills/db1-ast/recno/rec_put.c
asterisk-20.12.0/utills/db1-ast/recno/rec_search.c
asterisk-20.12.0/utills/db1-ast/recno/rec_seq.c
asterisk-20.12.0/utills/db1-ast/recno/rec_util.c
asterisk-20.12.0/utills/db1-ast/recno/recno.h
asterisk-20.12.0/utills/expr2.testinput
asterisk-20.12.0/utills/extconf.c
asterisk-20.12.0/utills/frame.c
asterisk-20.12.0/utills/frame.h
asterisk-20.12.0/utills/smsq.c
asterisk-20.12.0/utills/stereo12.c
asterisk-20.12.0/utills/streamplayer.c
asterisk-20.12.0/utills/utills.xml
asterisk-20.12.0/sounds/asterisk-core-sounds-en-gsm-1.6.1.tar.gz
asterisk-20.12.0/sounds/asterisk-moh-opsound-wav-2.63.tar.gz
root@debian:/usr/src#
```

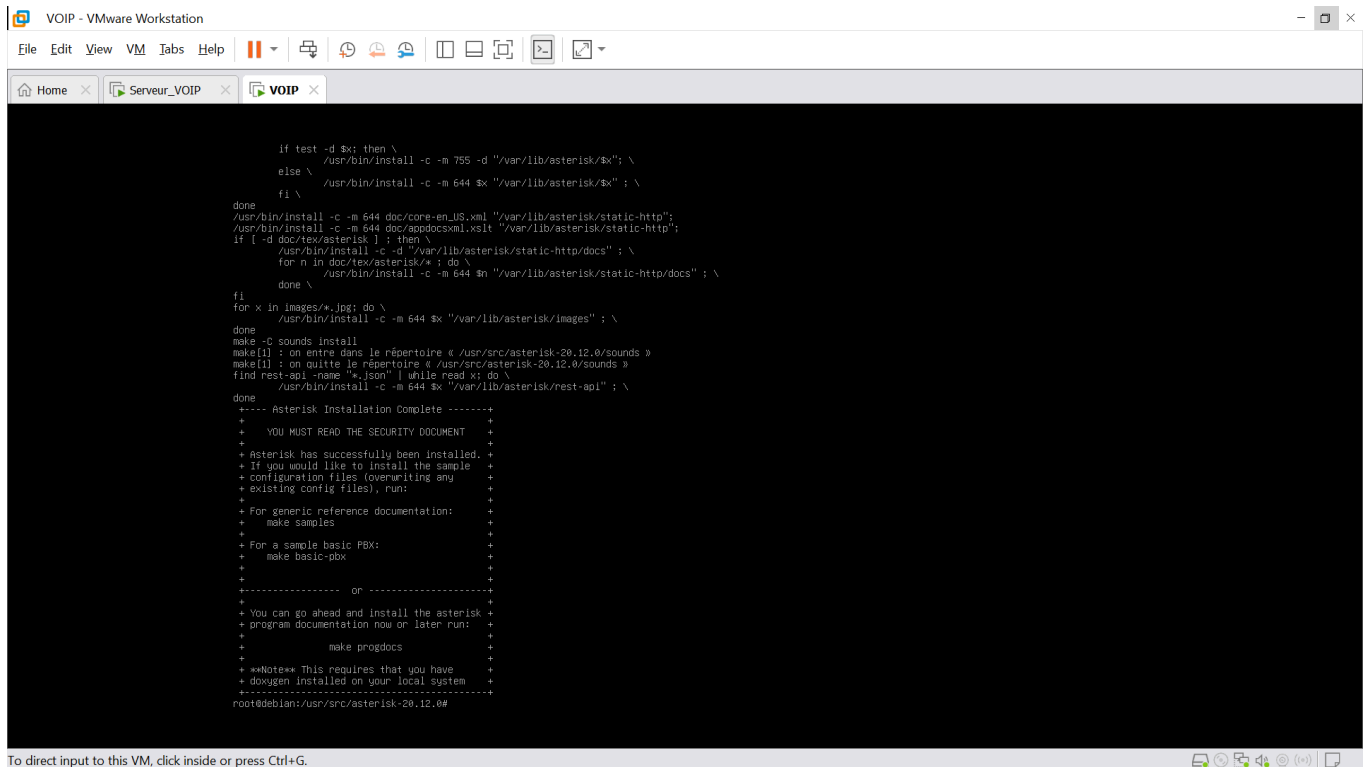
To direct input to this VM, click inside or press Ctrl+G.

Etape 7 : Se déplacer dans le dossier contenant les fichiers source extraits : cd asterisk-20



```
[CC] res_sorcery_astdb.o -> res_sorcery_astdb.o
[LD] res_sorcery_astdb.o -> res_sorcery_astdb.so
[CC] res_sorcery_config.o -> res_sorcery_config.o
[LD] res_sorcery_config.o -> res_sorcery_config.so
[CC] res_sorcery_memory.o -> res_sorcery_memory.o
[LD] res_sorcery_memory.o -> res_sorcery_memory.so
[CC] res_sorcery_memory_cache.o -> res_sorcery_memory_cache.o
[LD] res_sorcery_memory_cache.o -> res_sorcery_memory_cache.so
[CC] res_sorcery_realtime.o -> res_sorcery_realtime.o
[LD] res_sorcery_realtime.o -> res_sorcery_realtime.so
[CC] res_speech_aeap.o -> res_speech_aeap.o
[LD] res_speech_aeap.o -> res_speech_aeap.so
[CC] res_speech.o -> res_speech.o
[LD] res_speech.o -> res_speech.so
[CC] res_stasis_answer.o -> res_stasis_answer.o
[LD] res_stasis_answer.o -> res_stasis_answer.so
[CC] res_stasis.o -> res_stasis.o
[LD] res_stasis.o -> res_stasis.so
[CC] stasis/app.o -> stasis/app.o
[CC] stasis/command.o -> stasis/command.o
[LD] stasis/command.o -> stasis/command.so
[CC] stasis/control.o -> stasis/control.o
[LD] stasis/control.o -> stasis/control.so
[CC] stasis/messaging.o -> stasis/messaging.o
[LD] stasis/messaging.o -> stasis/messaging.so
[CC] stasis/stasis_bridge.o -> stasis/stasis_bridge.o
[LD] res_stasis.o stasis/app.o stasis/command.o stasis/control.o stasis/messaging.o stasis/stasis_bridge.o -> res_stasis.so
[CC] res_stasis_device_state.o -> res_stasis_device_state.o
[LD] res_stasis_device_state.o -> res_stasis_device_state.so
[CC] res_stasis_playback.o -> res_stasis_playback.o
[LD] res_stasis_playback.o -> res_stasis_playback.so
[CC] res_stasis_recording.o -> res_stasis_recording.o
[LD] res_stasis_recording.o -> res_stasis_recording.so
[CC] stasis_recording/stored.o -> stasis_recording/stored.o
[LD] res_stasis_recording.o stasis_recording/stored.o -> res_stasis_recording.so
[CC] res_stasis_snoop.o -> res_stasis_snoop.o
[LD] res_stasis_snoop.o -> res_stasis_snoop.so
[CC] res_stats.o -> res_stats.o
[LD] res_stats.o -> res_stats.so
[CC] res_stun_monitor.o -> res_stun_monitor.o
[LD] res_stun_monitor.o -> res_stun_monitor.so
[CC] res_timing_thread.o -> res_timing_thread.o
[LD] res_timing_thread.o -> res_timing_thread.so
[CC] res_timing_timerfd.o -> res_timing_timerfd.o
[LD] res_timing_timerfd.o -> res_timing_timerfd.so
[CC] res_tonedetect.o -> res_tonedetect.o
[LD] res_tonedetect.o -> res_tonedetect.so
Building documentation For: channels pbx apps codecs formats cdr cel bridges funcs tests main res addons
+----- Asterisk Build Complete -----+
+ Asterisk has successfully been built, and +
+ can be installed by running:             +
+                                           +
+ make install                             +
+-----+
root@debian:/usr/src/asterisk-20.12.0#
```

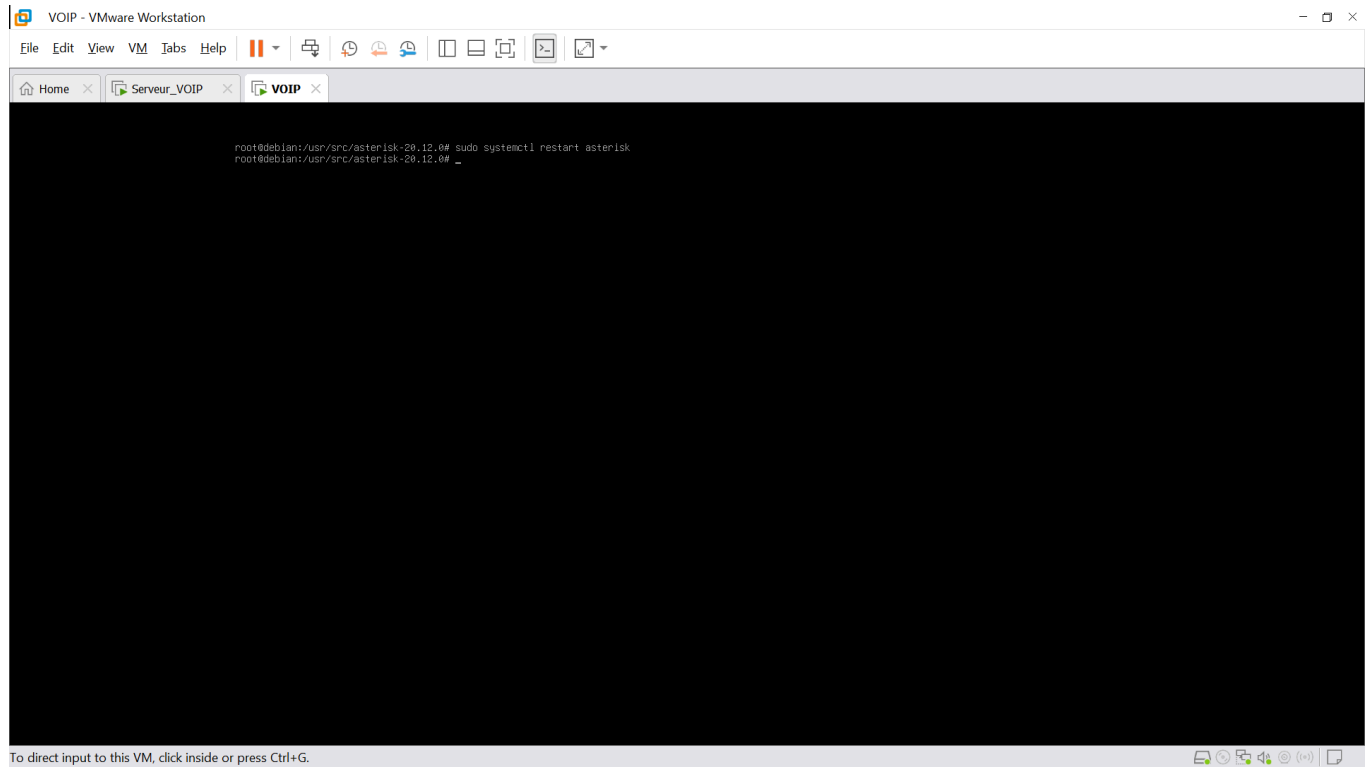
Etape 10 : Installer Asterisk sur le système : sudo make install



```
if test -d $x; then \
    /usr/bin/install -c -m 755 -d "/var/lib/asterisk/$x" ; \
else \
    /usr/bin/install -c -m 644 $x "/var/lib/asterisk/$x" ; \
fi \
done
/usr/bin/install -c -m 644 doc/core-en_US.xml "/var/lib/asterisk/static-http";
/usr/bin/install -c -m 644 doc/appdocs.xml.xslt "/var/lib/asterisk/static-http";
if [ -d doc/asterisk ]; then \
    /usr/bin/install -c -d "/var/lib/asterisk/static-http/docs" ; \
    for n in doc/asterisk/* ; do \
        /usr/bin/install -c -m 644 $n "/var/lib/asterisk/static-http/docs" ; \
    done \
fi
for x in Images/*.jpg; do \
    /usr/bin/install -c -m 644 $x "/var/lib/asterisk/Images" ; \
done
make -C sounds install
make[1]: on entre dans le repertoire « /usr/src/asterisk-20.12.0/sounds »
make[1]: on quitte le repertoire « /usr/src/asterisk-20.12.0/sounds »
find rest-api -name "*.json" | while read x; do \
    /usr/bin/install -c -m 644 $x "/var/lib/asterisk/rest-api" ; \
done
+----- Asterisk Installation Complete -----+
+ YOU MUST READ THE SECURITY DOCUMENT      +
+                                           +
+ Asterisk has successfully been installed. +
+ If you would like to install the sample  +
+ configuration files (overwriting any     +
+ existing config files), run:             +
+                                           +
+ For generic reference documentation:     +
+ make samples                             +
+                                           +
+ For a sample basic PBX:                   +
+ make basic-pbx                           +
+                                           +
+----- on -----+
+ You can go ahead and install the asterisk +
+ program documentation now or later run:   +
+                                           +
+ make progdocs                             +
+                                           +
+ **Note** This requires that you have     +
+ doxygen installed on your local system   +
+-----+
root@debian:/usr/src/asterisk-20.12.0#
```

Etape 11 : Installer les fichiers de configuration : sudo make samples

Etape 13 : Redémarrer Asterisk : `sudo systemctl restart asterisk`

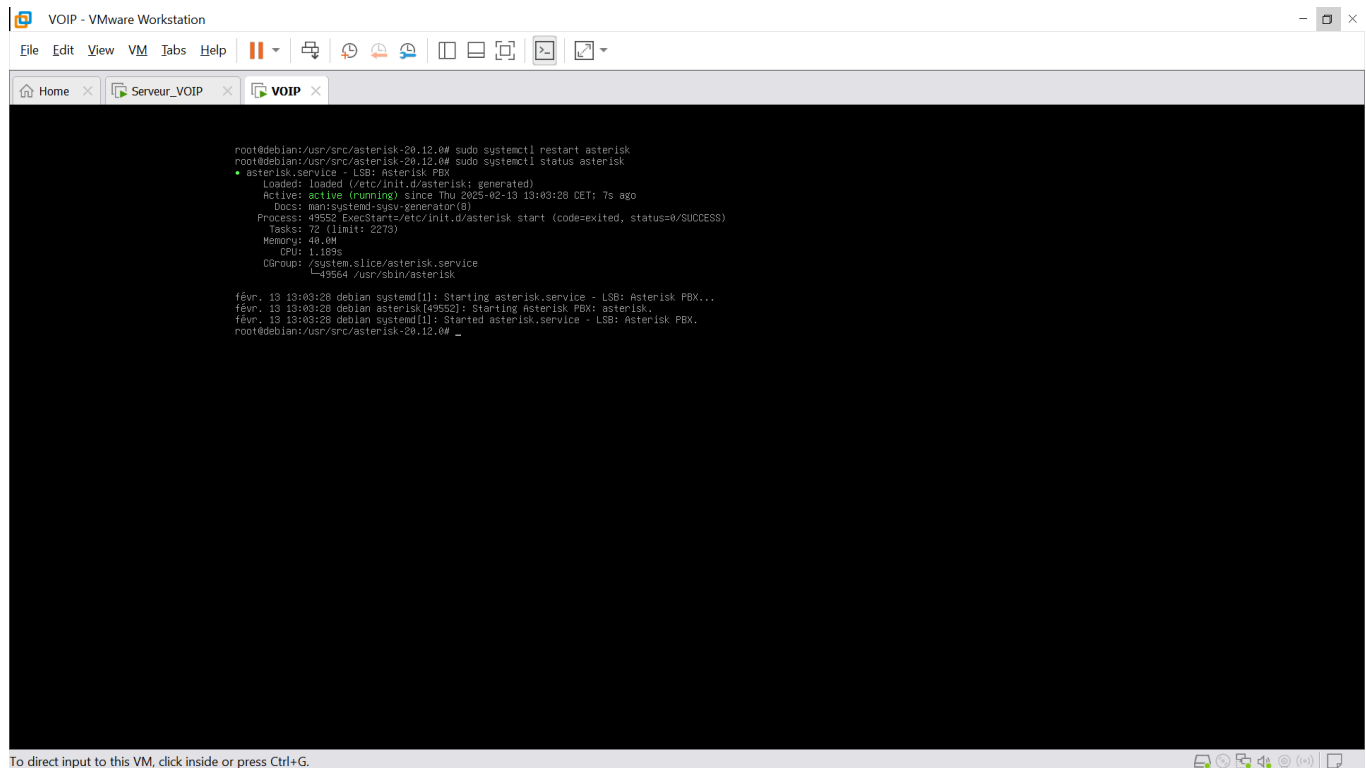


The screenshot shows a terminal window titled "VOIP - VMware Workstation". The terminal content is as follows:

```
root@debian:/usr/src/asterisk-20.12.0# sudo systemctl restart asterisk
root@debian:/usr/src/asterisk-20.12.0# _
```

The terminal window has a menu bar with "File", "Edit", "View", "VM", "Tabs", and "Help". Below the menu bar is a toolbar with various icons. The terminal window is part of a larger interface with tabs for "Home", "Serveur_VOIP", and "VOIP". At the bottom of the terminal window, there is a status bar that reads "To direct input to this VM, click inside or press Ctrl+G."

Etape 14 : Vérifier que Asterisk est activé : `sudo systemctl status asterisk`



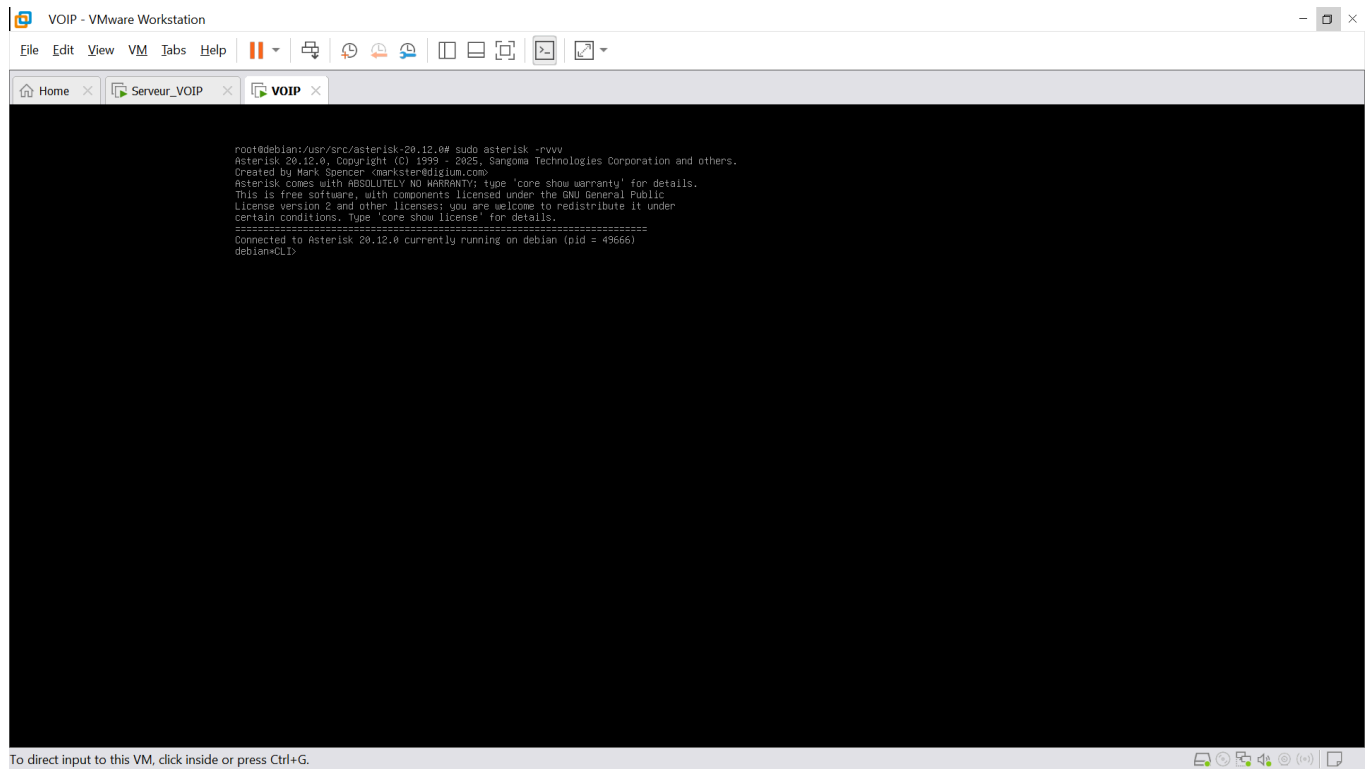
The screenshot shows a terminal window titled "VOIP - VMware Workstation". The terminal content is as follows:

```
root@debian:/usr/src/asterisk-20.12.0# sudo systemctl restart asterisk
root@debian:/usr/src/asterisk-20.12.0# sudo systemctl status asterisk
● asterisk.service - LSB: Asterisk PBX
   Loaded: loaded (/etc/init.d/asterisk; generated)
   Active: active (running) since Thu 2025-02-13 13:03:20 CET; 7s ago
     Docs: man:systemd-sysv-generator(8)
   Process: 49552 ExecStart=/etc/init.d/asterisk start (code=exited, status=0/SUCCESS)
    Tasks: 72 (limit: 2273)
   Memory: 48.0M
      CPU: 1.189s
   CGroup: /system.slice/asterisk.service
           └─49564 /usr/sbin/asterisk

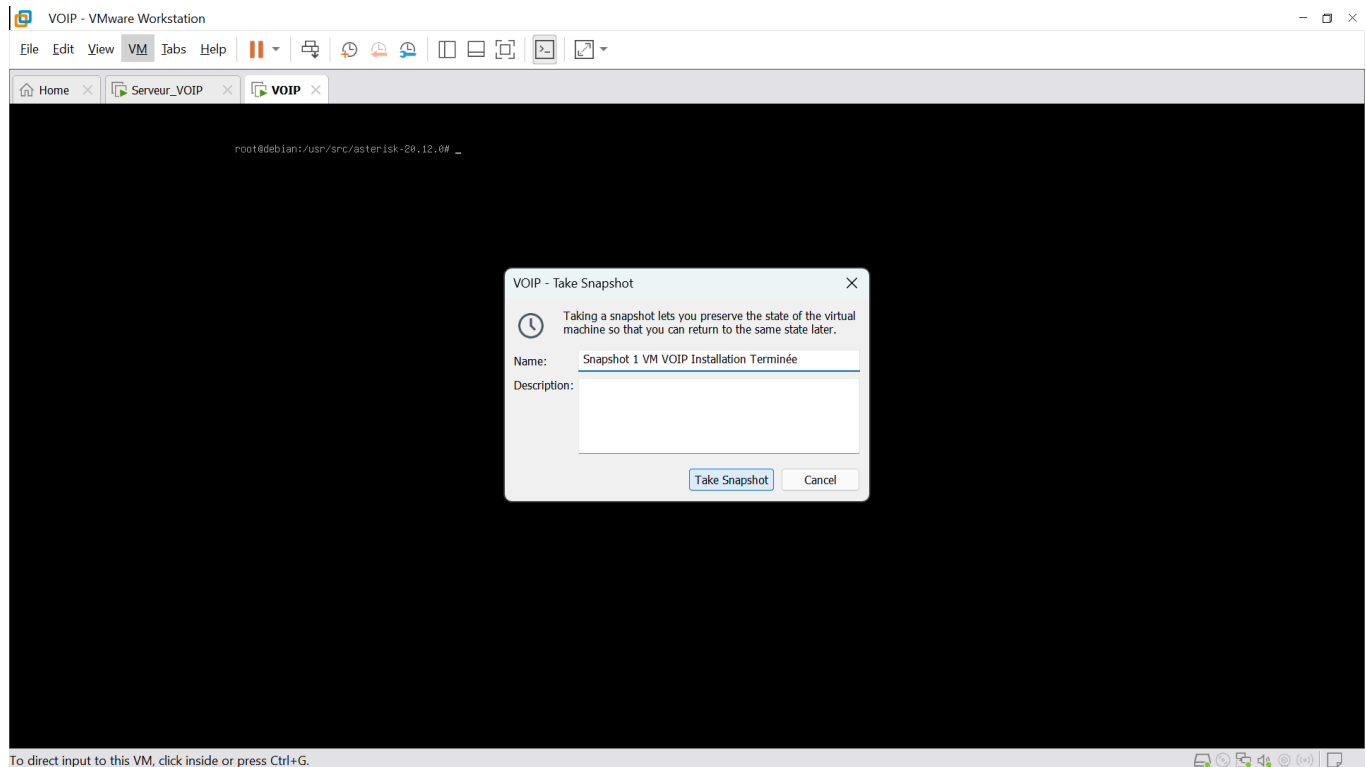
févr. 13 13:03:20 debian systemd[1]: Starting asterisk.service - LSB: Asterisk PBX...
févr. 13 13:03:20 debian asterisk[49552]: Starting Asterisk PBX: asterisk.
févr. 13 13:03:20 debian systemd[1]: Started asterisk.service - LSB: Asterisk PBX.
root@debian:/usr/src/asterisk-20.12.0# _
```

The terminal window has a menu bar with "File", "Edit", "View", "VM", "Tabs", and "Help". Below the menu bar is a toolbar with various icons. The terminal window is part of a larger interface with tabs for "Home", "Serveur_VOIP", and "VOIP". At the bottom of the terminal window, there is a status bar that reads "To direct input to this VM, click inside or press Ctrl+G."

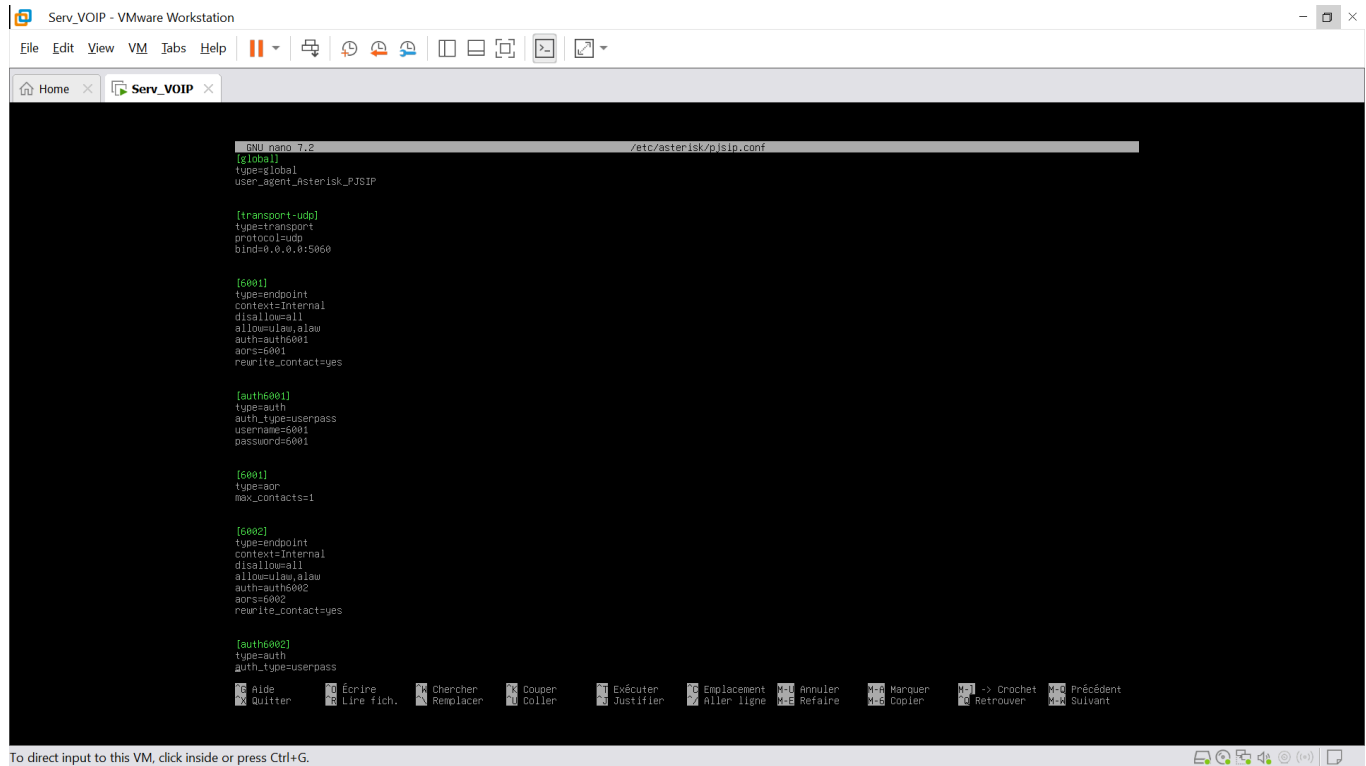
Etape 15 : Lancer le CLI d'Asterisk : sudo asterisk -rvvv



Etape 16 : Sauvegarder la VM : Snapshot 1



Etape 17 : Configurer vos utilisateurs SIP : sudo nano /etc/asterisk/pjsip.conf



```
GNU nano 7.2 /etc/asterisk/pjsip.conf
[global]
type=global
user_agent_asterisk_pjsip

[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0:5060

[6001]
type=endpoints
context=internal
disallow=all
allow=ulaw,alaw
auth=auth6001
aors=6001
rewrite_contact=yes

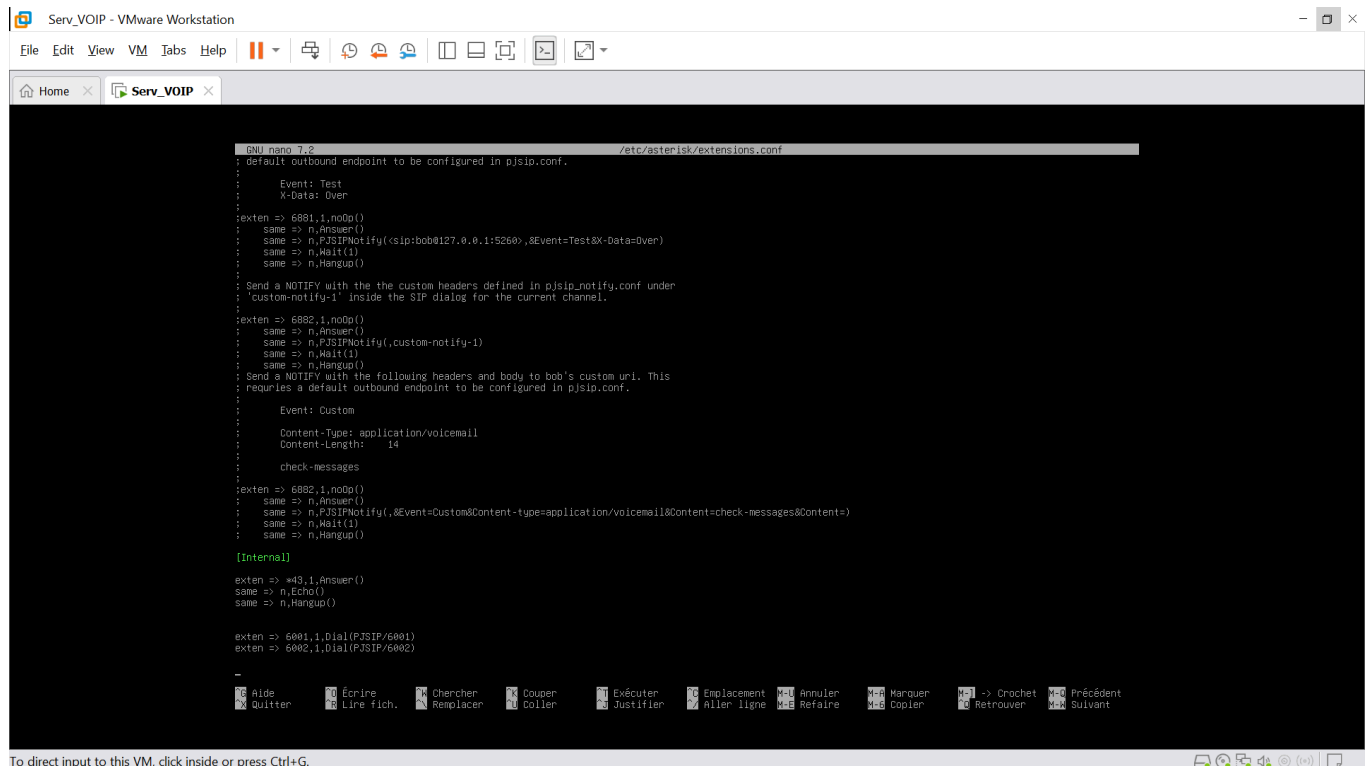
[auth6001]
type=auth
auth_type=userpass
username=6001
password=6001

[6001]
type=aor
max_contacts=1

[6002]
type=endpoints
context=internal
disallow=all
allow=ulaw,alaw
auth=auth6002
aors=6002
rewrite_contact=yes

[auth6002]
type=auth
auth_type=userpass
```

Etape 18 : Configurer le plan d'appel : sudo nano /etc/asterisk/extensions.conf

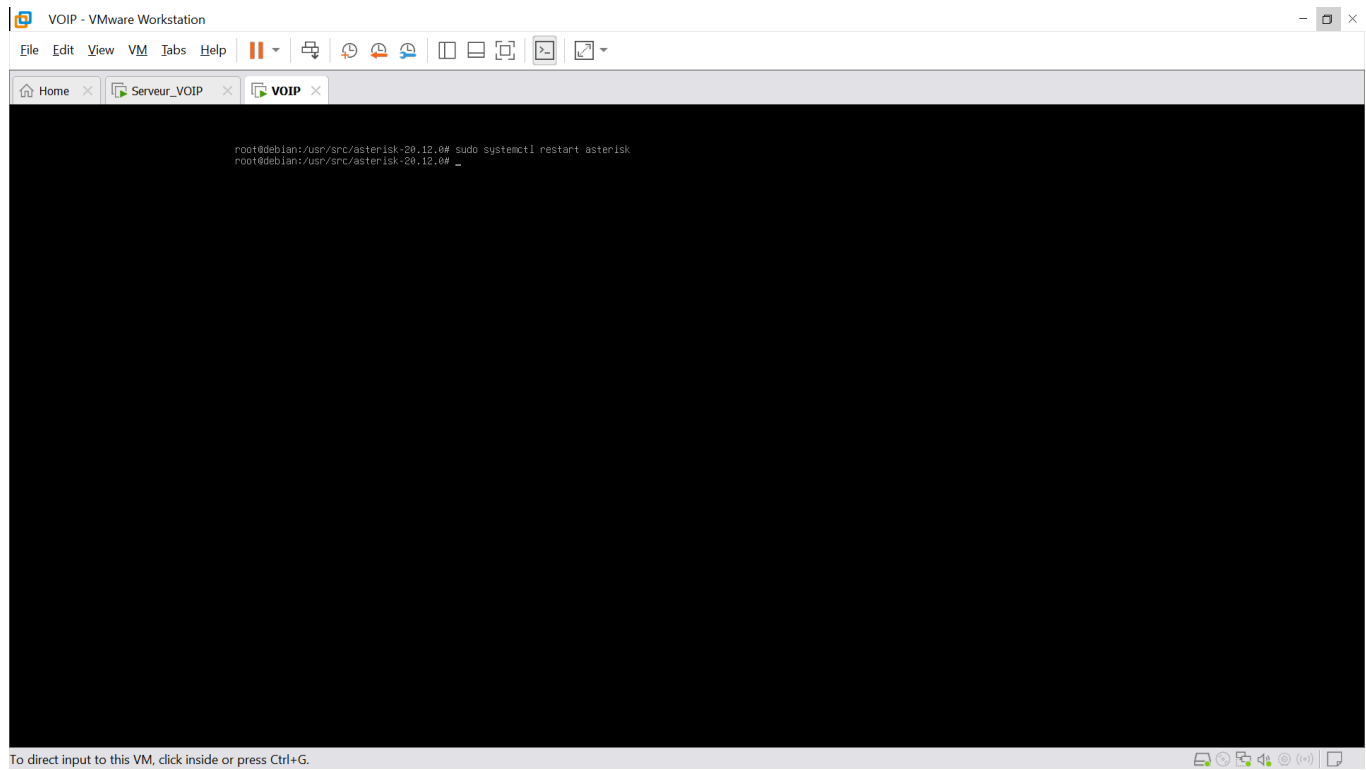


```
GNU nano 7.2 /etc/asterisk/extensions.conf
; default outbound endpoint to be configured in pjsip.conf.
;
;      Event: Test
;      X-Data: Over
;
;exten => 6001,1,noOp()
;      same => n,Answer()
;      same => n,PJSIPNotify(<slp:bob@127.0.0.1:5260>,&Event=Test&X-Data=Over)
;      same => n,Wait(1)
;      same => n,Hangup()
;
; Send a NOTIFY with the the custom headers defined in pjsip_notify.conf under
; 'custom-notify-1' inside the SIP dialog for the current channel.
;
;exten => 6002,1,noOp()
;      same => n,Answer()
;      same => n,PJSIPNotify(custom-notify-1)
;      same => n,Wait(1)
;      same => n,Hangup()
;
; Send a NOTIFY with the following headers and body to bob's custom uri. This
; requires a default outbound endpoint to be configured in pjsip.conf.
;
;      Event: Custom
;
;      Content-Type: application/voicemail
;      Content-Length: 14
;
;      check-messages
;
;exten => 6002,1,noOp()
;      same => n,Answer()
;      same => n,PJSIPNotify(&Event=Custom&Content-type=application/voicemail&Content=check-messages&Content=)
;      same => n,Wait(1)
;      same => n,Hangup()

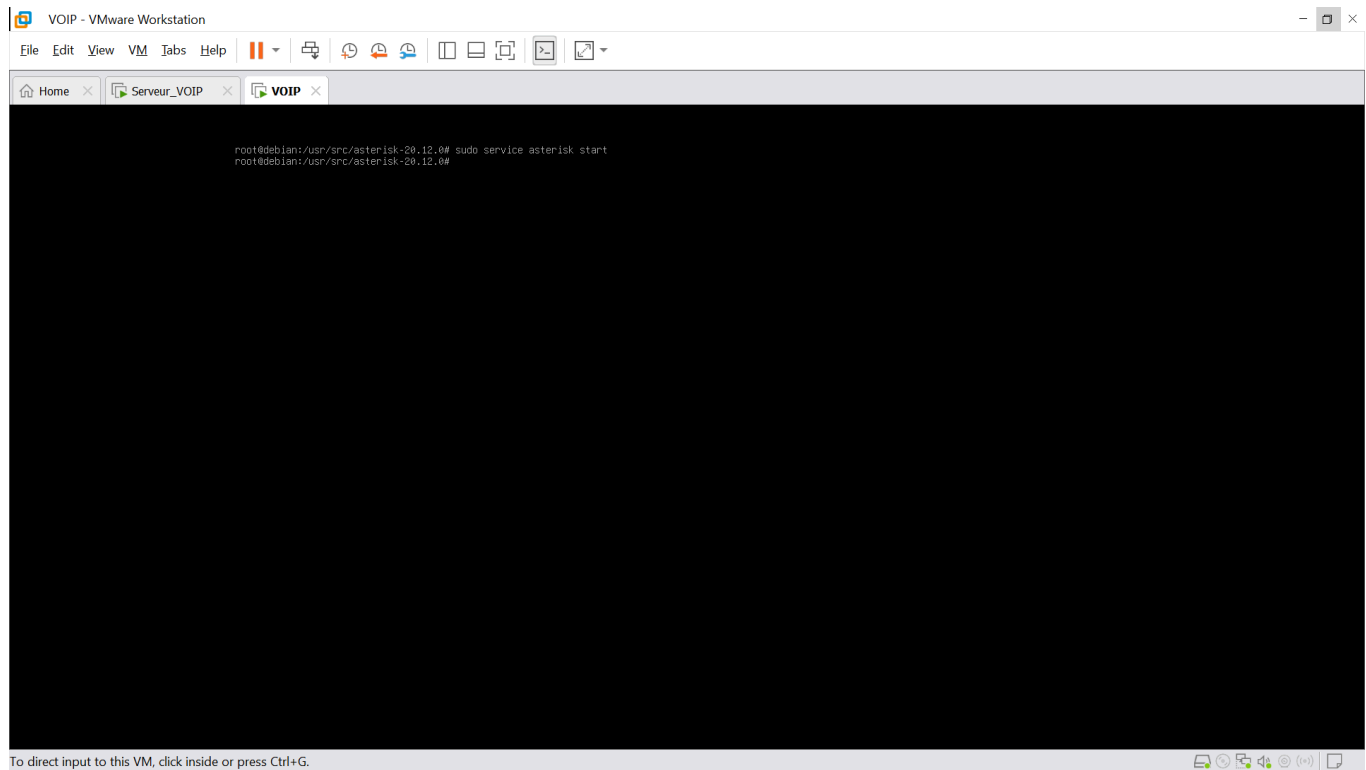
[internal]
exten => 6001,1,Answer()
same => n,Echo()
same => n,Hangup()

exten => 6001,1,Dial(PJSIP/6001)
exten => 6002,1,Dial(PJSIP/6002)
```

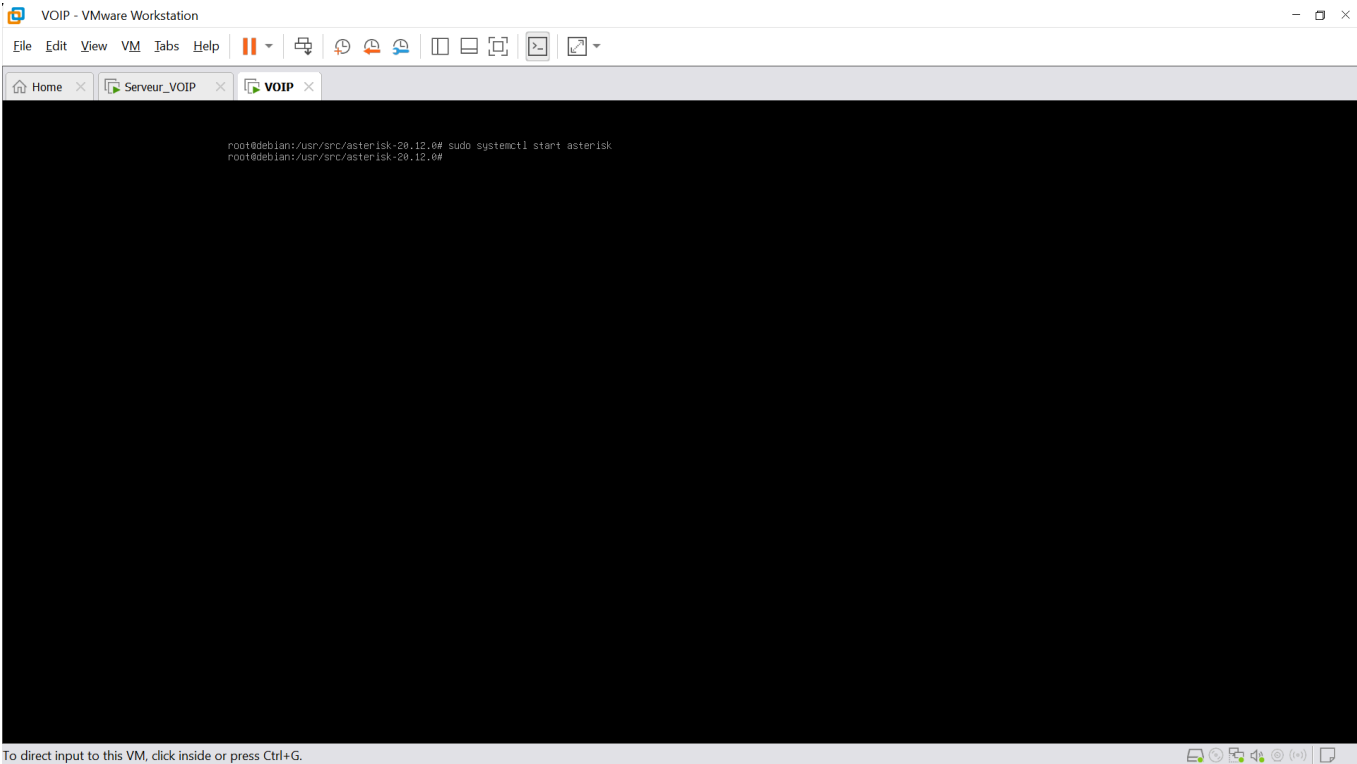
Etape 19 : Redémarrer Asterisk : sudo systemctl restart asterisk



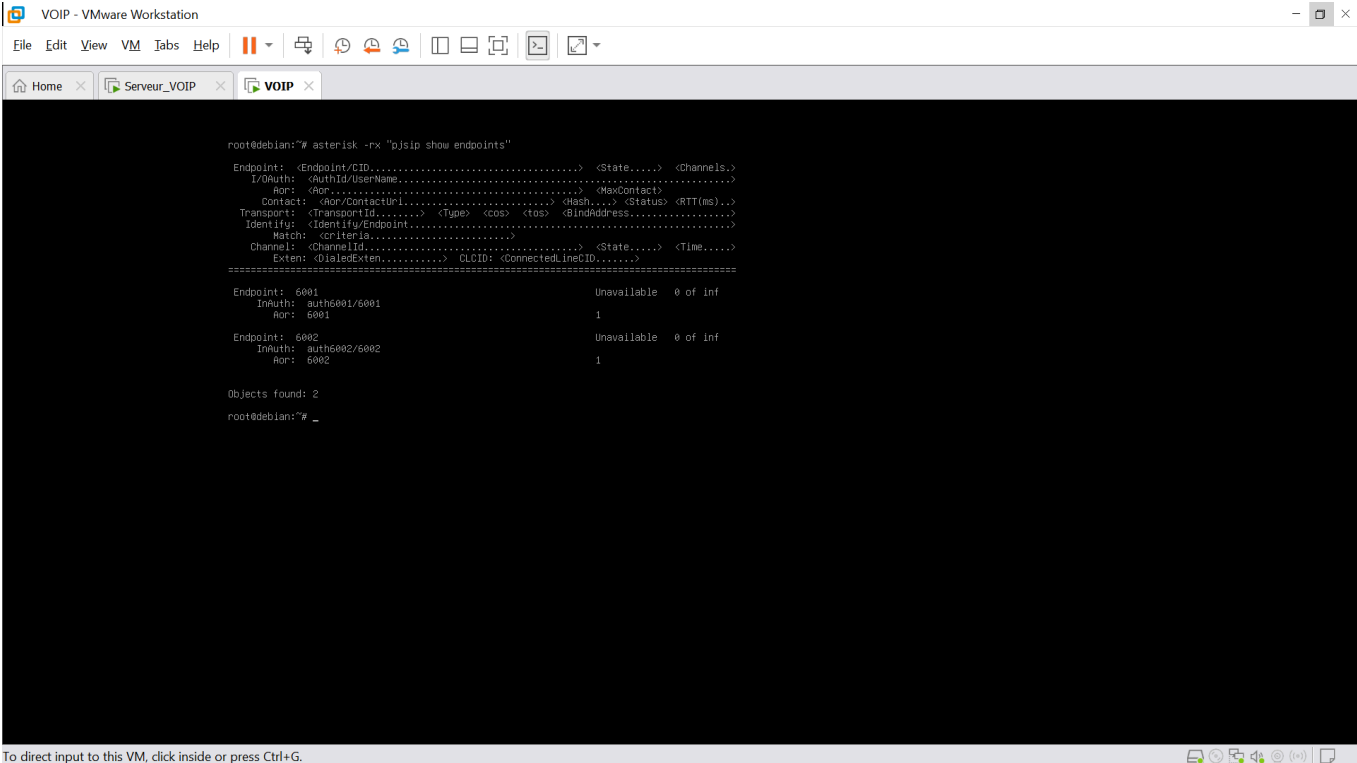
Etape 21 : Lancer le service asterisk : sudo service asterisk start



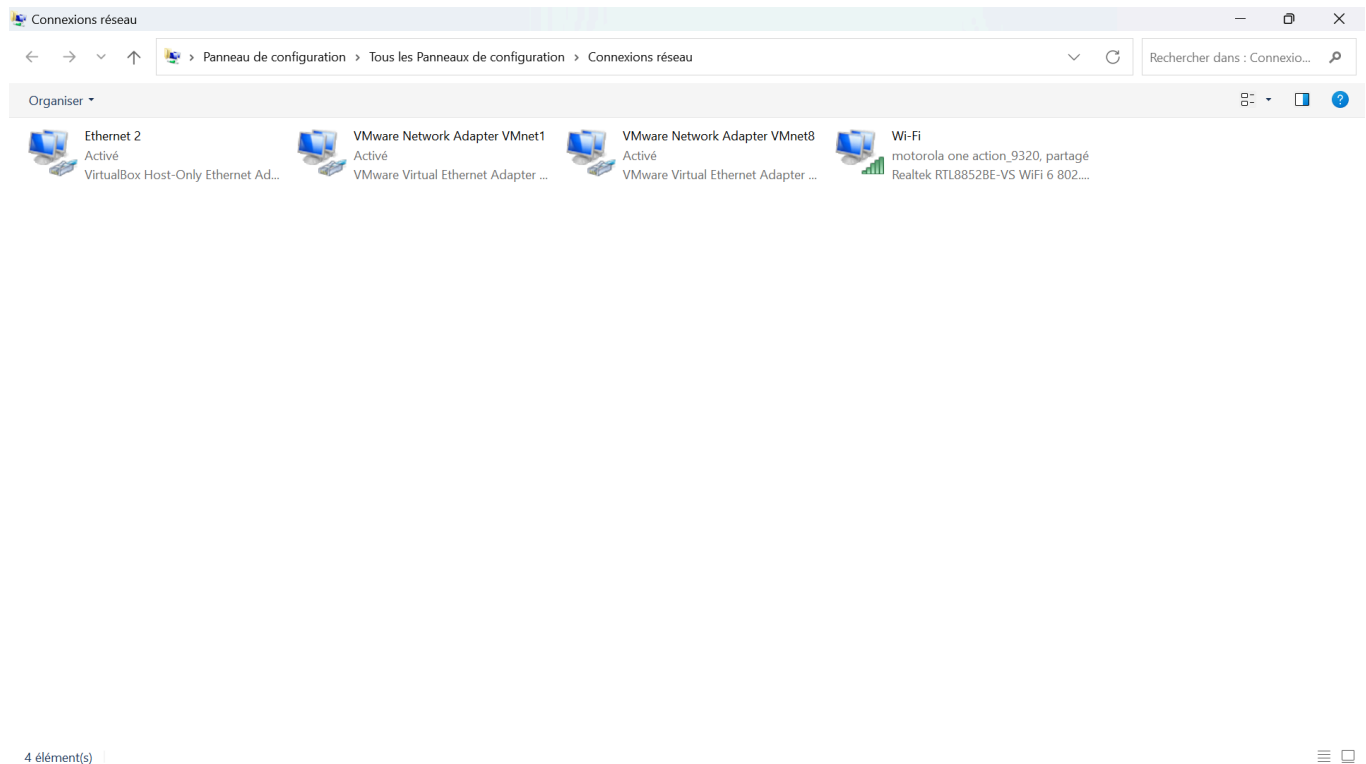
Etape 22 : Lancer le service asterisk : `sudo systemctl start asterisk`



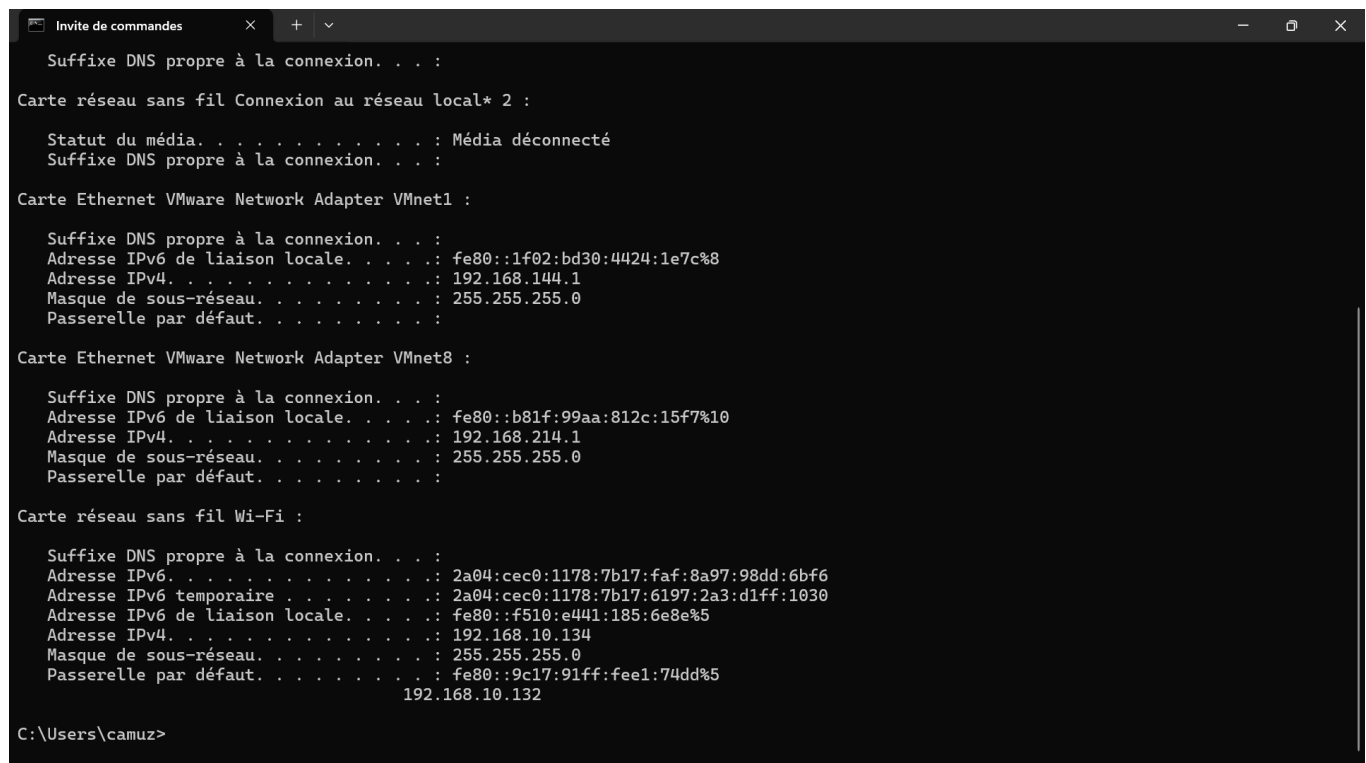
Etape 23 : Vérifier que les utilisateurs existent : asterisk -rx "pjsip show endpoints"



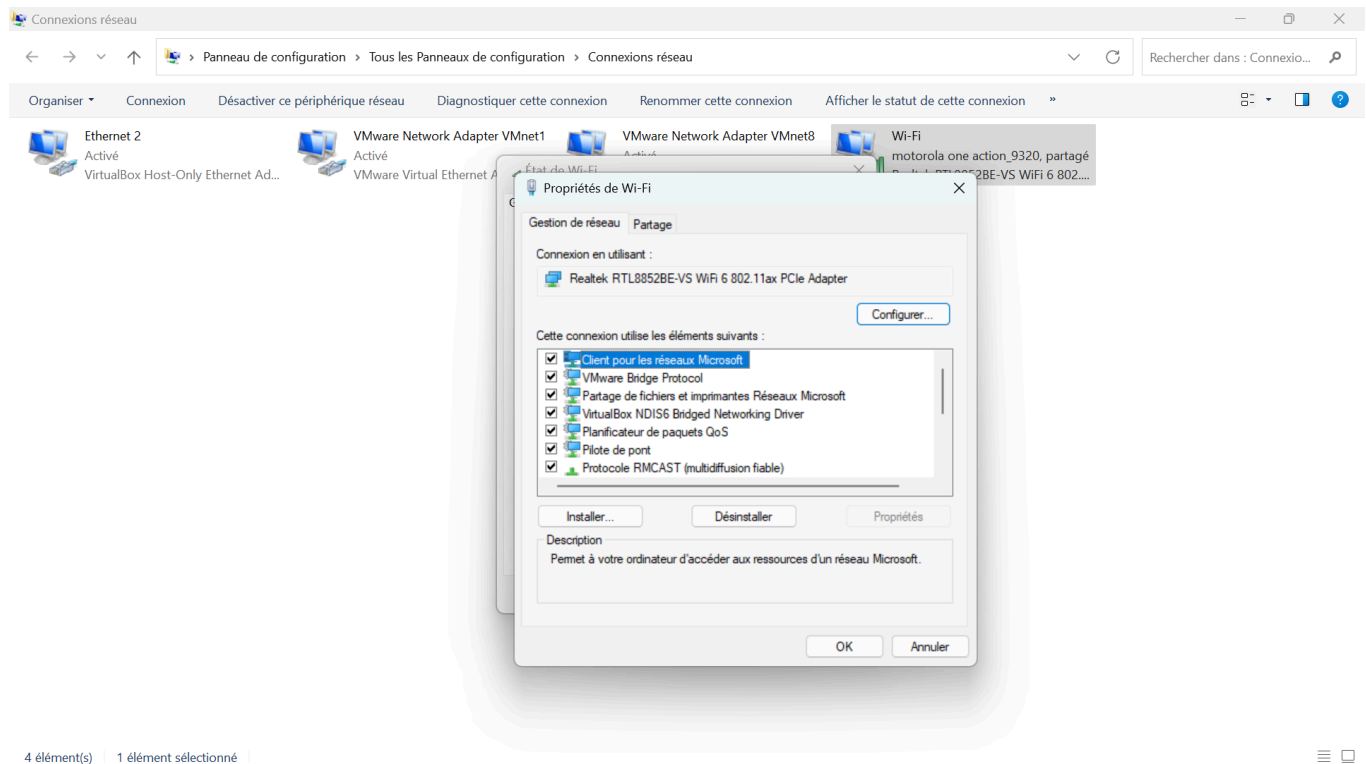
Etape 24 : Effectuer un partage de connexion depuis votre smartphone vers votre PC



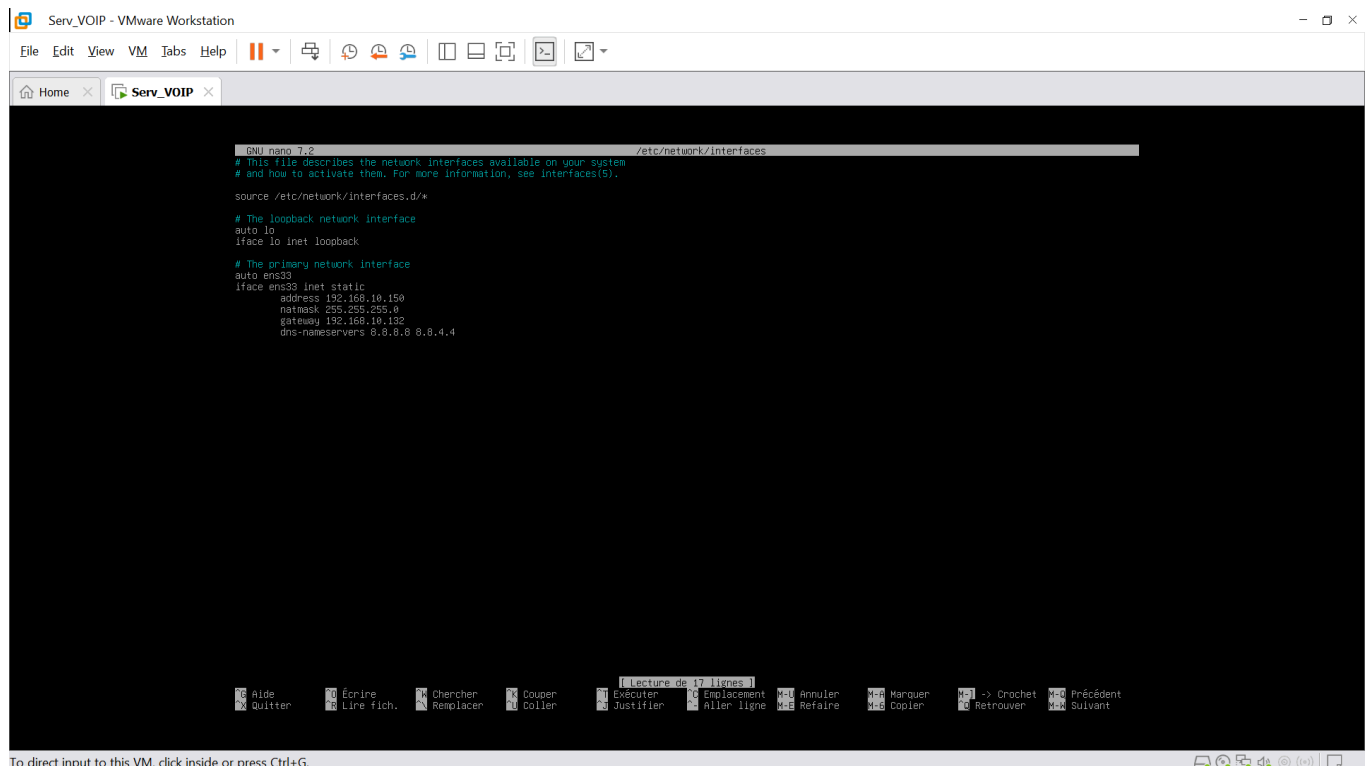
Etape 25 : Relever l'adresse IP attribuée à votre PC : ipconfig



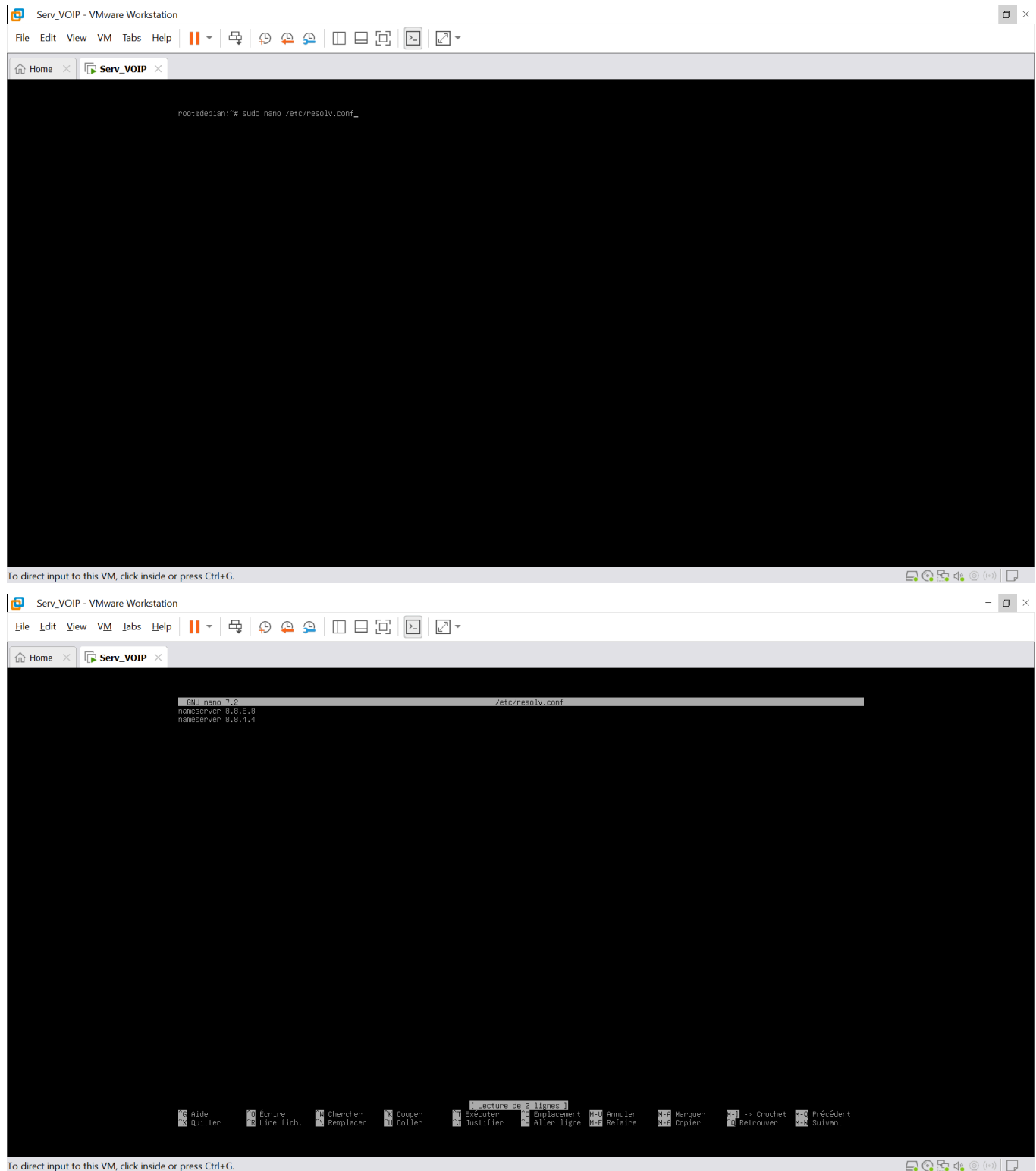
Etape 26 : Identifier la carte réseau utilisée par votre PC



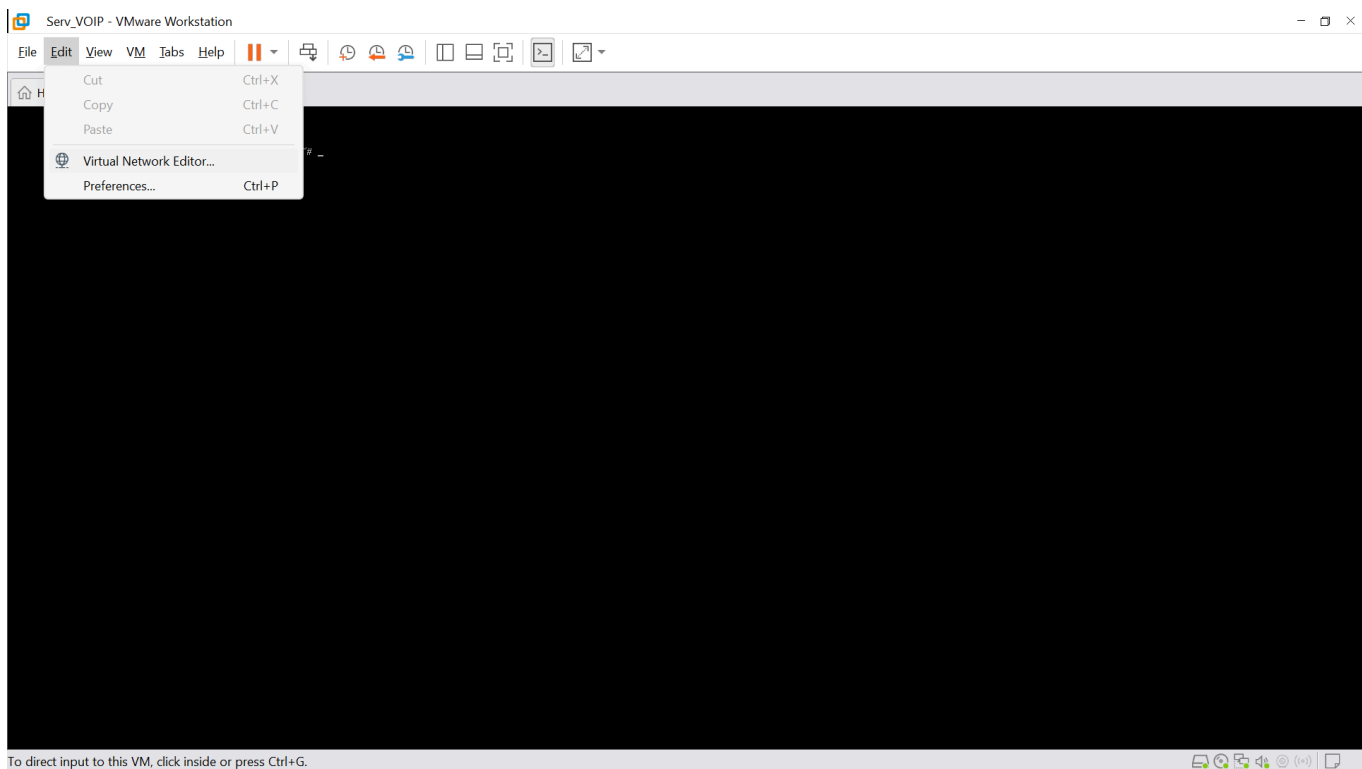
Etape 27: Attribuer une adresse IP statique à votre serveur (192.168.10.50) en tenant compte de l'adresse IPv4 de votre PC (192.168.144.1) ainsi que de la passerelle par défaut (192.168.10.134) : `sudo nano /etc/network/interfaces`



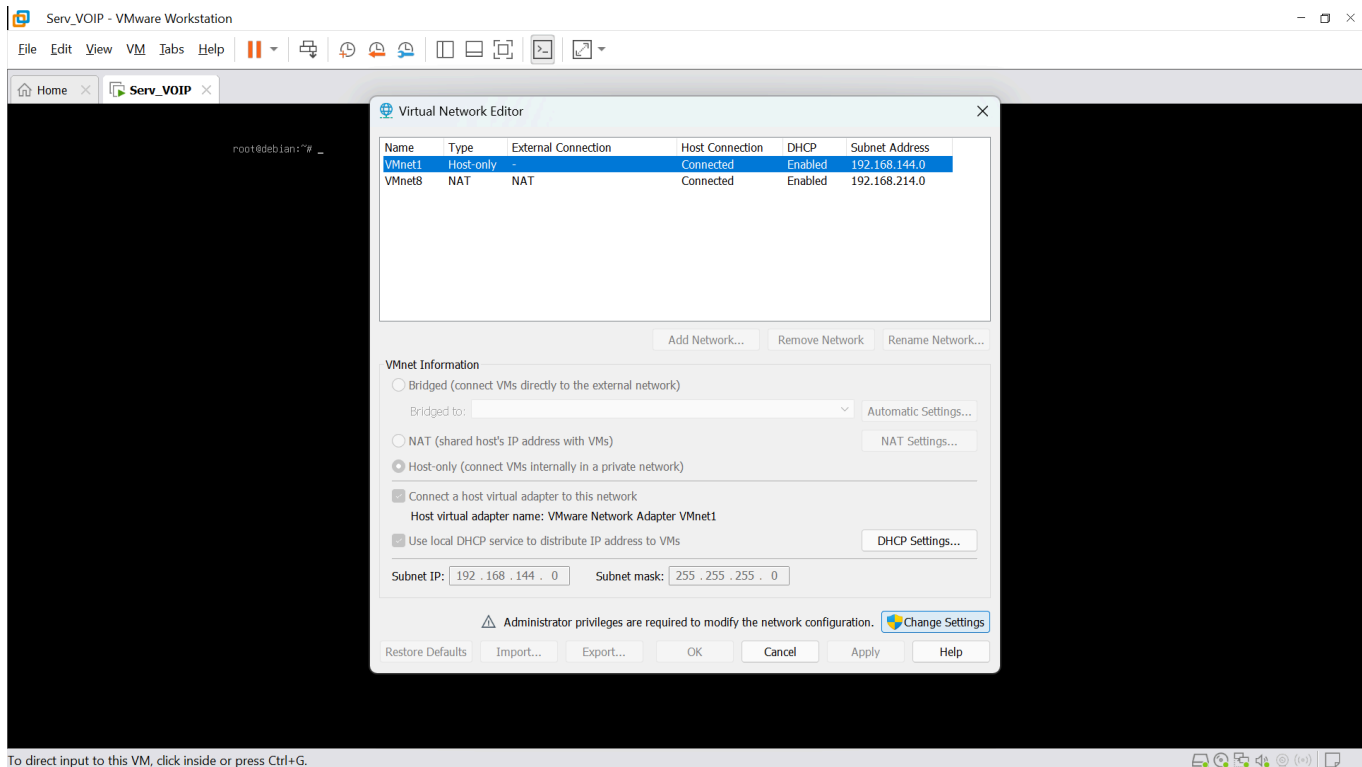
Etape 28 : Configurer votre fichier resolv.conf : /etc/resolv.conf



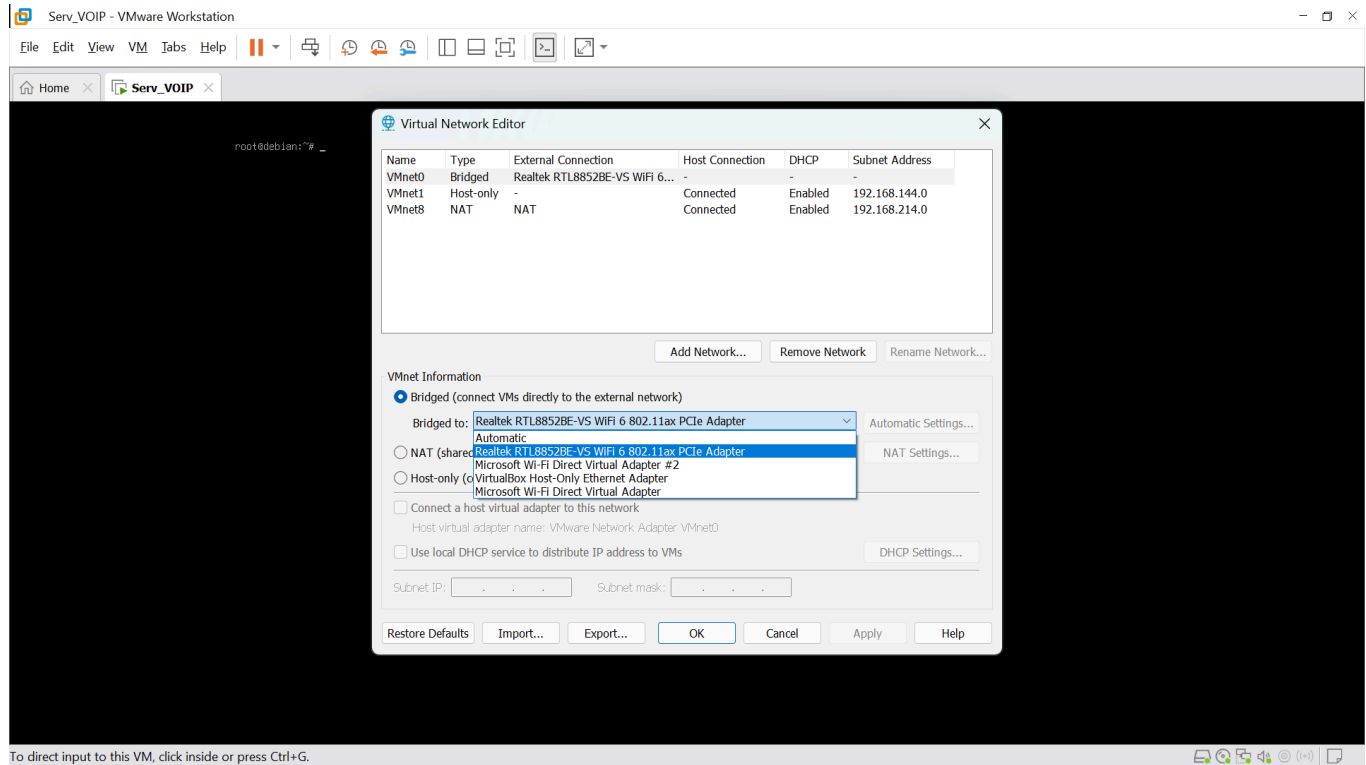
Etape 29 : Accéder au Virtual network editor



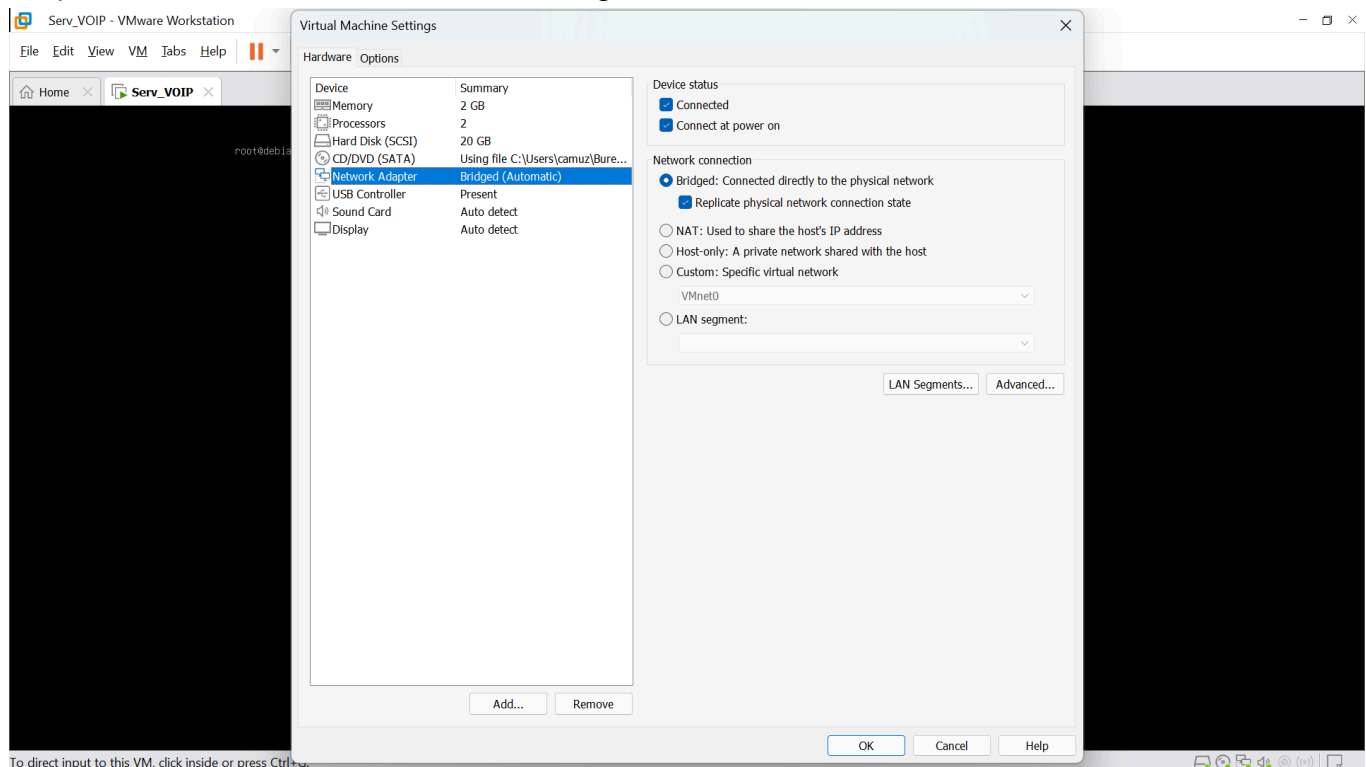
Etape 30 : Cliquer sur Change Settings



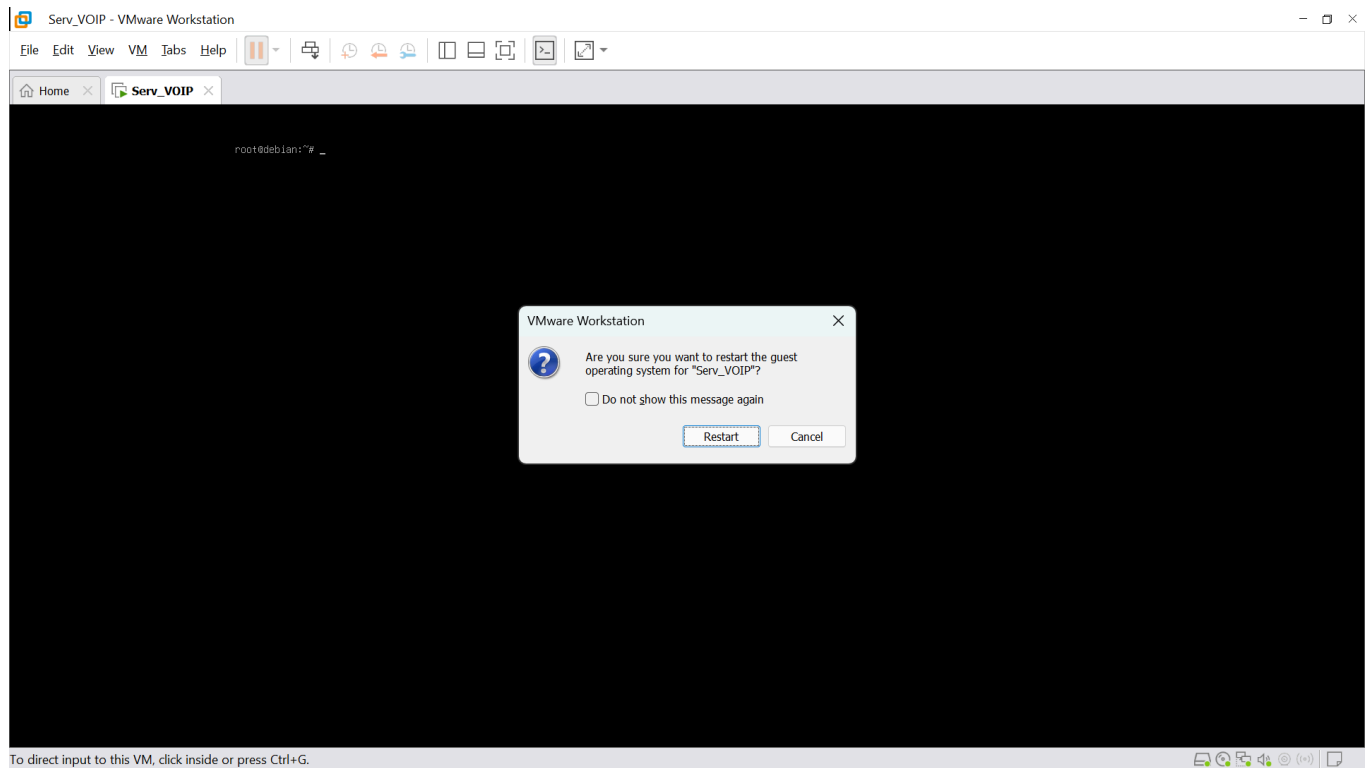
Etape31 : Configurer le mode bridge en sélectionnant la carte réseau identifiée lors de l'étape 26



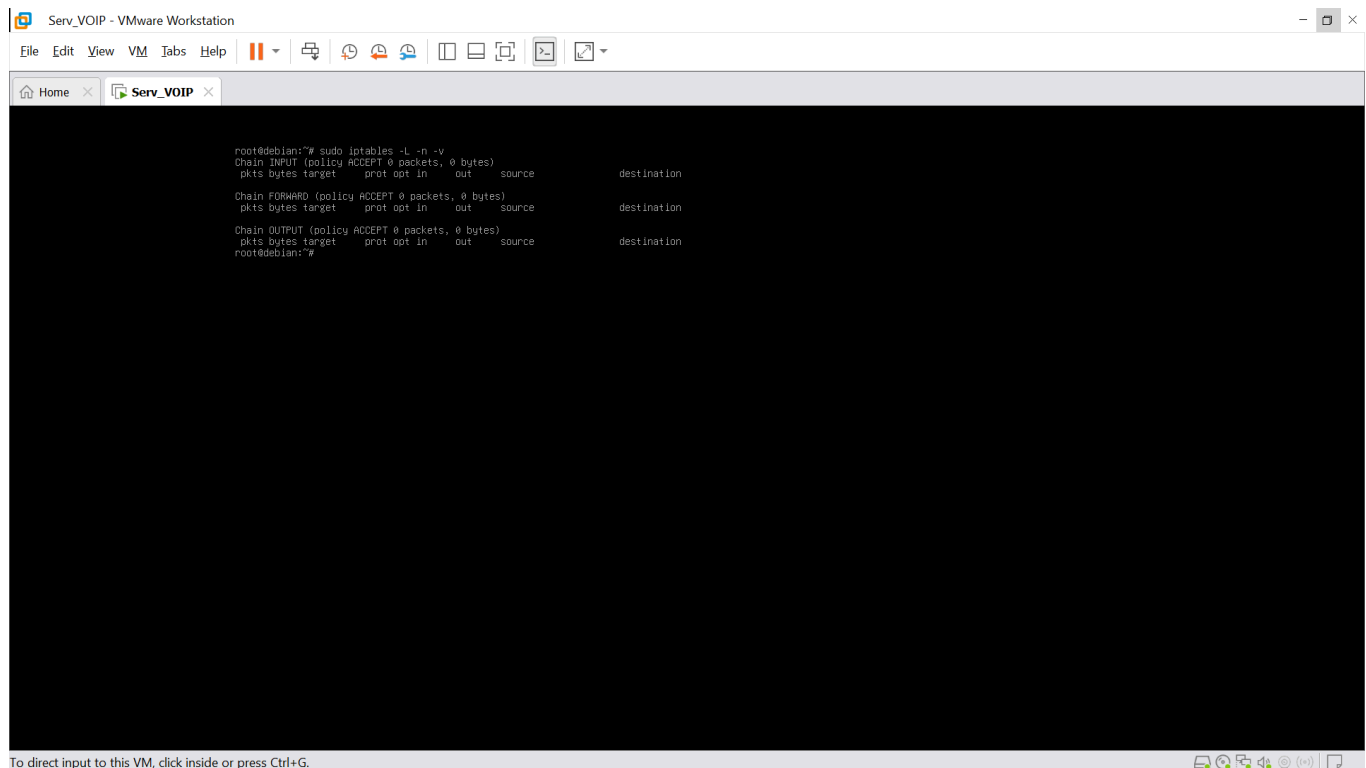
Etape 32 : Passer votre VM en mode bridge



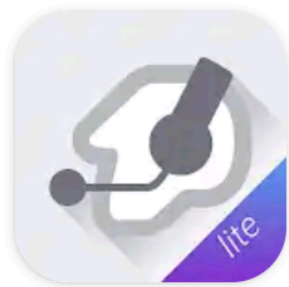
Etape 33 : Redémarrer votre VM



Etape 34 : Vérifier que vos équipements(PC, VM, Smartphone) appartiennent au même réseau : sudo iptables -L -n -v



Etape 35 : Télécharger Zoiper sur le Play Store



Zoiper IAX SIP VOIP Softphone

Securax EOOD

Achats via l'application

Désinstaller

Ouvrir

Nouveautés •

Mise à jour le 5 nov. 2024



v2.24.4

Fix extremely old purchases sync issue

Fix incorrect number shown after transferred on s...

Noter cette application

Donnez votre avis aux utilisateurs





[Rédiger un avis](#)

Assistance pour l'application



À propos de l'appli



Jeux



Applis



Rechercher



Livres



Etape 36 : Configurer un compte Zpoiler pour l'un de vos utilisateurs



Compte SIP

Nom du compte

6001@192.168.10.150

Authentication

Hôte

192.168.10.150

Nom d'utilisateur

6001

Mot de passe

Optionnel

Authentication de l'utilisateur

Proxy sortant

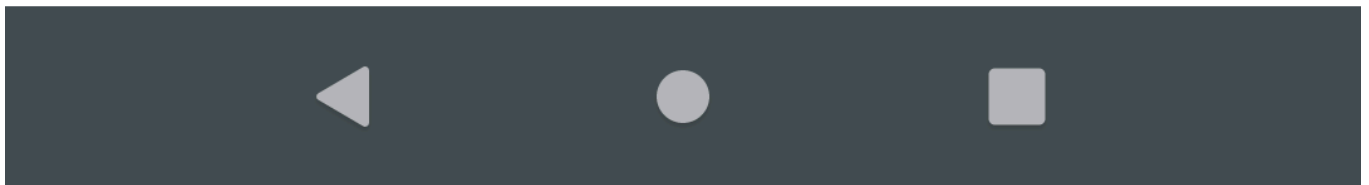
ID de l'appelant

Extension de messagerie vocale

Sonnerie

Activer la présence

Abonnez-vous pour la présence.



Etape 37: Vérifier que le compte soit bien connecté

Vous utilisez la version gratuite
de Zoiper



6001@192.16... ^

Le compte est prêt

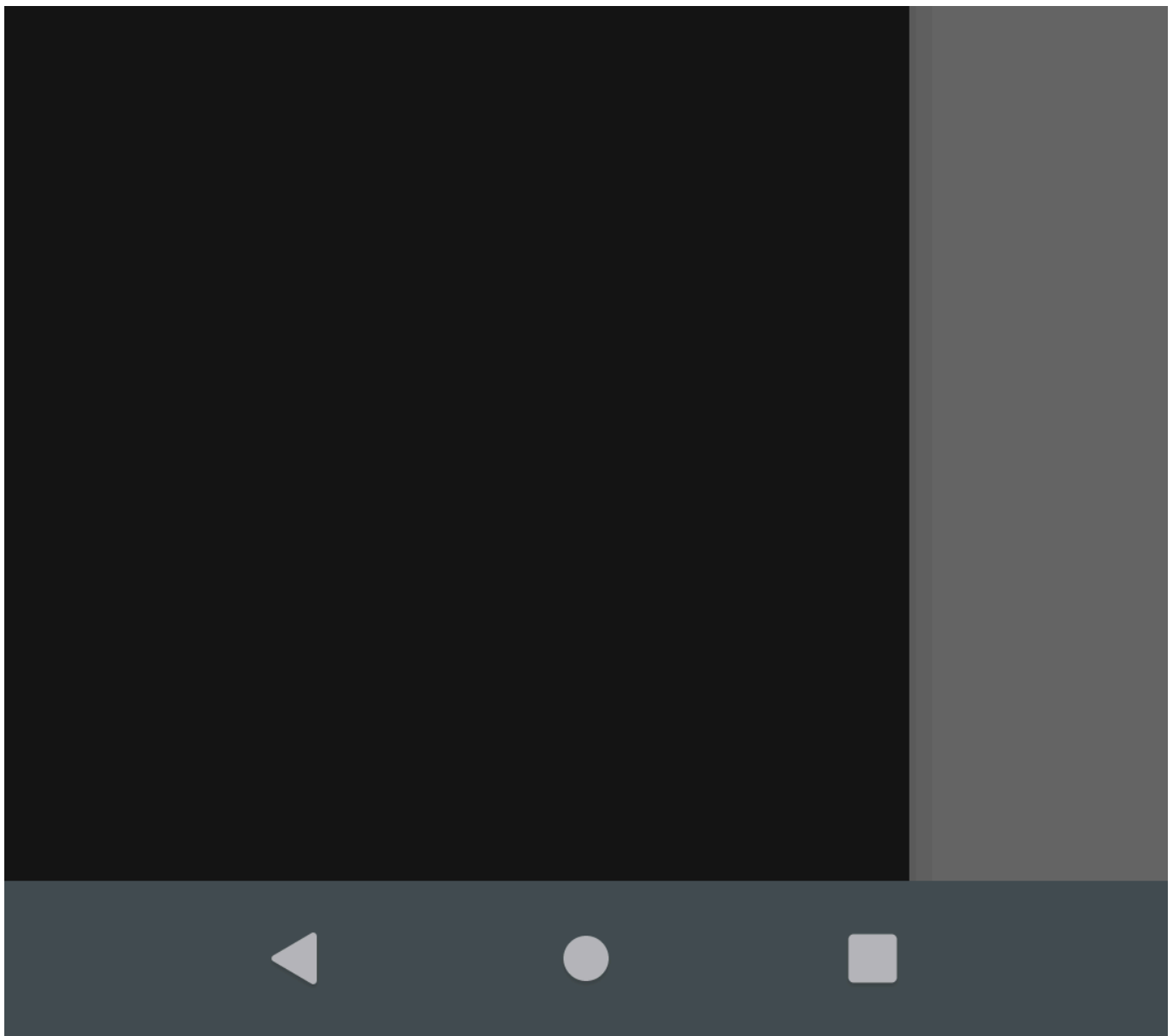


6001@192.168....



Ajouter un compte

activez



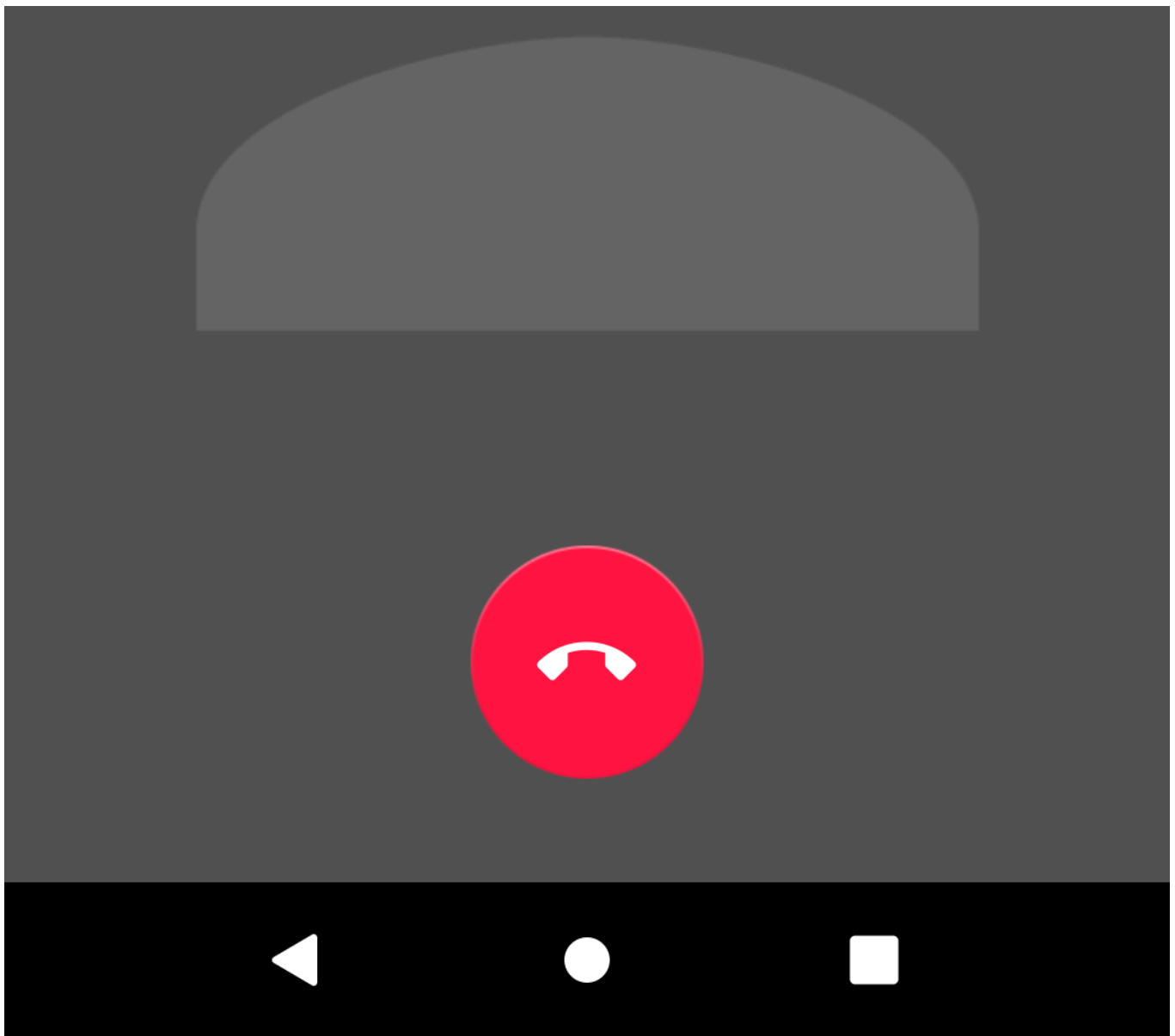
Etape 38 : Passer un appel depuis Zpoiler



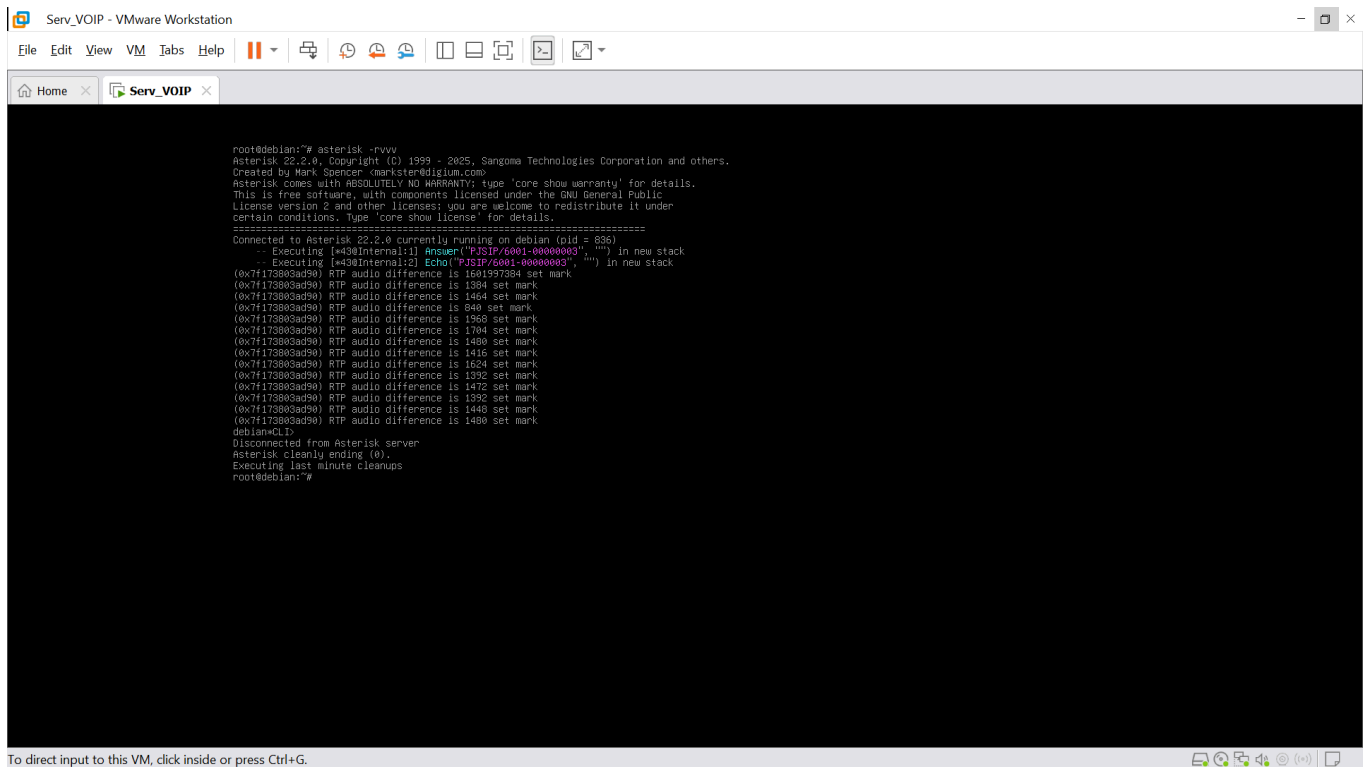
*43 (*43)

00:18

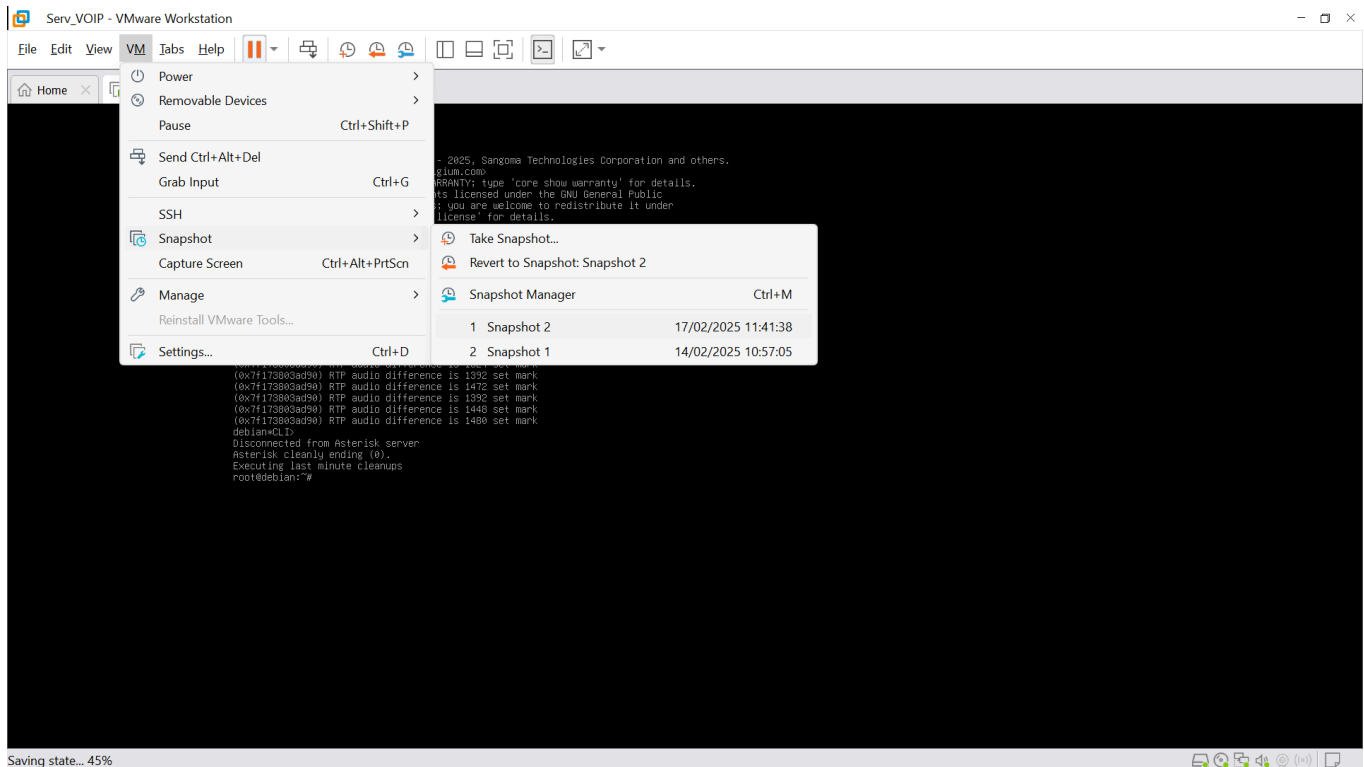




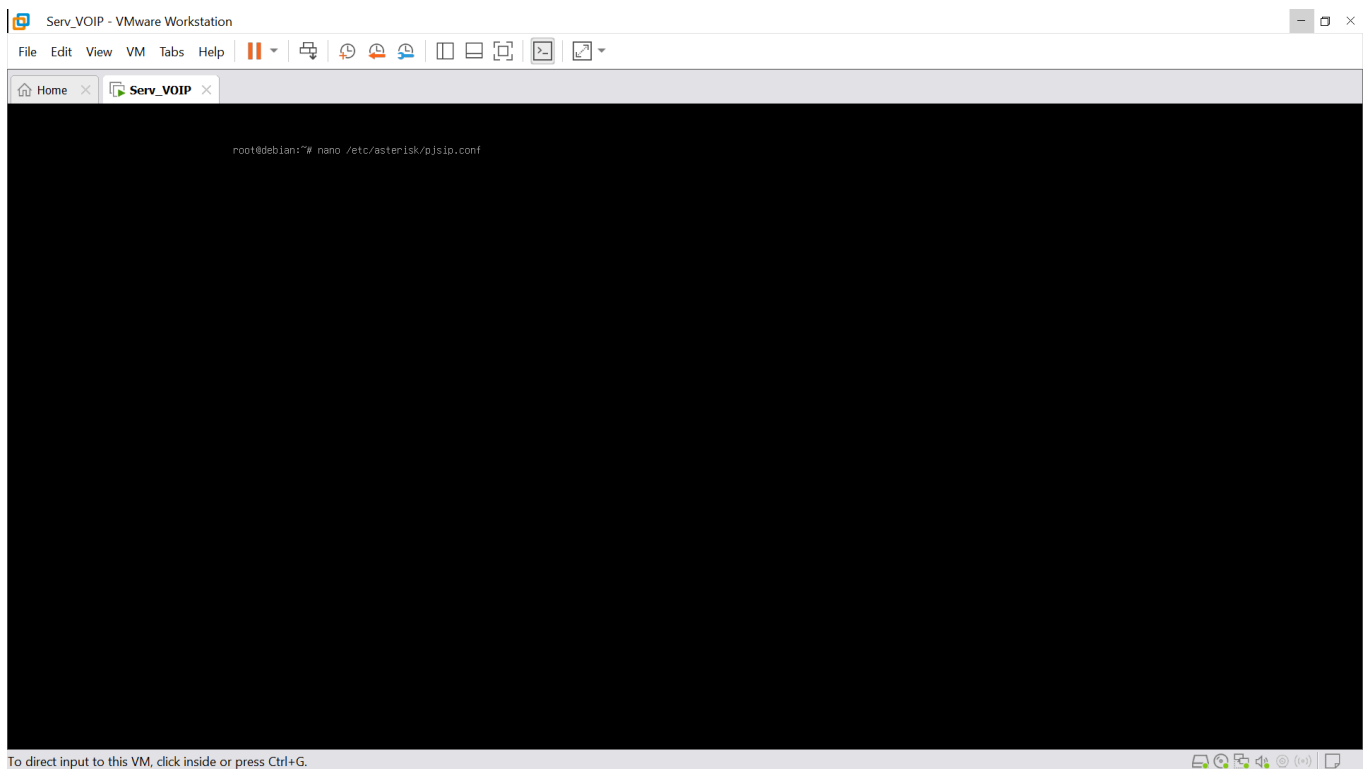
Etape 39 : Vérifier en direct sur le serveur que l'appel est bien passé



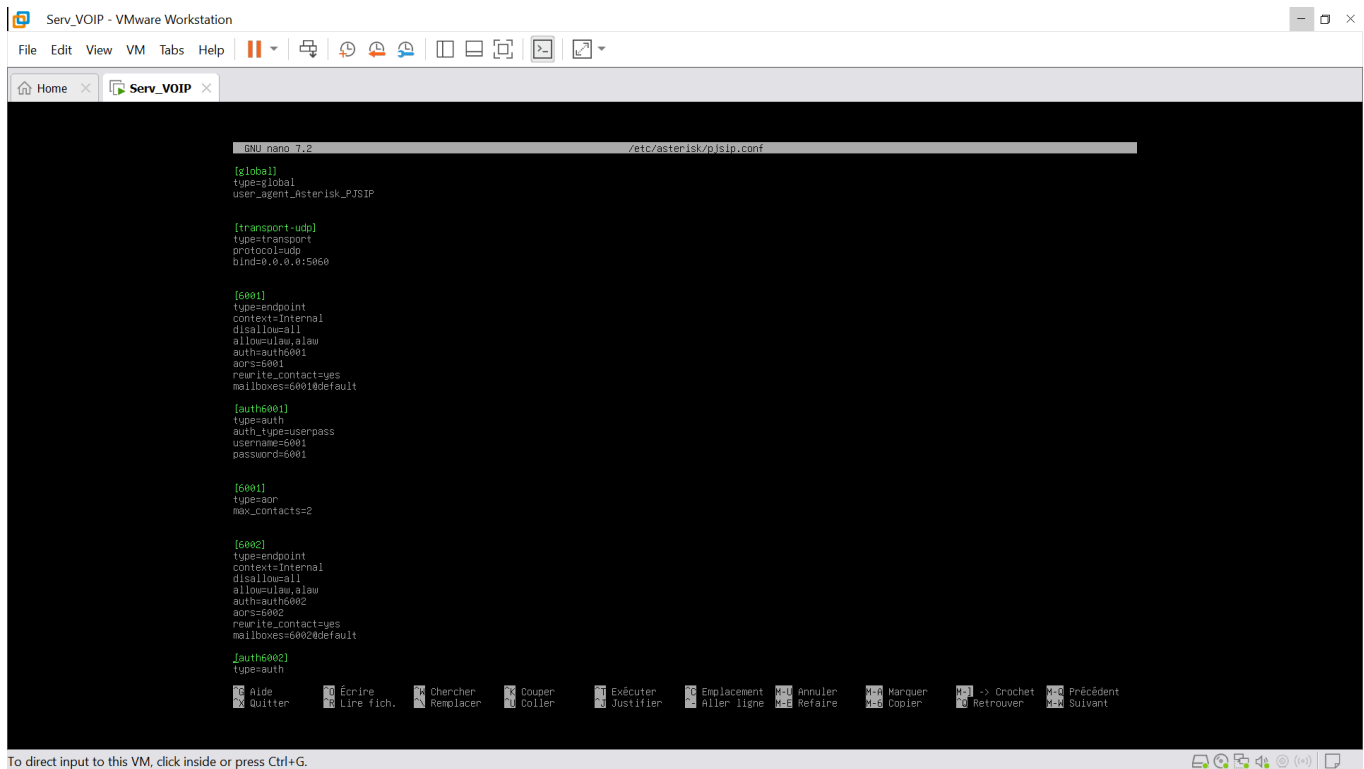
Etape 40 : Sauvegarder la VM (snapshot 2)



Etape 41 : Editer le fichier pjsip.conf : nano /etc/asterisk/pjsip.conf



Etape 42 : Configurer vos utilisateurs 6001 et 6002



Etape 43 : Configurer vos utilisateurs 8001 et 8002

Etape 44 : Vérifier la configuration de vos utilisateurs : asterisk -rx "pjsip show endpoints"

```
root@debian:~# asterisk -rx "pjsip show endpoints"

Endpoint: <Endpoint/CID.....> <State.....> <Channels..>
I/Chauth: <AuthID/UserName.....>
Aor: <Aor.....> <MaxContact>
Contact: <Aor/ContactUri.....> <Hash.....> <Status> <RTT(ms)..>
Transport: <TransportID.....> <Type> <cos> <tos> <BindAddress.....>
Identity: <IdentityEndpoint.....>
Match: <Criteria.....>
Channel: <ChannelID.....> <State.....> <Time.....>
Extent: <IsaledExten.....> CIDID: <ConnectedLineCID.....>
=====

Endpoint: 6001
InAuth: auth6001/6001
Aor: 6001
Contact: 6001/sip:6001@192.168.185.171:49748;rinsta 2fa89161ac NonQual
nan

Endpoint: 6002
InAuth: auth6002/6002
Aor: 6002
Unavallable
0 of Inf

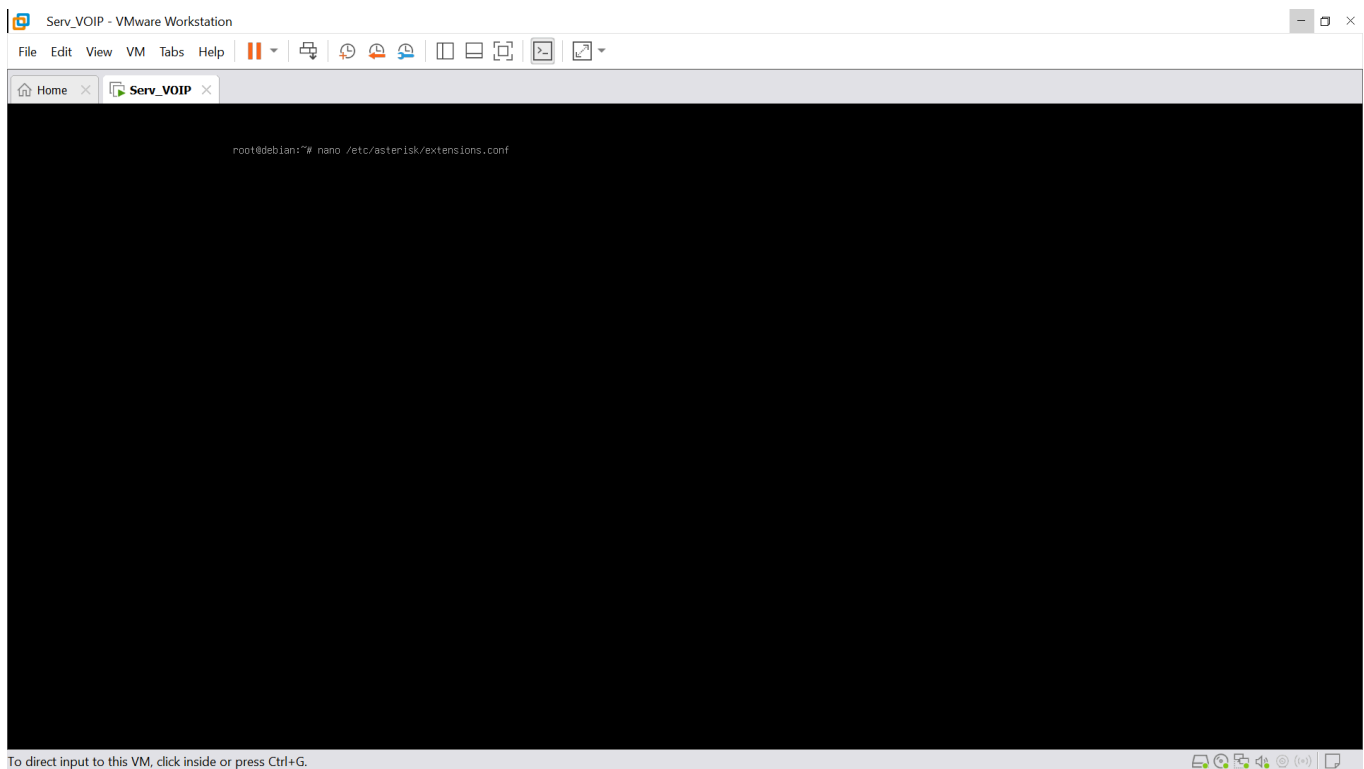
Endpoint: 6001
InAuth: auth6001/6001
Aor: 6001
Unavallable
0 of Inf

Endpoint: 6002
InAuth: auth6002/6002
Aor: 6002
Unavallable
0 of Inf

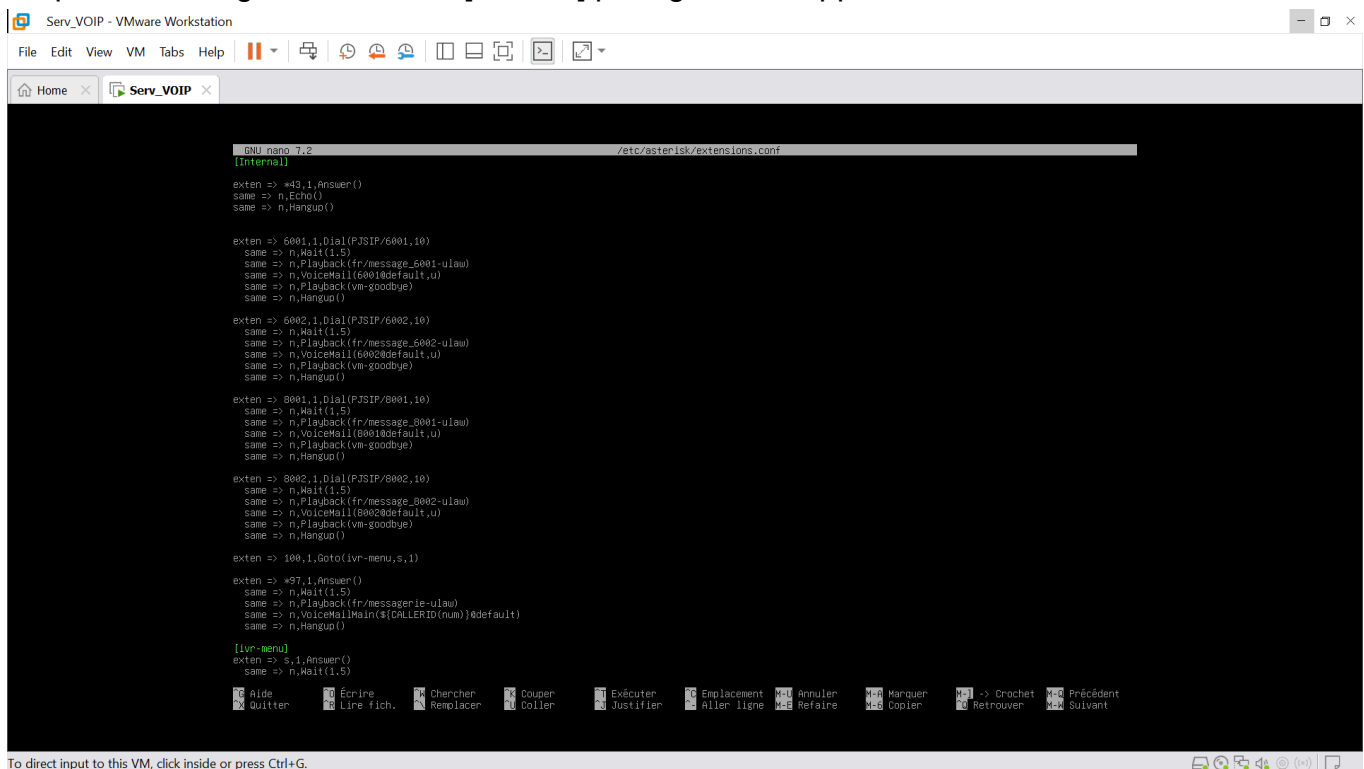
Objects found: 4

root@debian:~#
```

Etape 45 : Editer le fichier extensions.conf : nano /etc/asterisk/extensions.conf



Etape 46 : Configurer la section [internal] pour gérer les appels de vos utilisateurs en interne



Etape 47 : Vérifier la configuration de vos appels en interne : asterisk -rx "dialplan show Internal"

```
root@debian:/var/lib/asterisk/sounds/fr# asterisk -rx "dialplan show Internal"
[ Context: 'Internal' created by 'pbx_conf18' ]
'43' =>
1. Answer()
2. Echo()
3. Hangup()
'497' =>
1. Answer()
2. Wait(1.5)
3. Playback(fr/messagerie-ulaw)
4. VoiceMailMain(${CALLERID(num)}@default)
5. Hangup()
'100' =>
1. Goto(livr-menu,s,1)
'6001' =>
1. Dial(PJSIP/6001,10)
2. Wait(1.5)
3. Playback(fr/message_6001-ulaw)
4. VoiceMail(6001@default,u)
5. Playback(vm-goodbye)
6. Hangup()
'6002' =>
1. Dial(PJSIP/6002,10)
2. Wait(1.5)
3. Playback(fr/message_6002-ulaw)
4. VoiceMail(6002@default,u)
5. Playback(vm-goodbye)
6. Hangup()
'8001' =>
1. Dial(PJSIP/8001,10)
2. Wait(1.5)
3. Playback(fr/message_8001-ulaw)
4. VoiceMail(8001@default,u)
5. Playback(vm-goodbye)
6. Hangup()
'8002' =>
1. Dial(PJSIP/8002,10)
2. Wait(1.5)
3. Playback(fr/message_8002-ulaw)
4. VoiceMail(8002@default,u)
5. Playback(vm-goodbye)
6. Hangup()

== 7 extensions (33 priorities) in 1 context, ==
root@debian:/var/lib/asterisk/sounds/fr# _
```

Etape 48 : Configurer la section [ivr-menu] pour créer un menu vocal interactif

```
GNU nano 7.2 /etc/asterisk/extensions.conf
[ivr-menu]
exten => s,1,Answer()
same => n,Wait(1.5)
same => n,Playback(fr/bienvenue2-ulaw)
same => n,WaitExten(10)
same => n,ExecIf( "${EXTEN}" != "" )?Goto(${EXTEN},1)

exten => 1,1,Playback(fr/transfert_support-ulaw)
same => n,Dial(PJSIP/8001,20)
same => n,Wait(1.5)
same => n,Playback(fr/message_8001-ulaw)
same => n,VoiceMail(8001@default,u)
same => n,Playback(vm-goodbye)
same => n,Hangup()

exten => 2,1,Playback(fr/transfert_ventes-ulaw)
same => n,Dial(PJSIP/8002,20)
same => n,Wait(1.5)
same => n,Playback(fr/message_8002-ulaw)
same => n,VoiceMail(8002@default,u)
same => n,Playback(vm-goodbye)
same => n,Hangup()

exten => 3,1,Playback(fr/laissez_message-ulaw)
same => n,VoiceMail(1000@default,u)
same => n,Hangup()

exten => 1,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vir-menu,s,1)

exten => t,1,Playback(fr/aucune_saisie-ulaw)
same => n,Hangup()

exten => 1000,1,VoiceMailMain(1000@default)
same => n,Hangup()

exten => 4,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(livr-menu,s,1)

exten => 5,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(livr-menu,s,1)

exten => 6,1,Playback(fr/option_invalide-ulaw)
same=> n,Goto(livr-menu,s,1)

exten => 7,1,Playback(fr/option_invalide-ulaw)
```

```
GNU nano 7.2 /etc/asterisk/extensions.conf
same => n,VoiceMail(0001@default,u)
same => n,Playback(vm-goodbye)
same => n,Hangup()

exten => 2,1,Playback(fr/transfer_ventes-ulaw)
same => n,Dial(PJSIP/0002,20)
same => n,Wait(1,5)
same => n,Playback(fr/message_0002-ulaw)
same => n,VoiceMail(0002@default,u)
same => n,Playback(vm-goodbye)
same => n,Hangup()

exten => 3,1,Playback(fr/laissez_message-ulaw)
same => n,VoiceMail(1000@default,u)
same => n,Hangup()

exten => 4,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vlr-menu,s,1)

exten => t,1,Playback(fr/aucune_saisie-ulaw)
same => n,Hangup()

exten => 1000,1,VoiceMailMain(1000@default)
same => n,Hangup()

exten => 4,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vlr-menu,s,1)

exten => 5,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vlr-menu,s,1)

exten => 6,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vlr-menu,s,1)

exten => 7,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vlr-menu,s,1)

exten => 8,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vlr-menu,s,1)

exten => 9,1,Playback(fr/option_invalide-ulaw)
same => n,Goto(vlr-menu,s,1)

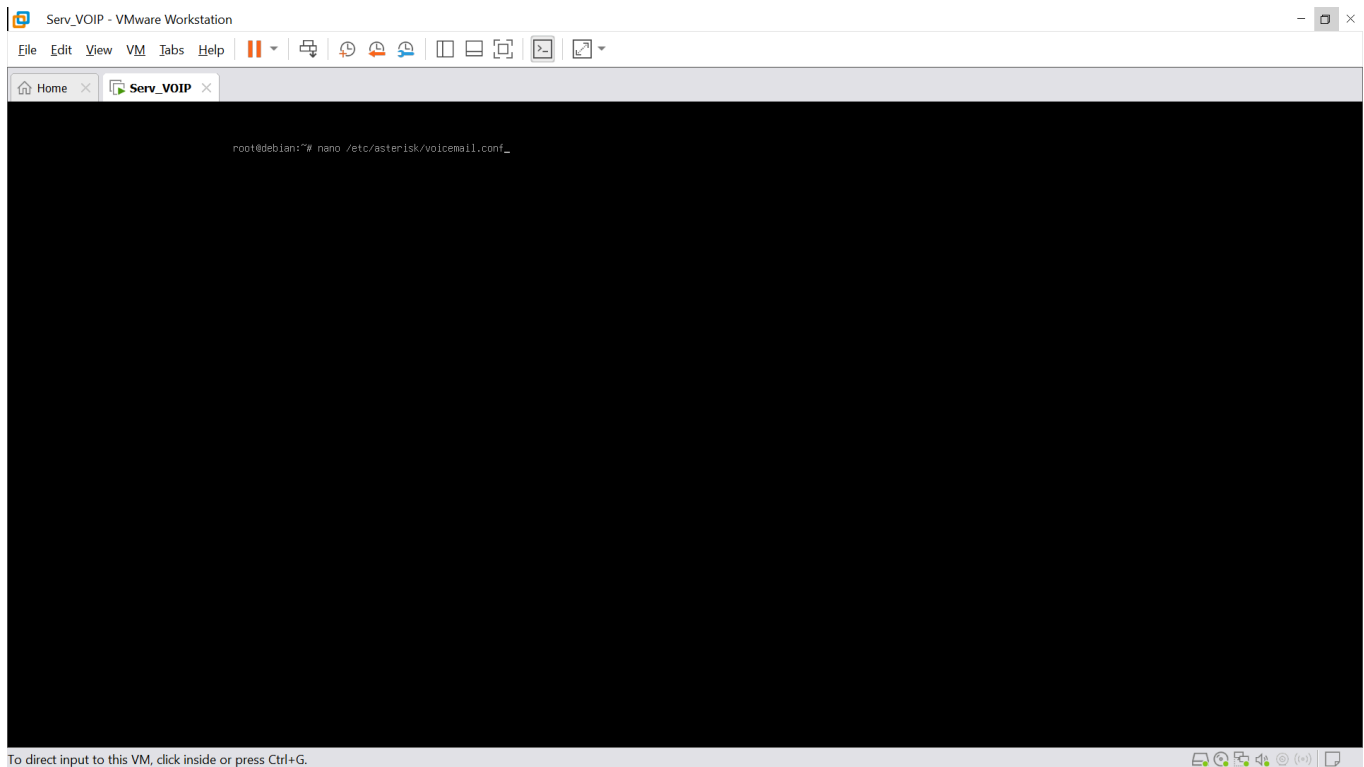
exten => *97,1,VoiceMailMain(${DALLERID(num)}@default)
same => n,Hangup()
-
```

Etape 49 : Vérifier la configuration de votre [ivr-menu] : asterisk -rx "dialplan show ivr-menu"

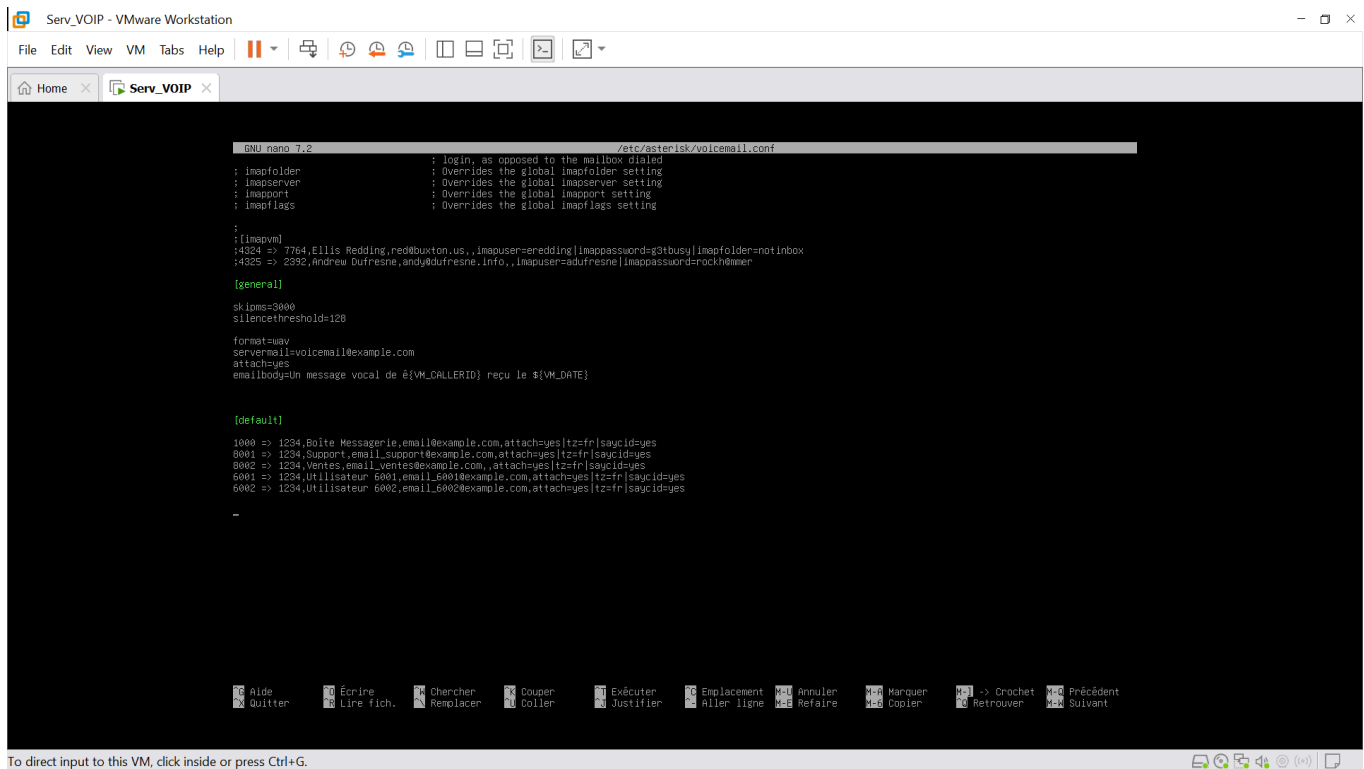
```
root@debian:/var/lib/asterisk/sounds/fr# asterisk -rx "dialplan show ivr-menu"
[ Context 'ivr-menu' created by 'pbx_config' ]
's97' => 1. VoiceMailMain(${DALLERID(num)}@default) [extensions.conf:979]
        2. Hangup() [extensions.conf:980]
'1' => 1. Playback(fr/transfer_support-ulaw) [extensions.conf:932]
        2. Dial(PJSIP/0001,20) [extensions.conf:933]
        3. Wait(1,5) [extensions.conf:934]
        4. Playback(fr/message_0001-ulaw) [extensions.conf:935]
        5. VoiceMail(0001@default,u) [extensions.conf:936]
        6. Playback(vm-goodbye) [extensions.conf:937]
        7. Hangup() [extensions.conf:938]
'1000' => 1. VoiceMailMain(1000@default) [extensions.conf:958]
        2. Hangup() [extensions.conf:959]
'2' => 1. Playback(fr/transfer_ventes-ulaw) [extensions.conf:940]
        2. Dial(PJSIP/0002,20) [extensions.conf:941]
        3. Wait(1,5) [extensions.conf:942]
        4. Playback(fr/message_0002-ulaw) [extensions.conf:943]
        5. VoiceMail(0002@default,u) [extensions.conf:944]
        6. Playback(vm-goodbye) [extensions.conf:945]
        7. Hangup() [extensions.conf:946]
'9' => 1. Playback(fr/laissez_message-ulaw) [extensions.conf:948]
        2. VoiceMail(1000@default,u) [extensions.conf:949]
        3. Hangup() [extensions.conf:950]
'4' => 1. Playback(fr/option_invalide-ulaw) [extensions.conf:961]
        2. Goto(vlr-menu,s,1) [extensions.conf:962]
'5' => 1. Playback(fr/option_invalide-ulaw) [extensions.conf:964]
        2. Goto(vlr-menu,s,1) [extensions.conf:965]
'6' => 1. Playback(fr/option_invalide-ulaw) [extensions.conf:967]
        2. Goto(vlr-menu,s,1) [extensions.conf:968]
'7' => 1. Playback(fr/option_invalide-ulaw) [extensions.conf:970]
        2. Goto(vlr-menu,s,1) [extensions.conf:971]
'8' => 1. Playback(fr/option_invalide-ulaw) [extensions.conf:972]
        2. Goto(vlr-menu,s,1) [extensions.conf:974]
'9' => 1. Playback(fr/option_invalide-ulaw) [extensions.conf:976]
        2. Goto(vlr-menu,s,1) [extensions.conf:977]
'1' => 1. () [extensions.conf:952]
        2. Goto(vlr-menu,s,1) [extensions.conf:953]
's' => 1. Answer() [extensions.conf:926]
        2. Wait(1,5) [extensions.conf:927]
        3. Playback(fr/bienvenue2-ulaw) [extensions.conf:928]
        4. WaitExten(10) [extensions.conf:929]
        5. ExecIf($[${EXTEN}] != '') ?Goto(${EXTEN},1) [extensions.conf:930]
't' => 1. Playback(fr/aucune_saisie-ulaw) [extensions.conf:955]
        2. Hangup() [extensions.conf:956]

-= 14 extensions (42 priorities) in 1 context. -=
root@debian:/var/lib/asterisk/sounds/fr#
```

Etape 50 : Editer le fichier voicemail.conf : nano /etc/asterisk/voicemail.conf



Etape 51 : Configurer la section [general] et la section [default]



Etape 52 : Verifier la configuration de votre fichier voicemail.conf : asterisk -rx "show users for default"

```
root@debian:/var/lib/asterisk/sounds/fr# asterisk -rx "voicemail show users for default"
Context:  hbox User:                               Zone:      NewMsg
default:  1234 Example Mailbox                      0
default:  1000 Boite Messagerie                      3
default:  8001 Support                               0
default:  8002 Ventes                                0
4 voicemail users configured.
root@debian:/var/lib/asterisk/sounds/fr# _
```

Etape 53 : Installer sox pour éditer des fichiers audio

```
root@debian:~# sudo apt install sox
```

Etape 54 : Dans le dossier `/var/lib/asterisk/sounds/fr/` éditer vos fichiers audio avant de les convertir en ulaw-wav

```

Serv_VOIP - VMware Workstation
File Edit View VM Tabs Help
Home Serv_VOIP

root@debian:~# pico2wave -l fr-FR -w /var/lib/asterisk/sounds/fr/bienvenue.wav "Bienvenue."
root@debian:~# pico2wave -l fr-FR -w /var/lib/asterisk/sounds/fr/transfert_support.wav "Transfert vers le support."
root@debian:~# pico2wave -l fr-FR -w /var/lib/asterisk/sounds/fr/transfert_ventes.wav "Transfert vers le service des ventes."
root@debian:~# pico2wave -l fr-FR -w /var/lib/asterisk/sounds/fr/laissez_message.wav "Veuillez laisser un message après le bip."
root@debian:~# pico2wave -l fr-FR -w /var/lib/asterisk/sounds/fr/option_invalide.wav "Option invalide. Veuillez réessayer."
root@debian:~# pico2wave -l fr-FR -w /var/lib/asterisk/sounds/fr/aucune_saisie.wav "Aucune saisie détectée. Au revoir."
root@debian:~# ls -l /var/lib/asterisk/sounds/fr
total 464
-rw-r--r-- 1 root root 86348 18 févr. 15:45 aucune_saisie.wav
-rw-r--r-- 1 root root 37548 18 févr. 15:39 bienvenue.wav
-rw-r--r-- 1 root root 78380 18 févr. 15:43 laissez_message.wav
-rw-r--r-- 1 root root 98732 18 févr. 15:44 option_invalide.wav
-rw-r--r-- 1 root root 67884 18 févr. 15:40 transfert_support.wav
-rw-r--r-- 1 root root 76460 18 févr. 15:41 transfert_ventes.wav
root@debian:~# sox /var/lib/asterisk/sounds/fr/bienvenue.wav -r 8000 -c 1 -b 16 /var/lib/asterisk/sounds/fr/bienvenue-ulaw.wav
root@debian:~# sox /var/lib/asterisk/sounds/fr/transfert_support.wav -r 8000 -c 1 -b 16 /var/lib/asterisk/sounds/fr/transfert_support-ulaw.wav
root@debian:~# sox /var/lib/asterisk/sounds/fr/transfert_ventes.wav -r 8000 -c 1 -b 16 /var/lib/asterisk/sounds/fr/transfert_ventes-ulaw.wav
sox FAIL formats: can't open output file /var/lib/asterisk/sounds/fr/transfert_ventes-ulaw.wav: No such file or directory
root@debian:~# sox /var/lib/asterisk/sounds/fr/transfert_ventes.wav -r 8000 -c 1 -b 16 /var/lib/asterisk/sounds/fr/transfert_ventes-ulaw.wav
root@debian:~# sox /var/lib/asterisk/sounds/fr/laissez_message.wav -r 8000 -c 1 -b 16 /var/lib/asterisk/sounds/fr/laissez_message-ulaw.wav
root@debian:~# sox /var/lib/asterisk/sounds/fr/option_invalide.wav -r 8000 -c 1 -b 16 /var/lib/asterisk/sounds/fr/option_invalide-ulaw.wav
root@debian:~# sox /var/lib/asterisk/sounds/fr/aucune_saisie.wav -r 8000 -c 1 -b 16 /var/lib/asterisk/sounds/fr/aucune_saisie-ulaw.wav
root@debian:~# ls -l /var/lib/asterisk/sounds/fr
total 704
-rw-r--r-- 1 root root 49136 18 févr. 15:58 aucune_saisie-ulaw.wav
-rw-r--r-- 1 root root 86348 18 févr. 15:45 aucune_saisie.wav
-rw-r--r-- 1 root root 16736 18 févr. 15:47 bienvenue-ulaw.wav
-rw-r--r-- 1 root root 37548 18 févr. 15:39 bienvenue.wav
-rw-r--r-- 1 root root 39212 18 févr. 15:55 laissez_message-ulaw.wav
-rw-r--r-- 1 root root 70380 18 févr. 15:43 laissez_message.wav
-rw-r--r-- 1 root root 49380 18 févr. 15:56 option_invalide-ulaw.wav
-rw-r--r-- 1 root root 98732 18 févr. 15:44 option_invalide.wav
-rw-r--r-- 1 root root 33354 18 févr. 15:43 transfert_support-ulaw.wav
-rw-r--r-- 1 root root 67884 18 févr. 15:40 transfert_support.wav
-rw-r--r-- 1 root root 38252 18 févr. 15:53 transfert_ventes-ulaw.wav
-rw-r--r-- 1 root root 76460 18 févr. 15:41 transfert_ventes.wav
root@debian:~# _
```

Etape 55 : Se connecter à votre console asterisk : asterisk -rvvv

```

Serv_VOIP - VMware Workstation
File Edit View VM Tabs Help
Home Serv_VOIP

root@debian:/var/lib/asterisk/sounds/fr# asterisk -rvvv
Asterisk 22.2.0, Copyright (C) 1999 - 2025, Sangoma Technologies Corporation and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 22.2.0 currently running on debian (pid = 859)
debian@CLI> _
```

Etape 56 : Tester un appel de 6001 à 6002 et vérifier en direct sur votre console asterisk : taper le numéro 6002 sur votre interface Zoiper

Etape 57 : Tester un appel de 6002 à 6001 et vérifier en direct sur votre console asterisk : taper le numéro 6001 sur votre interface Zoiper

Etape 58 : Tester un appel vers votre menu vocal interactif et vérifier en direct sur votre console asterisk : taper le numéro 100 sur votre interface Zoiper

```
root@debian:/var/lib/asterisk/sounds/fr# asterisk -rvvv
Asterisk 22.2.0, Copyright (C) 1999 - 2025, Sangoma Technologies Corporation and others.
Created by Mark Spencer <markster@digium.com>
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This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 22.2.0 currently running on debian (pid = 859)
-- Executing [1000Internal:1] Goto("PJSIP/6001-00000011", "ivr-menu,s,1") in new stack
-- Goto (ivr-menu,s,1)
-- Executing [s1ivr-menu:1] Answer("PJSIP/6001-00000011", "") in new stack
-- Executing [s1ivr-menu:2] Wait("PJSIP/6001-00000011", "1.5") in new stack
-- Executing [s1ivr-menu:3] Playback("PJSIP/6001-00000011", "fr/bienvenue2-ulaw") in new stack
<PJSIP/6001-00000011> Playing 'fr/bienvenue2-ulaw.slin' (language 'en')
-- Spawn extension (ivr-menu, s, 3) exited non-zero on 'PJSIP/6001-00000011'
debian@CLD
```

Etape 59 : Tester le numéro 1 de votre menu vocal interactif et vérifier en direct sur votre console asterisk : composer le 100 puis taper 1

```
root@debian:/var/lib/asterisk/sounds/fr# asterisk -rvvv
Asterisk 22.2.0, Copyright (C) 1999 - 2025, Sangoma Technologies Corporation and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 22.2.0 currently running on debian (pid = 859)
-- Executing [1000Internal:1] Goto("PJSIP/6002-00000013", "ivr-menu,s,1") in new stack
-- Goto (ivr-menu,s,1)
-- Executing [s1ivr-menu:1] Answer("PJSIP/6002-00000013", "") in new stack
-- Executing [s1ivr-menu:2] Wait("PJSIP/6002-00000013", "1.5") in new stack
-- Executing [s1ivr-menu:3] Playback("PJSIP/6002-00000013", "fr/bienvenue2-ulaw") in new stack
<PJSIP/6002-00000013> Playing 'fr/bienvenue2-ulaw.slin' (language 'en')
-- Executing [s1ivr-menu:4] WaitExten("PJSIP/6002-00000013", "10") in new stack
-- Executing [10ivr-menu:1] Playback("PJSIP/6002-00000013", "fr/transfert_support-ulaw") in new stack
<PJSIP/6002-00000013> Playing 'fr/transfert_support-ulaw.slin' (language 'en')
-- Executing [10ivr-menu:2] Dial("PJSIP/6002-00000013", "PJSIP/0001,20") in new stack
-- Called PJSIP/0001
-- PJSIP/0001-00000014 is ringing
-- PJSIP/0001-00000014 answered PJSIP/6002-00000013
-- Channel PJSIP/0001-00000014 joined 'single_bridge' basic-bridge <3C2e2746-dda4-4b33-93a9-35f482a7fcc0>
-- Channel PJSIP/6002-00000013 joined 'single_bridge' basic-bridge <3C2e2746-dda4-4b33-93a9-35f482a7fcc0>
(0x1f9029031e70) RTP audio difference is 140512680 set mark
-- Channel PJSIP/0001-00000014 left 'native_rtp' basic-bridge <3C2e2746-dda4-4b33-93a9-35f482a7fcc0>
-- Channel PJSIP/6002-00000013 left 'native_rtp' basic-bridge <3C2e2746-dda4-4b33-93a9-35f482a7fcc0>
-- Spawn extension (ivr-menu, 1, 2) exited non-zero on 'PJSIP/6002-00000013'
debian@CLD _
```

Etape 60 : Tester le numéro 2 de votre menu vocal interactif et vérifier en direct sur votre console asterisk: composer le 100 puis taper 2


```

Serv_VOIP - VMware Workstation
File Edit View VM Tabs Help
Home Serv_VOIP

root@debian:/var/lib/asterisk/sounds/fr# asterisk -rvvv
Asterisk 22.2.0, Copyright (C) 1999 - 2025, Sangoma Technologies Corporation and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 22.2.0 currently running on debian (pid = 859)
-- Removed contact 'sip:8001@192.168.185.171:49748;rinstance=86e8f429098bdfd18' from AOR '8001' due to request
== Contact 8001/sip:8001@192.168.185.171:49748;rinstance=86e8f429098bdfd18 has been deleted
-- Endpoint 8001 is now Unreachable
-- Added contact 'sip:8002@192.168.185.171:49748;rinstance=86e8f429098bdfd18' to AOR '8002' with expiration of 60 seconds
-- Endpoint 8002 is now Reachable
-- Executing [100@internal:1] Goto("PJSIP/6002-00000015", "ivr-menu,s,1") in new stack
-- Goto (ivr-menu,s,1)
-- Executing [s@ivr-menu:1] Answer("PJSIP/6002-00000015", "") in new stack
-- Executing [s@ivr-menu:2] Wait("PJSIP/6002-00000015", "1.5") in new stack
-- Executing [s@ivr-menu:3] Playback("PJSIP/6002-00000015", "fr/bienvenue2-ulaw") in new stack
-- PJSIP/6002-00000015: Playing 'fr/bienvenue2-ulaw.slin' (language 'en')
-- Executing [s@ivr-menu:4] WaitExten("PJSIP/6002-00000015", "10") in new stack
-- Executing [z@ivr-menu:1] Playback("PJSIP/6002-00000015", "fr/transfert_ventes-ulaw") in new stack
-- PJSIP/6002-00000015: Playing 'fr/transfert_ventes-ulaw.slin' (language 'en')
-- Executing [z@ivr-menu:2] Dial("PJSIP/6002-00000015", "PJSIP/8002,20") in new stack
-- Called PJSIP/8002
-- PJSIP/6002-00000015 is ringing
== Everyone is busy/congested at this time (1:1/0/0)
-- Executing [z@ivr-menu:3] Wait("PJSIP/6002-00000015", "1.5") in new stack
-- Executing [z@ivr-menu:4] Playback("PJSIP/6002-00000015", "fr/message_8002-ulaw") in new stack
-- PJSIP/6002-00000015: Playing 'fr/message_8002-ulaw.slin' (language 'en')
== Spawn extension (ivr-menu, 2, 4) exited non-zero on 'PJSIP/6002-00000015'
debian@CLI _
```

Etape 61 : Tester le numéro 3 de votre menu vocal interactif et vérifier en direct sur votre console asterisk : composer le 100 puis taper le 3

```

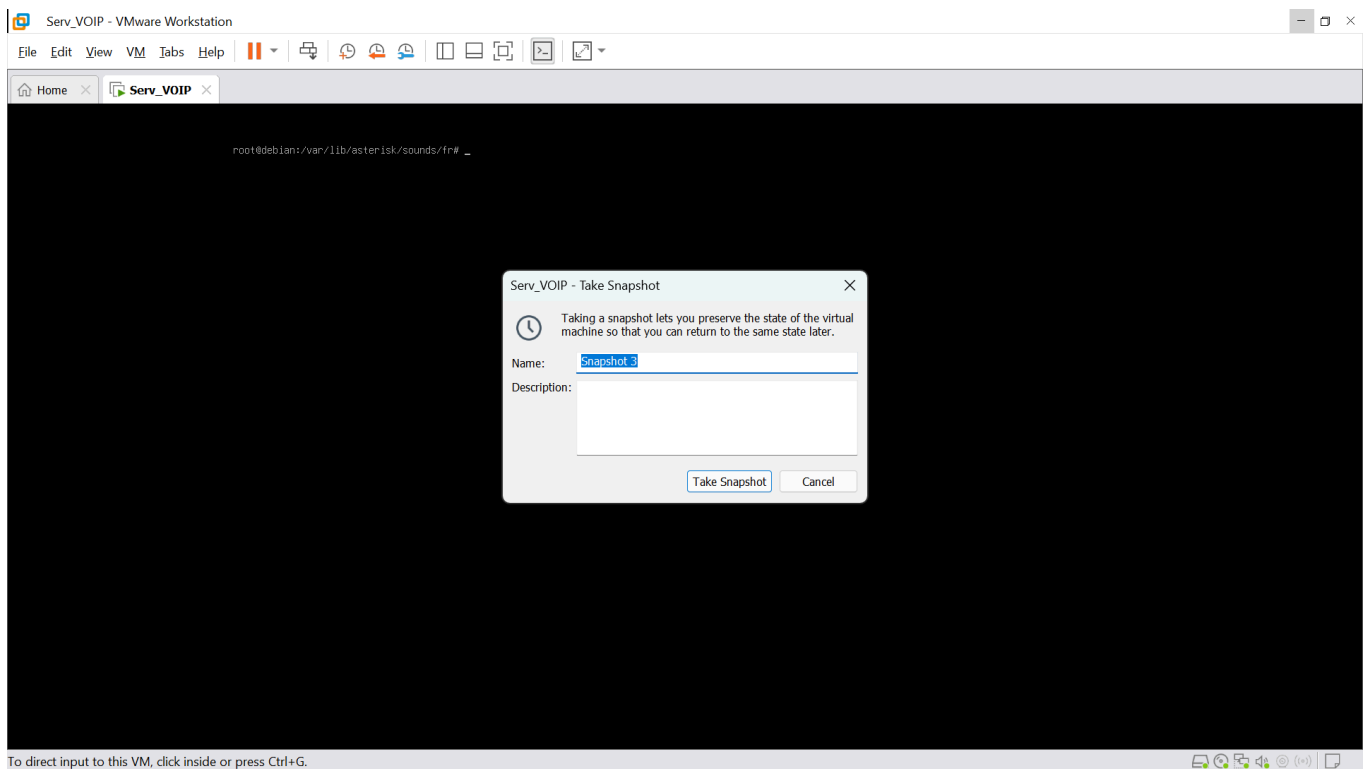
Serv_VOIP - VMware Workstation
File Edit View VM Tabs Help
Home Serv_VOIP

root@debian:/var/lib/asterisk/sounds/fr# asterisk -rvvv
Asterisk 22.2.0, Copyright (C) 1999 - 2025, Sangoma Technologies Corporation and others.
Created by Mark Spencer <markster@digium.com>
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License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 22.2.0 currently running on debian (pid = 859)
-- Executing [100@internal:1] Goto("PJSIP/6002-00000018", "ivr-menu,s,1") in new stack
-- Goto (ivr-menu,s,1)
-- Executing [s@ivr-menu:1] Answer("PJSIP/6002-00000018", "") in new stack
-- Executing [s@ivr-menu:2] Wait("PJSIP/6002-00000018", "1.5") in new stack
-- Executing [s@ivr-menu:3] Playback("PJSIP/6002-00000018", "fr/bienvenue2-ulaw") in new stack
-- PJSIP/6002-00000018: Playing 'fr/bienvenue2-ulaw.slin' (language 'en')
-- Executing [s@ivr-menu:4] WaitExten("PJSIP/6002-00000018", "10") in new stack
-- Executing [s@ivr-menu:1] Playback("PJSIP/6002-00000018", "fr/laissez_message-ulaw") in new stack
-- PJSIP/6002-00000018: Playing 'fr/laissez_message-ulaw.slin' (language 'en')
-- Executing [s@ivr-menu:2] VoiceMail("PJSIP/6002-00000018", "1000@default,u") in new stack
-- PJSIP/6002-00000018: Playing 'vm-theperson.gsm' (language 'en')
-- PJSIP/6002-00000018: Playing 'digits/1.gsm' (language 'en')
-- PJSIP/6002-00000018: Playing 'digits/0.gsm' (language 'en')
-- PJSIP/6002-00000018: Playing 'digits/0.gsm' (language 'en')
-- PJSIP/6002-00000018: Playing 'vm-isumavall.gsm' (language 'en')
-- PJSIP/6002-00000018: Playing 'vm-intro.gsm' (language 'en')
-- PJSIP/6002-00000018: Playing 'beep.gsm' (language 'en')
-- Recording the message
-- User hung up
== Spawn extension (ivr-menu, 3, 2) exited non-zero on 'PJSIP/6002-00000018'
debian@CLI _
```

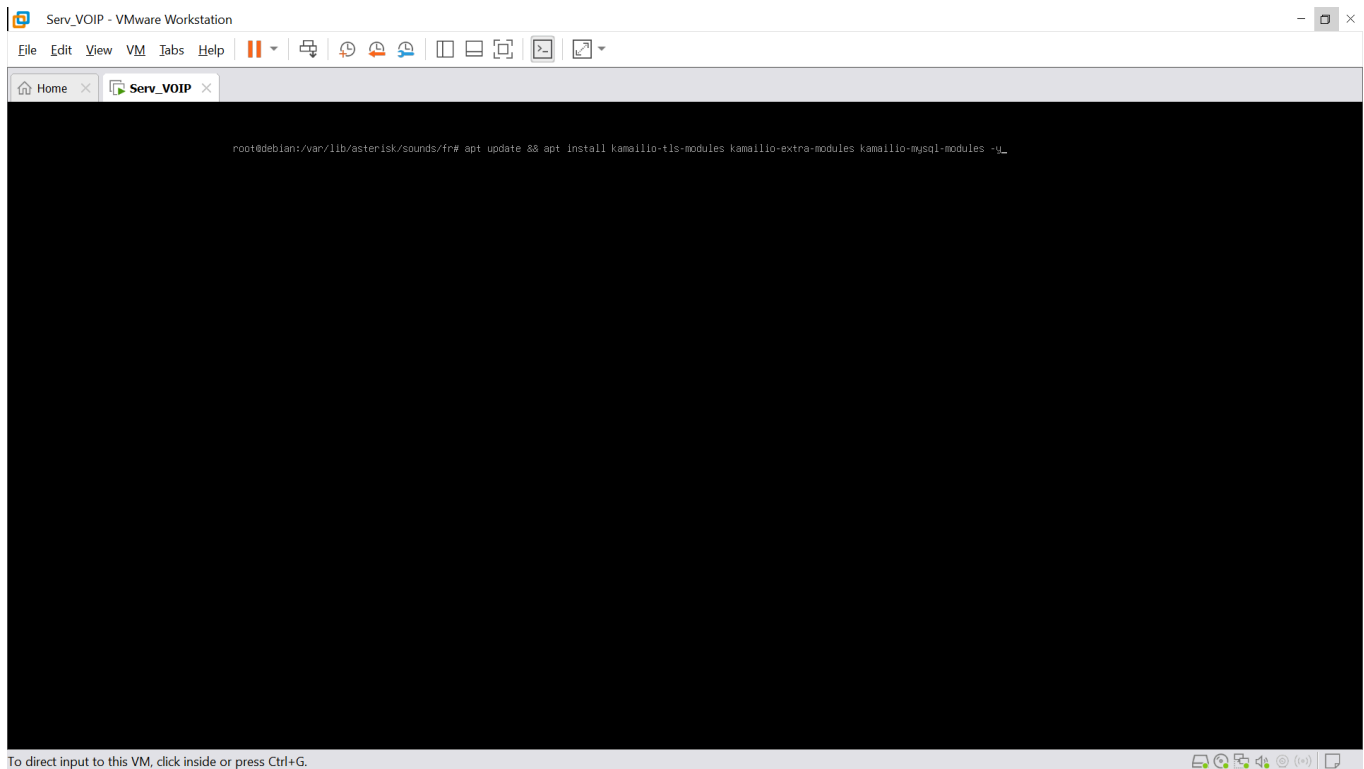
Etape 62 : Tester le numéro 4,5,6,7,8,9 de votre menu vocal interactif et vérifier en direct sur votre console asterisk: composer le 100 puis taper 4,5,6,7,8 ou 9

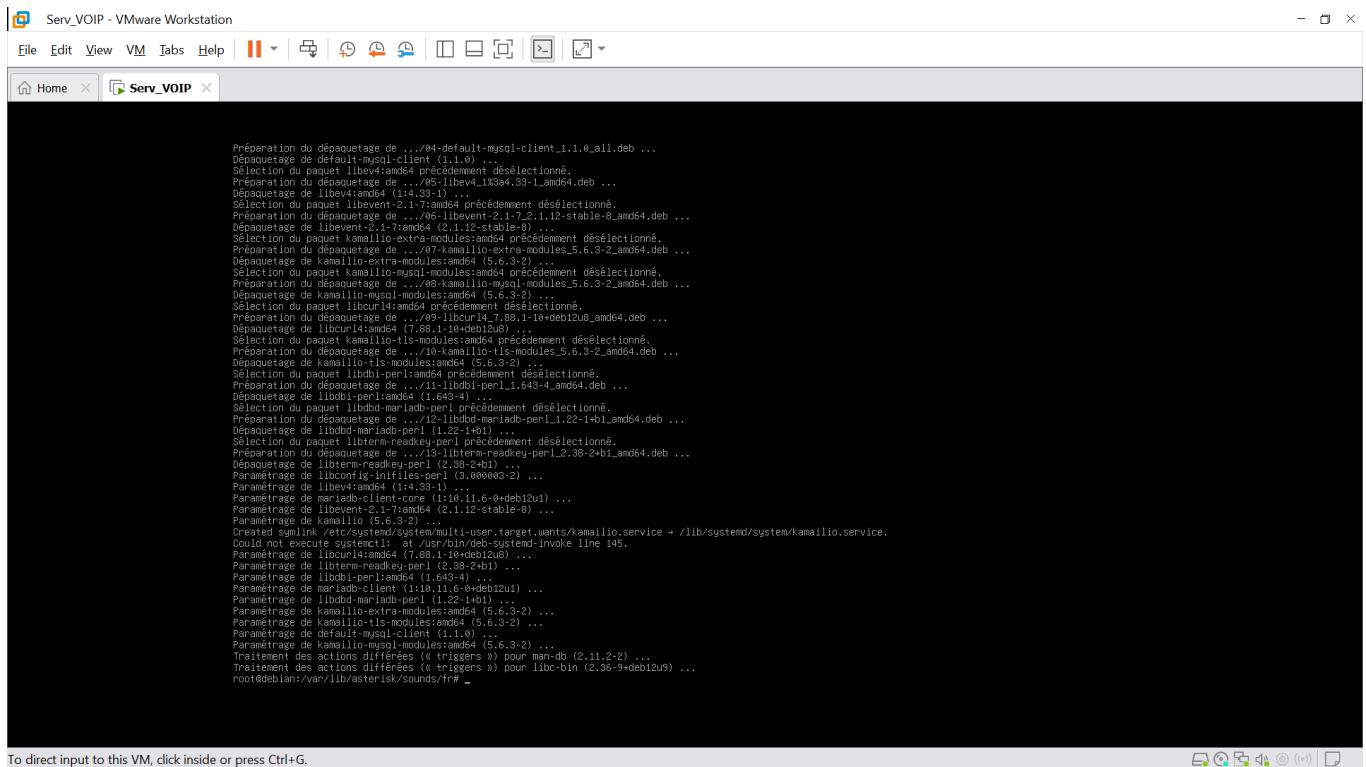
Etape 63 : Tester la messagerie vocale de votre utilisateur 6001 et vérifier en direct sur votre console asterisk : composer le 97

Etape 64 : Sauvegarder votre VM : snapshot 3



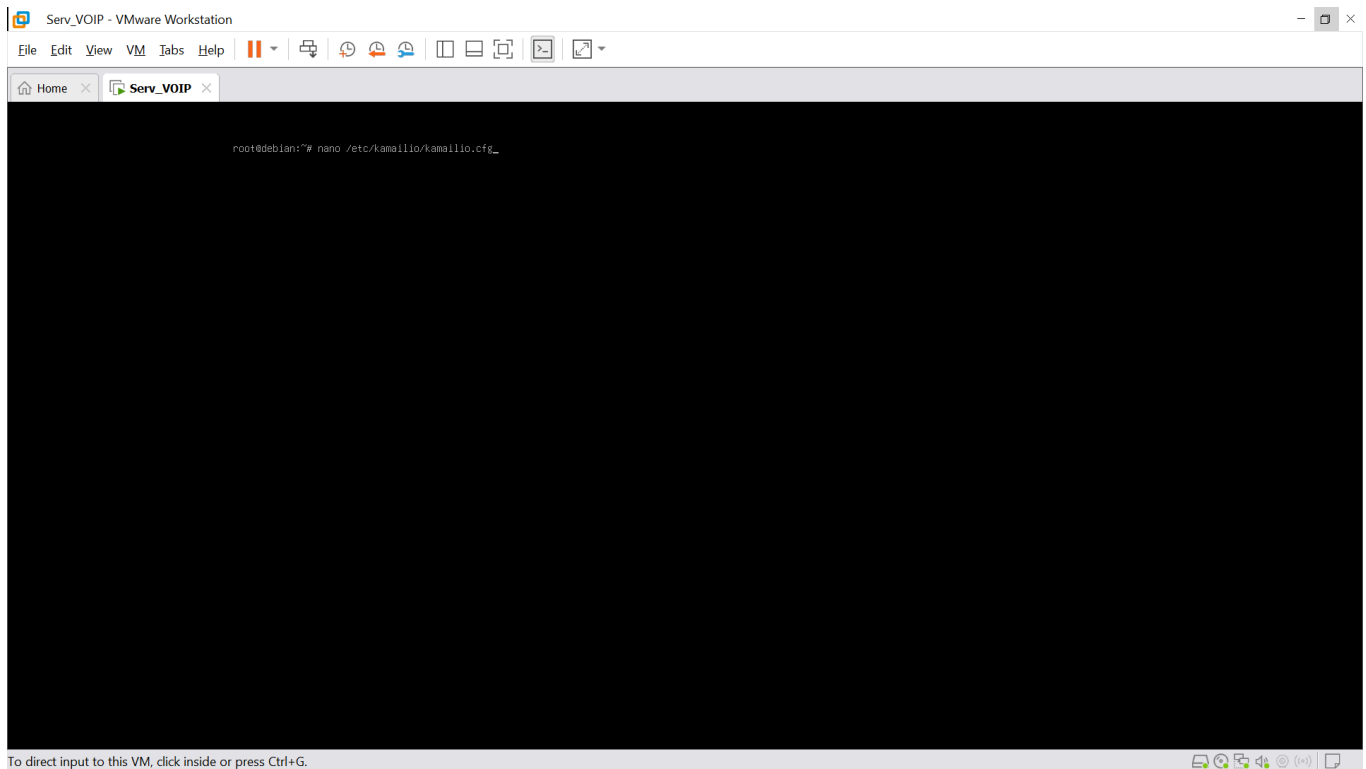
Etape 65 : Télécharger et installer Kamailio : apt update && apt install kamailio-tls-modules kamailio-extra-modules kamailio-mysql-modules -y





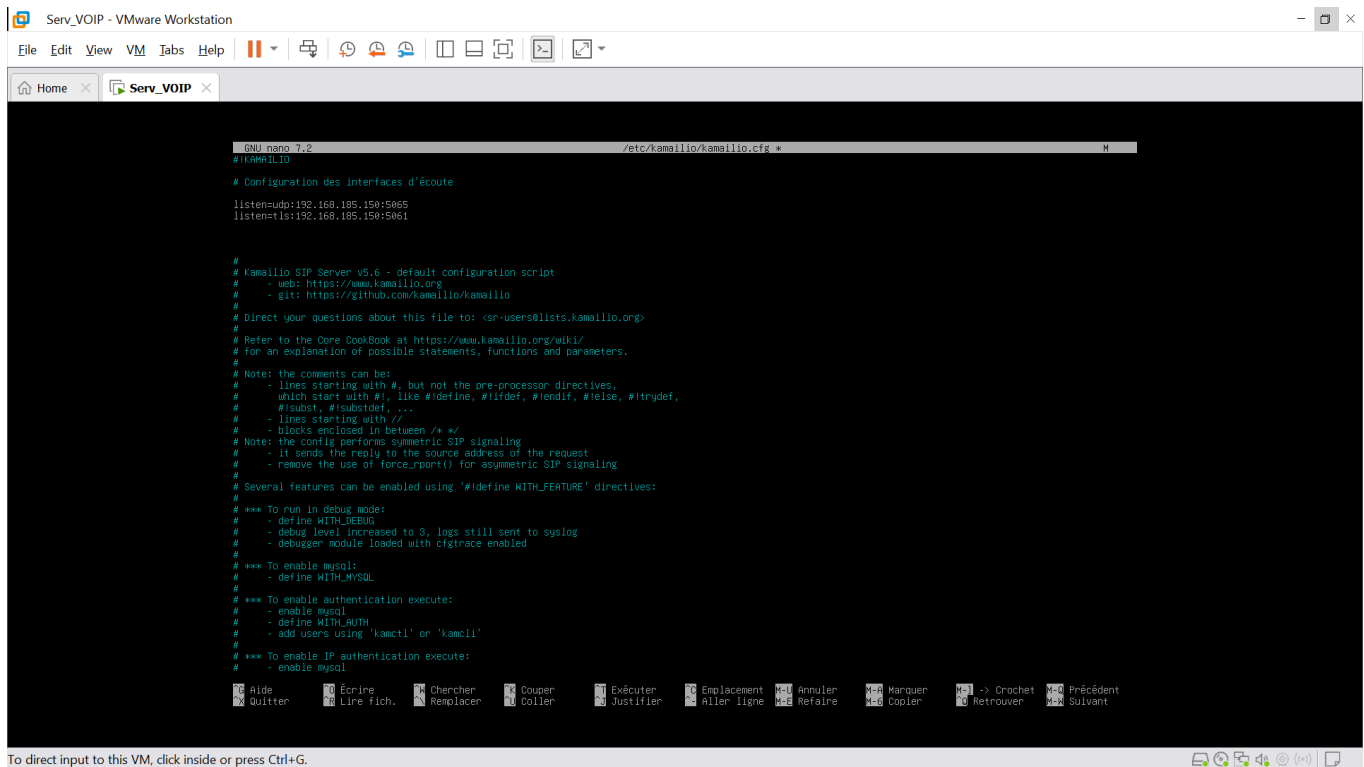
```
Préparation du dépaquetage de .../04-default-mysql-client_1.1.0_all.deb ...
Dépaquetage de default-mysql-client (1.1.0) ...
Sélection du paquet libevent2.1-7amd64 précédemment désélectionné.
Préparation du dépaquetage de .../05-libevent2.1-7amd64.deb ...
Dépaquetage de libevent2.1-7amd64 (2.1.12-stable-8) ...
Sélection du paquet libevent2.1-7amd64 précédemment désélectionné.
Préparation du dépaquetage de .../06-libevent2.1-7amd64.deb ...
Dépaquetage de libevent2.1-7amd64 (2.1.12-stable-8) ...
Sélection du paquet kamailio-extra-modules:amd64 précédemment désélectionné.
Préparation du dépaquetage de .../07-kamailio-extra-modules_5.6.3-2_amd64.deb ...
Dépaquetage de kamailio-extra-modules:amd64 (5.6.3-2) ...
Sélection du paquet kamailio-mysql-modules:amd64 précédemment désélectionné.
Préparation du dépaquetage de .../08-kamailio-mysql-modules_5.6.3-2_amd64.deb ...
Dépaquetage de kamailio-mysql-modules:amd64 (5.6.3-2) ...
Sélection du paquet libcurl4:amd64 précédemment désélectionné.
Préparation du dépaquetage de .../09-libcurl4_7.80.1-10+deb12u8_amd64.deb ...
Dépaquetage de libcurl4:amd64 (7.80.1-10+deb12u8) ...
Sélection du paquet kamailio-tls-modules:amd64 précédemment désélectionné.
Préparation du dépaquetage de .../10-kamailio-tls-modules_5.6.3-2_amd64.deb ...
Dépaquetage de kamailio-tls-modules:amd64 (5.6.3-2) ...
Sélection du paquet libdbi-perl:amd64 précédemment désélectionné.
Préparation du dépaquetage de .../11-libdbi-perl_1.643-4_amd64.deb ...
Dépaquetage de libdbi-perl:amd64 (1.643-4) ...
Sélection du paquet libdbd-mariadb-perl précédemment désélectionné.
Préparation du dépaquetage de .../12-libdbd-mariadb-perl_1.22-1+b1_amd64.deb ...
Dépaquetage de libdbd-mariadb-perl (1.22-1+b1) ...
Sélection du paquet libterm-readkey-perl précédemment désélectionné.
Préparation du dépaquetage de .../13-libterm-readkey-perl_2.38-2+b1_amd64.deb ...
Dépaquetage de libterm-readkey-perl (2.38-2+b1) ...
Paramétrage de libcurl4:amd64 (7.80.1-10+deb12u8) ...
Paramétrage de libevent:amd64 (1:2.1.12-stable-8) ...
Paramétrage de kamailio-client-core (1:10.11.6-0+deb12u1) ...
Paramétrage de kamailio (5.6.3-2) ...
Created symlink /etc/systemd/system/multi-user.target.wants/kamailio.service → /lib/systemd/system/kamailio.service.
Could not execute systemctl: at /usr/bin/deb-systemd-invoke line 145.
Paramétrage de libterm-readkey-perl (2.38-2+b1) ...
Paramétrage de libdbi-perl:amd64 (1.643-4) ...
Paramétrage de libdbd-mariadb-perl (1.22-1+b1) ...
Paramétrage de kamailio-extra-modules:amd64 (5.6.3-2) ...
Paramétrage de kamailio-tls-modules:amd64 (5.6.3-2) ...
Paramétrage de default-mysql-client (1.1.0) ...
Paramétrage de kamailio-mysql-modules:amd64 (5.6.3-2) ...
Traitement des actions différées (« triggers ») pour man-db (2.11.2-2) ...
Traitement des actions différées (« triggers ») pour libc-bin (2.36-9+deb12u9) ...
root@debian:~#
```

Etape 66 : Editer votre fichier kamailio.cfg : nano /etc/kamailio.cfg



```
root@debian:~# nano /etc/kamailio/kamailio.cfg
```

Etape 67 : Configurer vos interfaces d'écoute

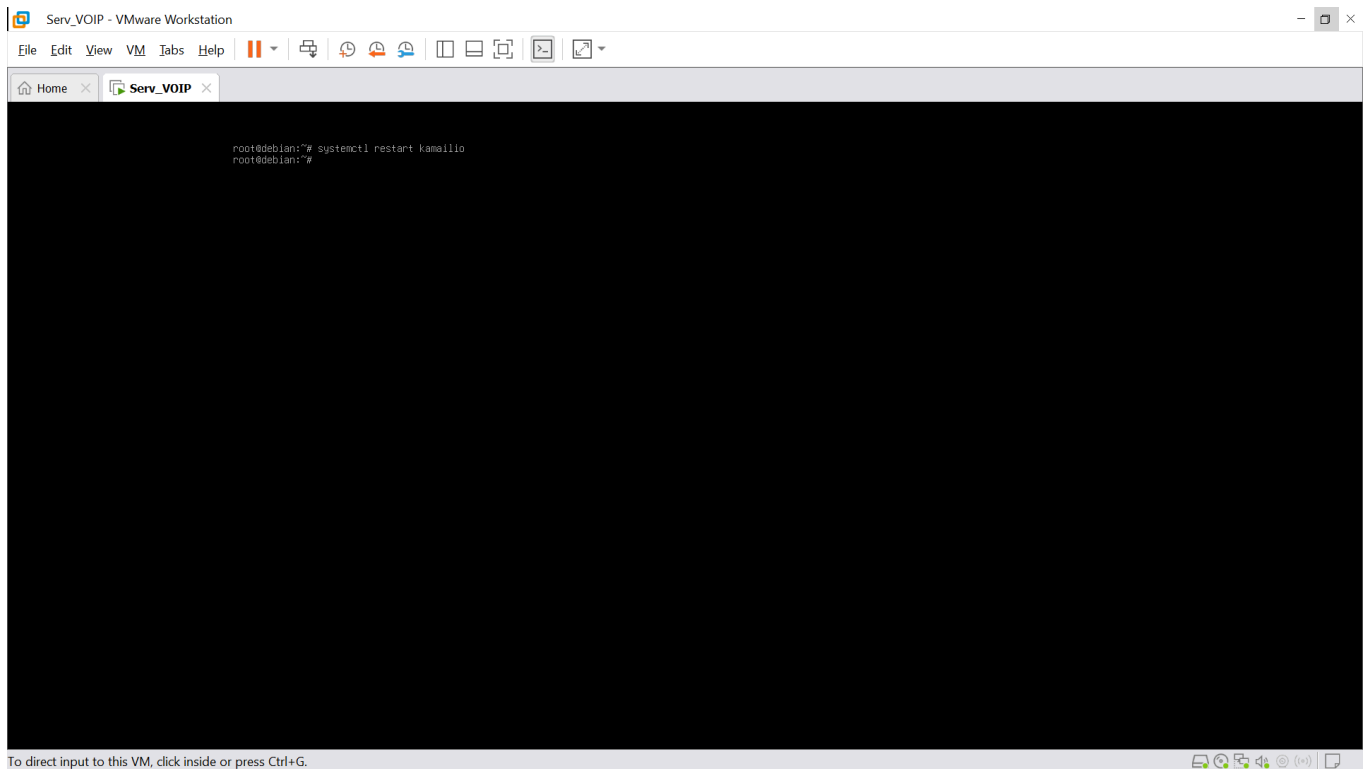


```
GNU nano 7.2 /etc/kamailio/kamailio.cfg
#KAMAILIO

# Configuration des interfaces d'écoute
listenudp:192.168.185.156:5065
listenudp:192.168.185.156:5061

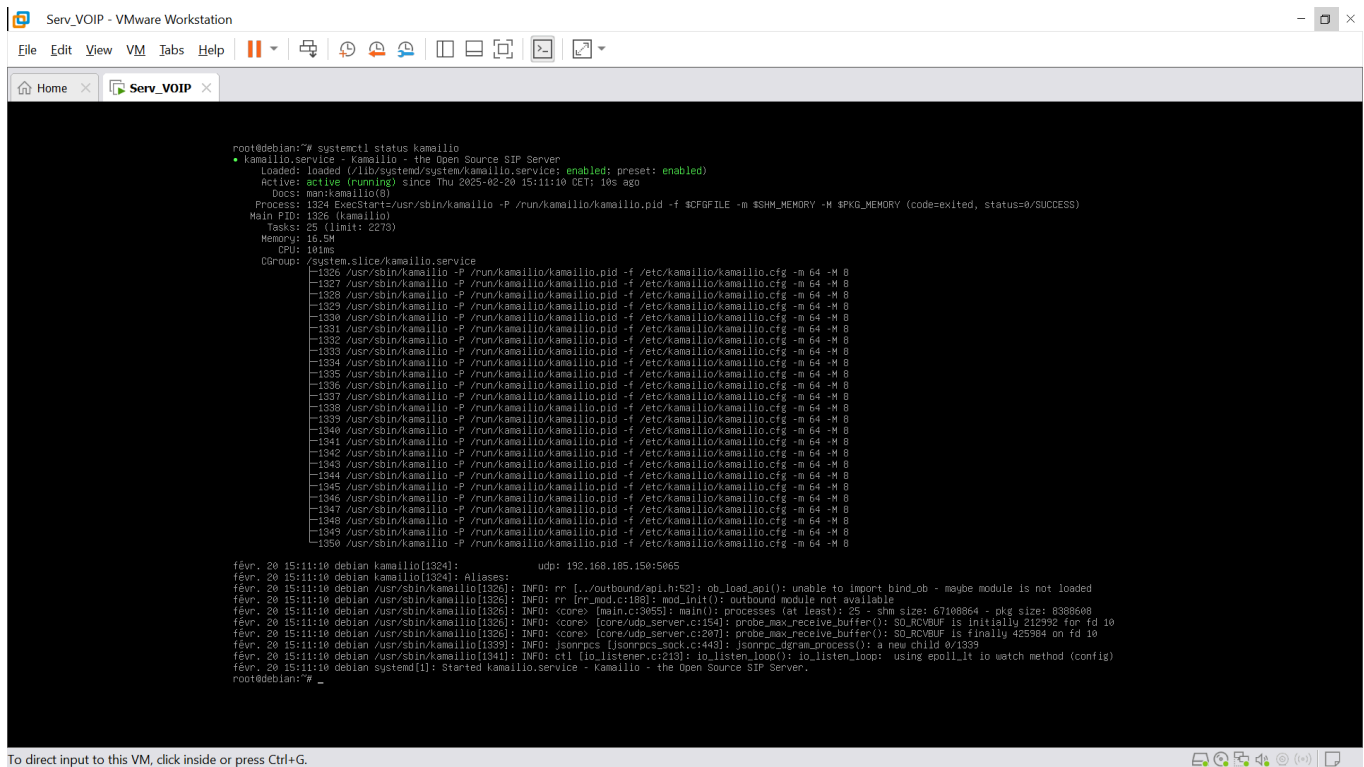
#
# Kamailio SIP Server v5.6 - default configuration script
# web: https://www.kamailio.org
# git: https://github.com/kamailio/kamailio
#
# Direct your questions about this file to: <sr-users@lists.kamailio.org>
#
# Refer to the Core Cookbook at https://www.kamailio.org/wiki/
# for an explanation of possible statements, functions and parameters.
#
# Note: the comments can be:
# - lines starting with #, but not the pre-processor directives,
# - which start with #!, like #define, #ifdef, #endif, #else, #ifndef,
# - #subst, #substdef, ...
# - lines starting with //
# - blocks enclosed in between /* */
# Note: the config performs symmetric SIP signaling
# - it sends the reply to the source address of the request
# - remove the use of force_rport() for asymmetric SIP signaling
#
# Several features can be enabled using '#ifdef WITH_FEATURE' directives:
#
# *** To run in debug mode:
# - define WITH_DEBUG
# - debug level increased to 3, logs still sent to syslog
# - debugger module loaded with cfgtrace enabled
#
# *** To enable mysql:
# - define WITH_MYSQL
#
# *** To enable authentication execute:
# - enable mysql
# - define WITH_AUTH
# - add users using 'kamctl' or 'kamcli'
#
# *** To enable IP authentication execute:
# - enable mysql
```

Etape 68 : Redémarrer Kamailio : systemctl restart kamailio



```
root@debian:~# systemctl restart kamailio
root@debian:~#
```

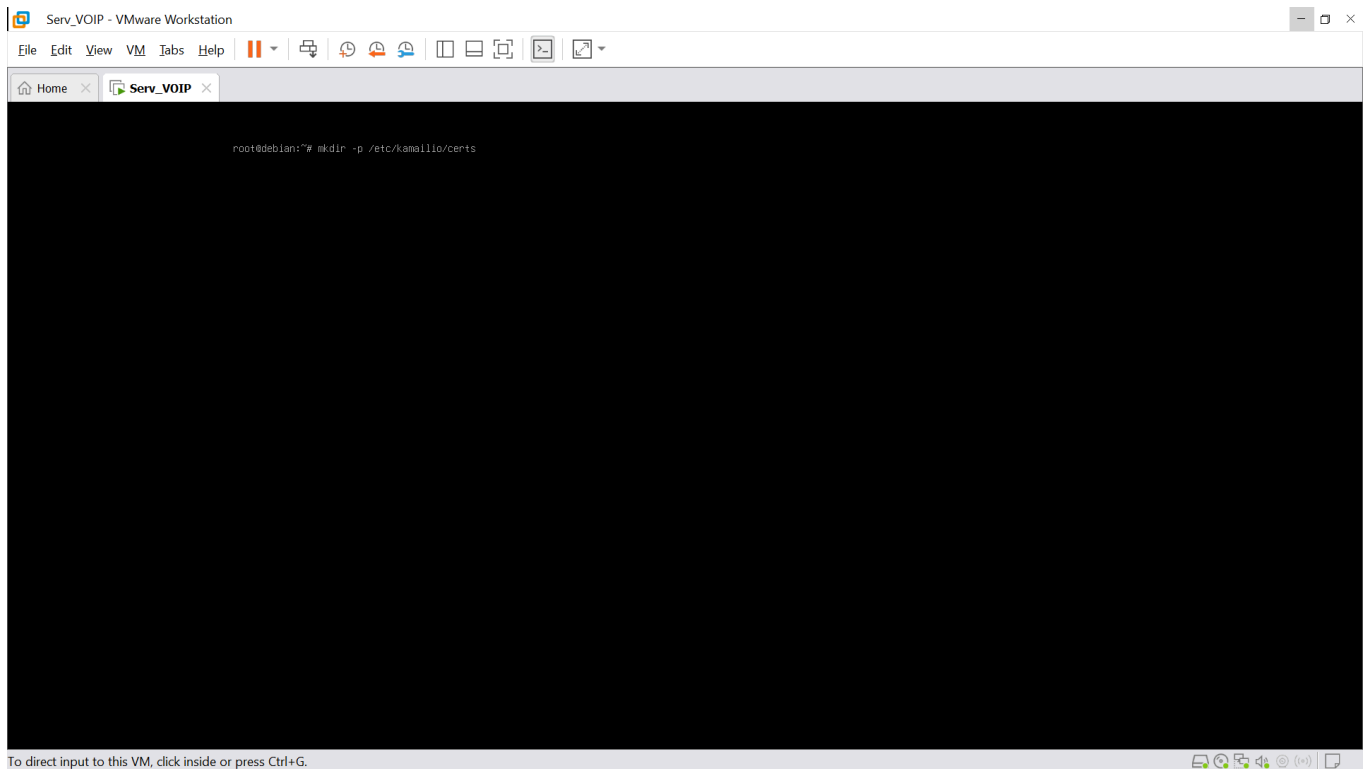
Etape 69 : Vérifier que Kamailio est actif : systemctl status Kamailio



```
root@debian:~# systemctl status kamailio
• kamailio.service - Kamailio - the Open Source SIP Server
   Loaded: loaded (/lib/systemd/system/kamailio.service; enabled; preset: enabled)
   Active: active (running) since Thu 2025-02-20 15:11:10 CET; 10s ago
     Docs: man:kamailio(8)
   Process: 1324 ExecStart=/usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f $CFGFILE -m $SHM_MEMORY -H $PKG_MEMORY (code=exited, status=0/SUCCESS)
   Main PID: 1326 (kamailio)
     Tasks: 25 (limit: 2273)
    Memory: 16.5M
       CPU: 101ms
   CGroup: /system.slice/kamailio.service
           └─ 1326 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1327 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1328 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1329 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1330 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1331 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1332 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1333 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1334 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1335 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1336 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1337 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1338 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1339 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1340 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1341 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1342 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1343 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1344 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1345 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1346 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1347 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1348 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1349 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0
             1350 /usr/sbin/kamailio -P /run/kamailio/kamailio.pid -f /etc/kamailio/kamailio.cfg -m 64 -M 0

févr. 20 15:11:10 debian kamailio[1324]:      udp: 192.168.185.150:5065
févr. 20 15:11:10 debian kamailio[1324]: Aliases:
févr. 20 15:11:10 debian /usr/sbin/kamailio[1326]: INFO: rr [/outbound/api.h:52]: ob_load_api(): unable to import bind_ob - maybe module is not loaded
févr. 20 15:11:10 debian /usr/sbin/kamailio[1326]: INFO: rr [rr_mod.c:188]: mod_init(): outbound module not available
févr. 20 15:11:10 debian /usr/sbin/kamailio[1326]: INFO: <core> [main.c:3055]: main(): processes (at least): 25 - shm size: 67168864 - pkg size: 8388608
févr. 20 15:11:10 debian /usr/sbin/kamailio[1326]: INFO: <core> [core/udp_server.c:154]: probe_max_receive_buffer(): SO_RCVBUF is initially 212392 for fd 10
févr. 20 15:11:10 debian /usr/sbin/kamailio[1326]: INFO: <core> [core/udp_server.c:297]: probe_max_receive_buffer(): SO_RCVBUF is finally 45584 on fd 10
févr. 20 15:11:10 debian /usr/sbin/kamailio[1339]: INFO: jsonrpcs [jsonrpcs_sock.c:443]: jsonrpc_dgram_process(): a new child 0/1339
févr. 20 15:11:10 debian /usr/sbin/kamailio[1341]: INFO: ctl [io_listener.c:213]: io_listen_loop(): io_listen_loop() using epoll_it to watch method (config)
févr. 20 15:11:10 debian systemd[1]: Started kamailio.service - Kamailio - the Open Source SIP Server.
root@debian:~# _
```

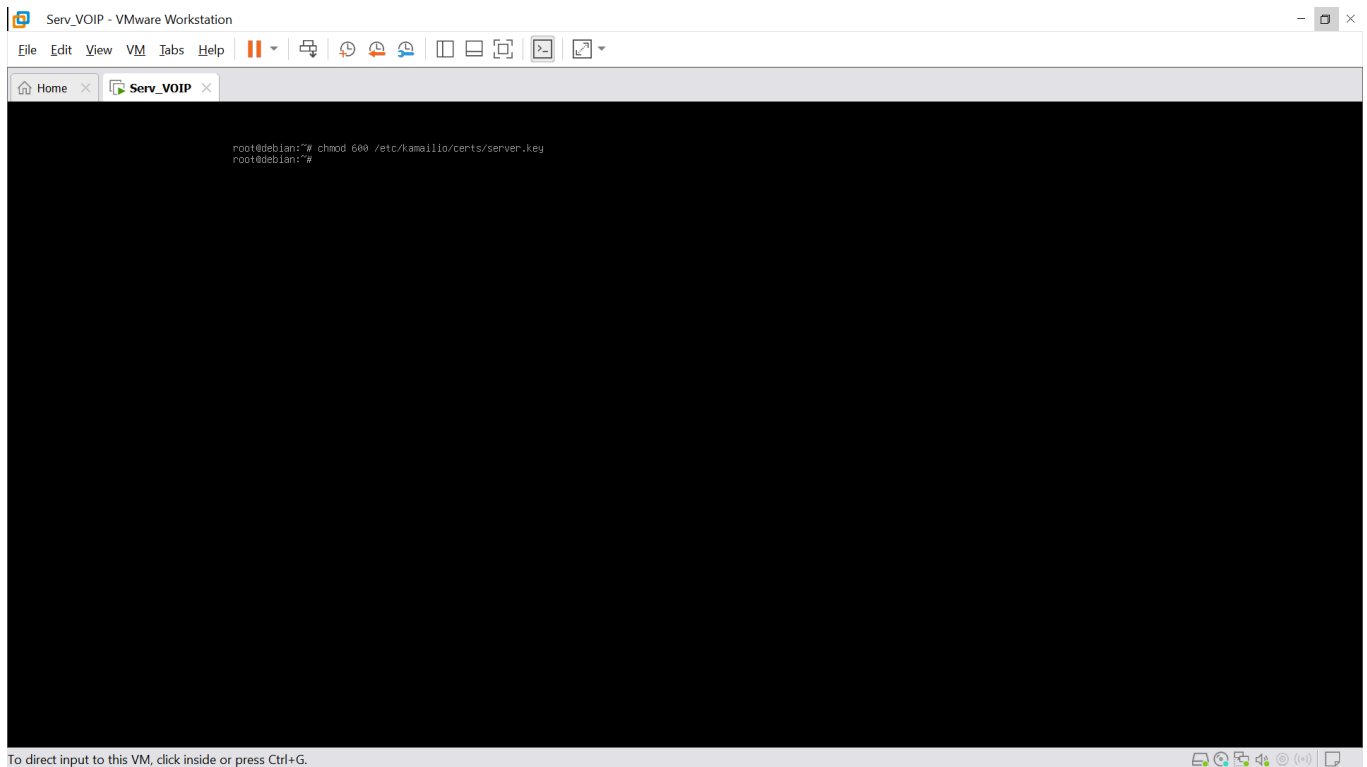
Etape 70 : Créer un dossier : mkdir -p /etc/kamailio/certs



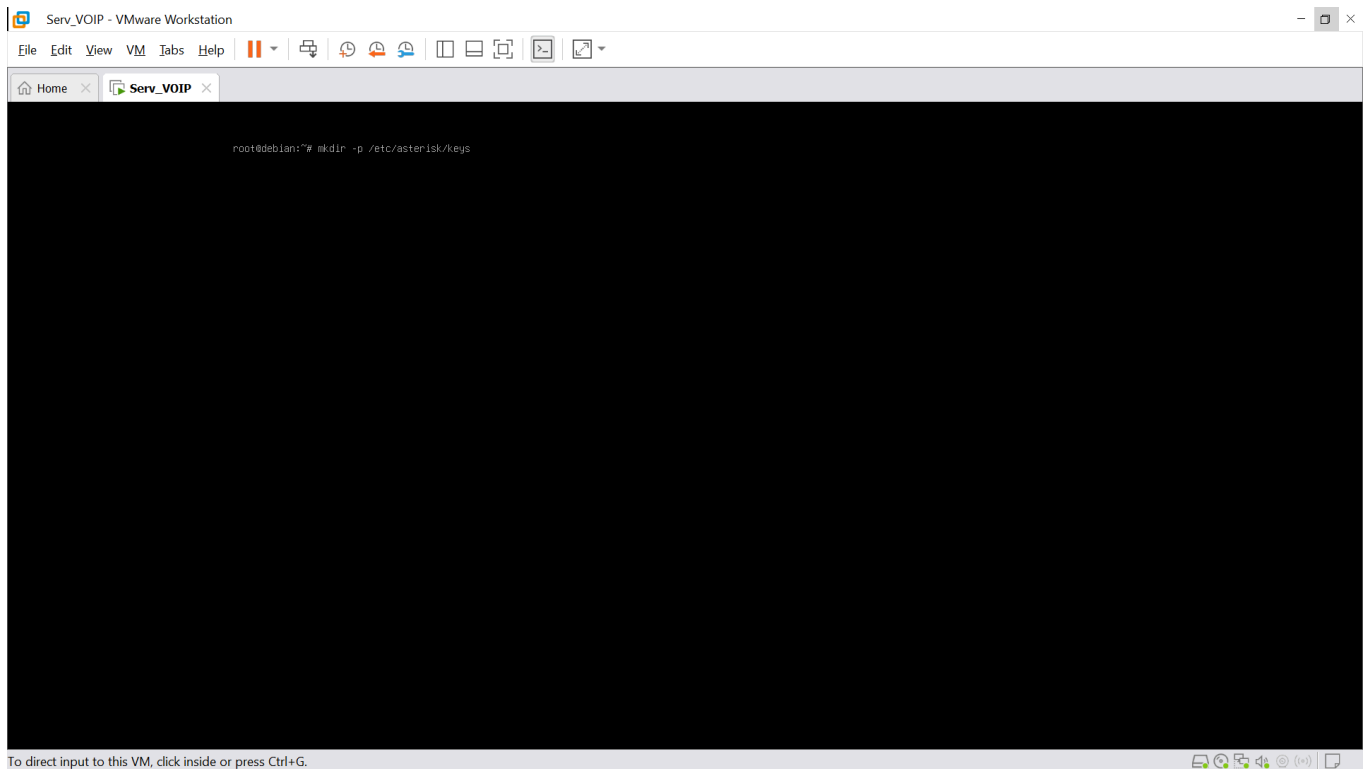
```
root@debian:~# mkdir -p /etc/kamailio/certs
```

Etape 71 : Générer un certificat TLS : openssl req -new -x509 -day -days 365 -nodes -out /etc/kamailio/certs/server.pem -keyout /etc/kamailio/certs/server.key

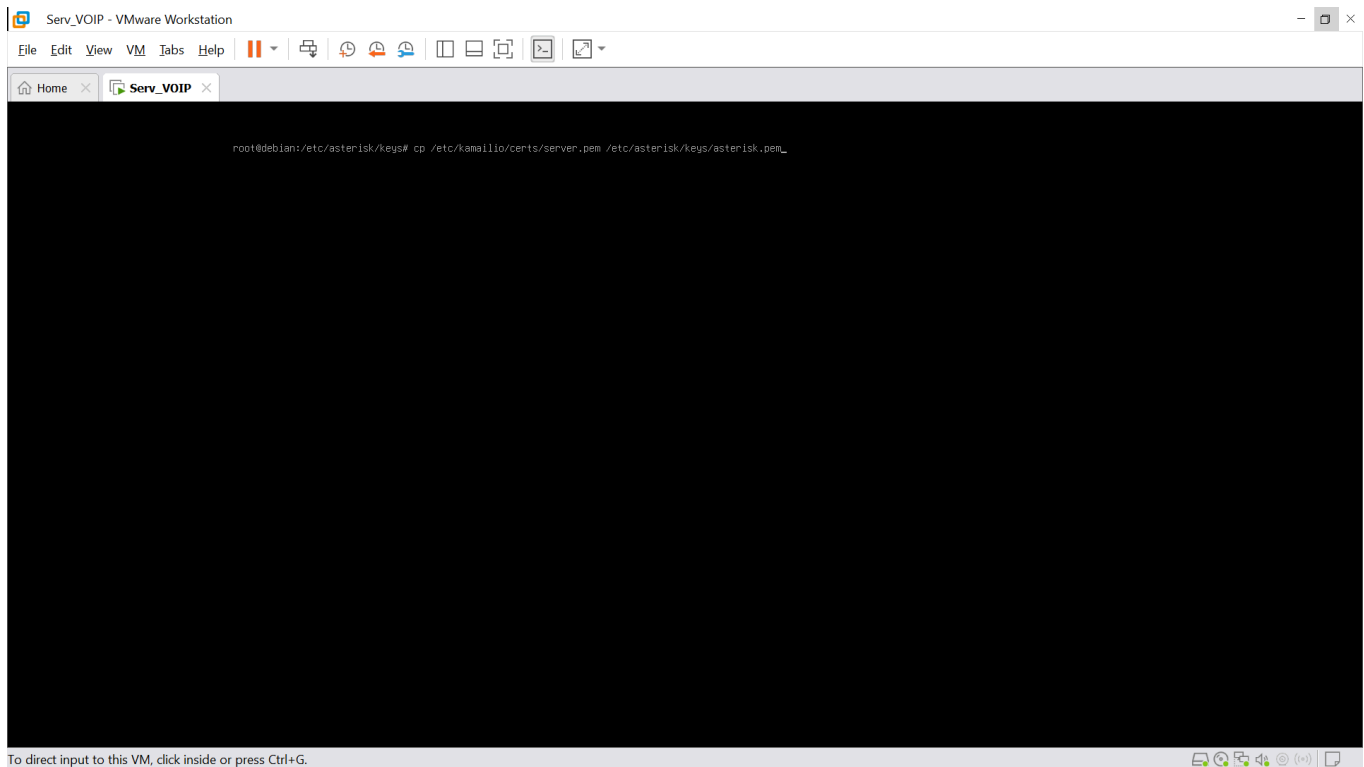
Etape 72 : Appliquer les droits utilisateurs requis pour votre dossier server.key : `chmod 600 /etc/kamailio/certs/server.key`



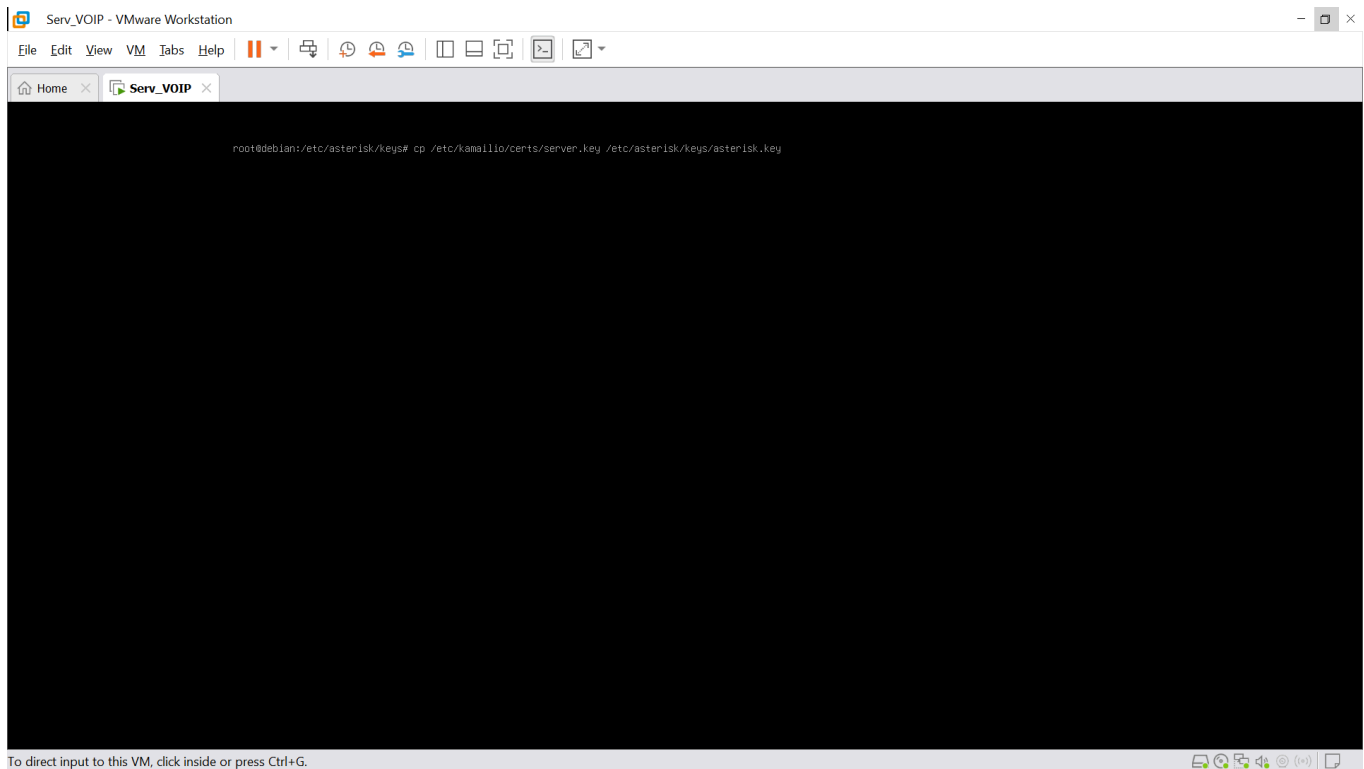
Etape 73 : Créer un dossier keys : `mkdir -p /etc/asterisk/keys`



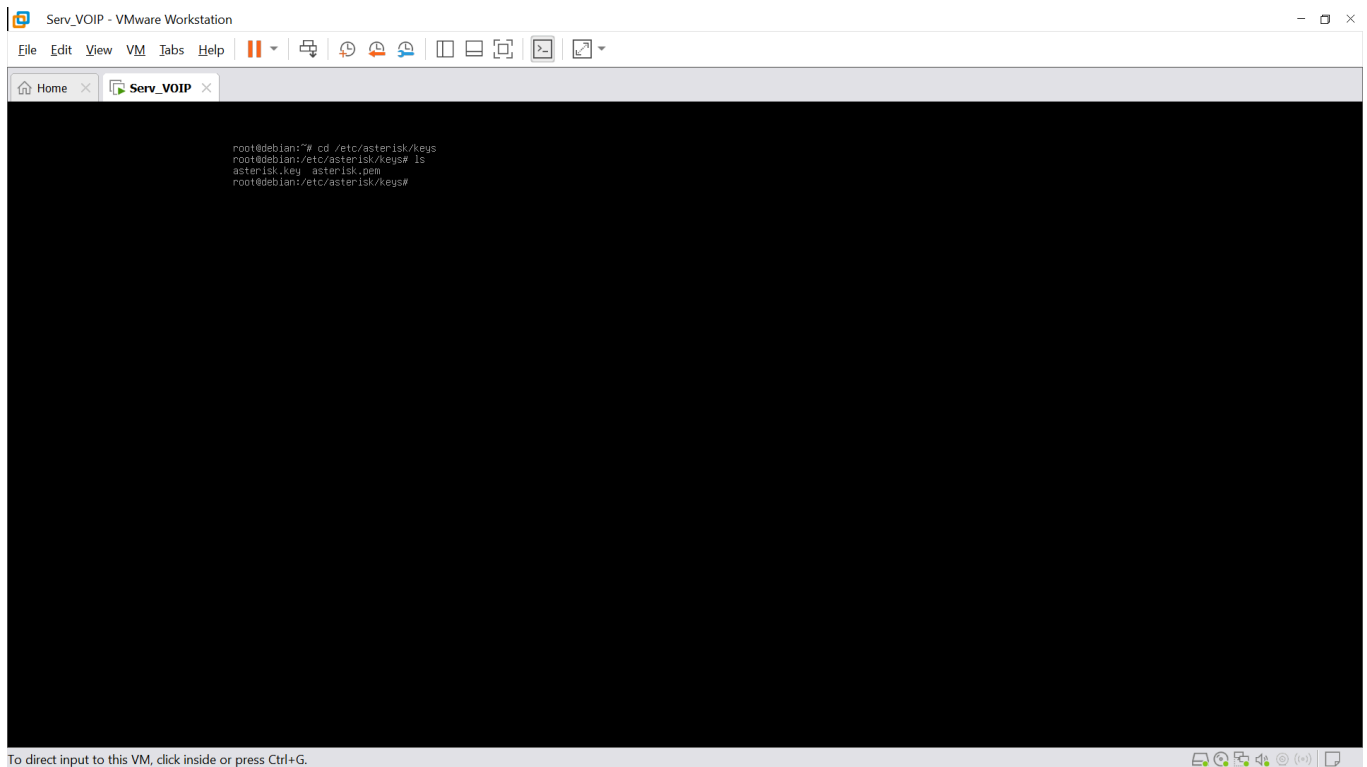
Etape 74 : Copier votre fichier server.pem dans votre dossier keys : `cp /etc/kamailio/certs/server.pem /etc/asterisk/keys/asterisk.pem`



Etape 75 : Copier votre fichier server.key dans votre dossier keys : `cp /etc/kamailio/certs/server.pem /etc/asterisk/keys/asterisk.key`

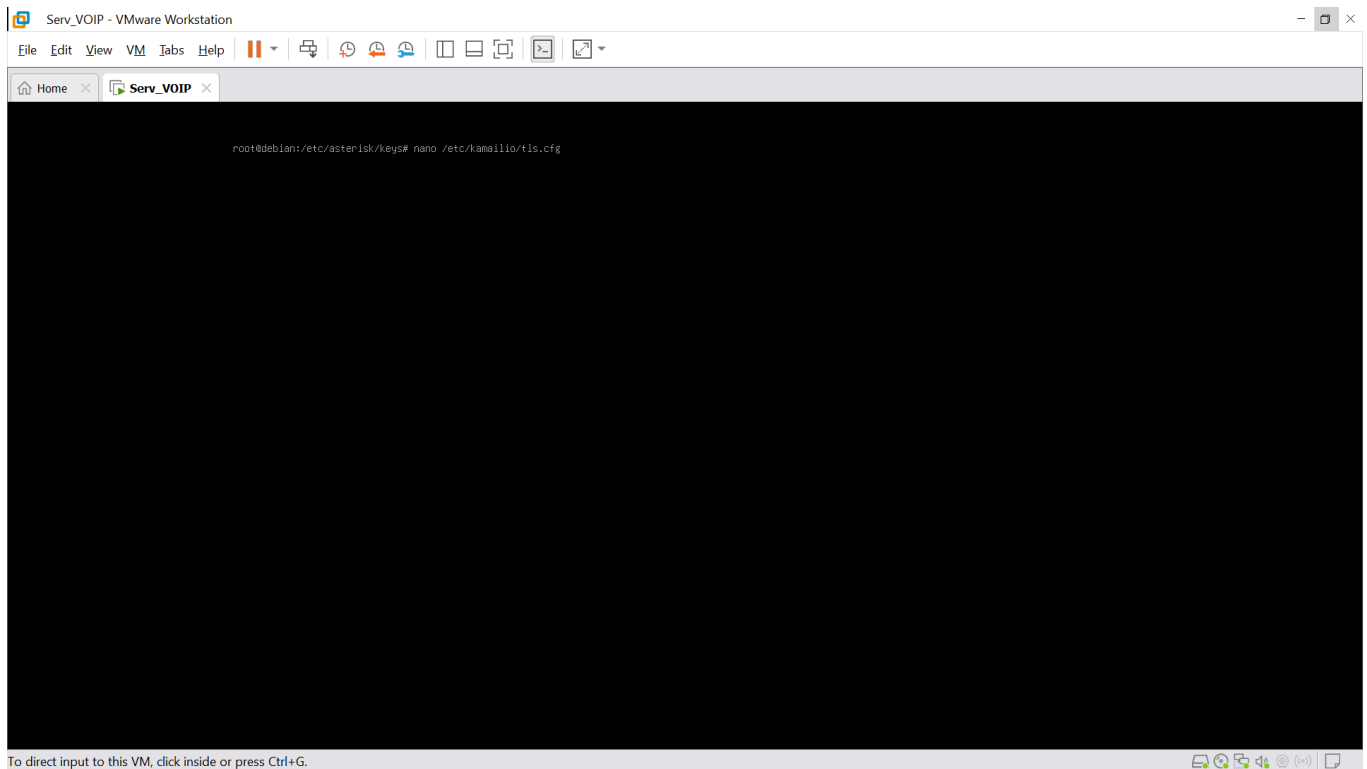


Etape 76 : Se déplacer dans votre dossier keys : `cd /etc/asterisk/keys` et lister son contenu : `ls`



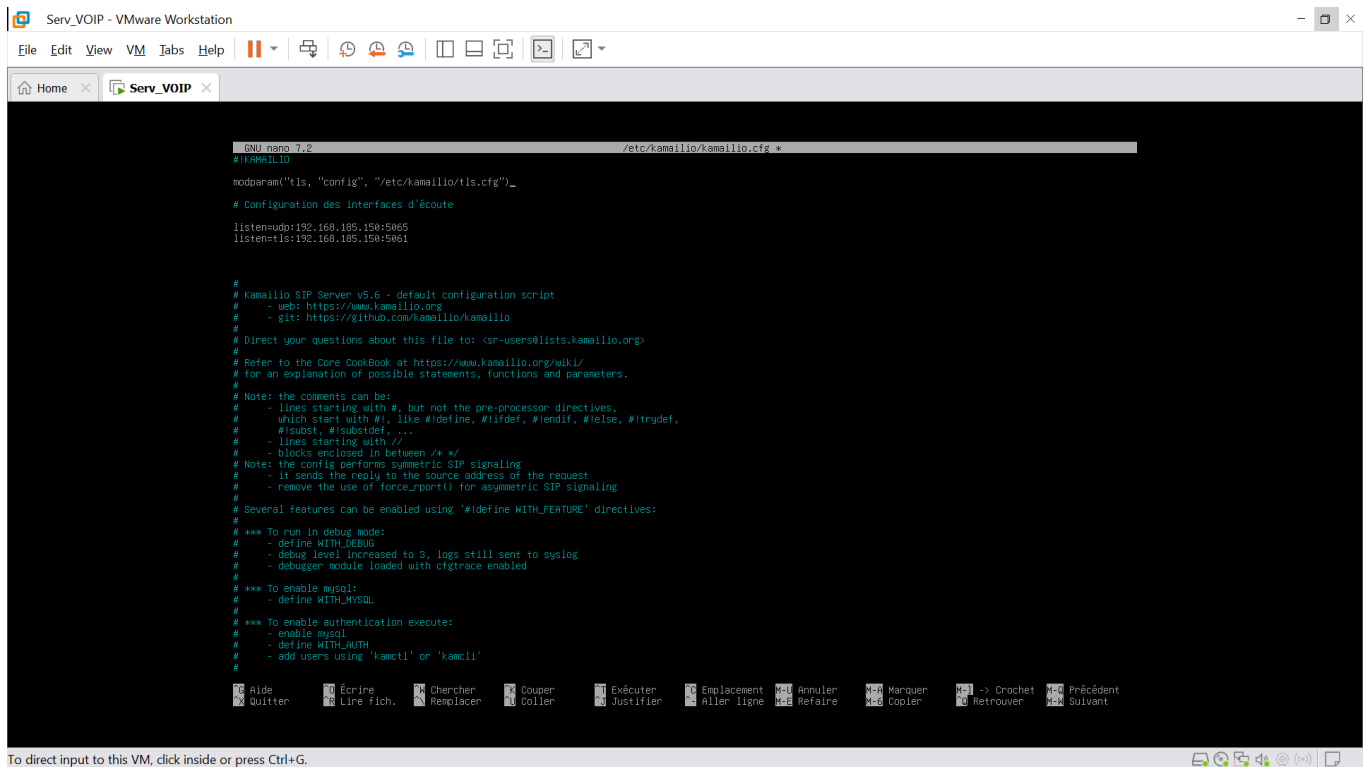
```
root@debian:~# cd /etc/asterisk/keys
root@debian:/etc/asterisk/keys# ls
asterisk.key  asterisk.pem
root@debian:/etc/asterisk/keys#
```

Etape 77 : Editer votre fichier kamailio.cfg : nano /etc/kamailio/kamailio.cfg



```
root@debian:/etc/asterisk/keys# nano /etc/kamailio/tls.cfg
```

Etape 78 : Configurer votre fichier kamailio.cfg :



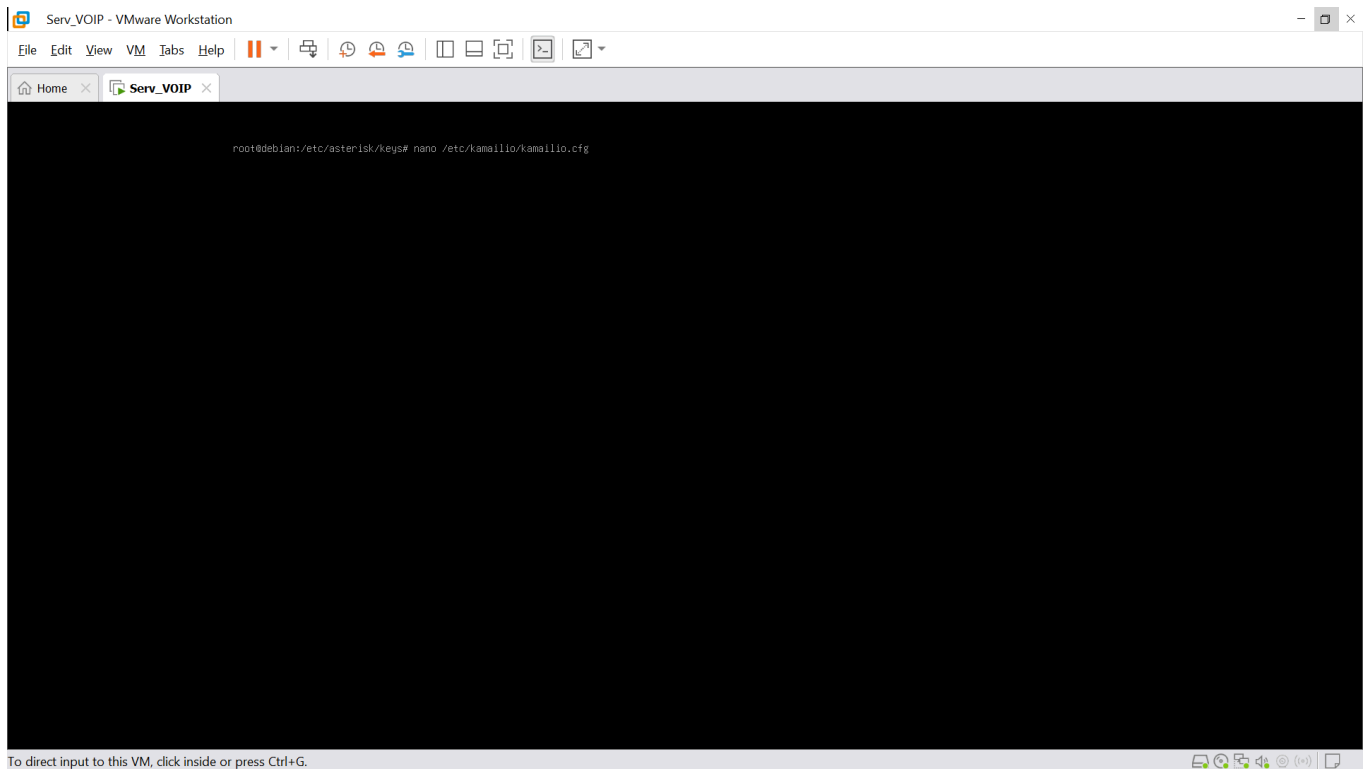
```
GNU nano 7.2 /etc/kamailio/kamailio.cfg
#KAMAILIO

modparam("tls", "config", "/etc/kamailio/tls.cfg")_

# Configuration des interfaces d'écoute
listen=udp:192.168.185.158:5865
listen=tls:192.168.185.158:5861

#
# Kamailio SIP Server v5.6 - default configuration script
# - web: https://www.kamailio.org
# - git: https://github.com/kamailio/kamailio
#
# Direct your questions about this file to: <sr-users@lists.kamailio.org>
#
# Refer to the Core Cookbook at https://www.kamailio.org/wiki/
# for an explanation of possible statements, functions and parameters.
#
# Note: the comments can be:
# - lines starting with #, but not the pre-processor directives,
#   which start with #!, like #define, #ifdef, #endif, #else, #ifndef,
#   #subst, #substdef, ...
# - lines starting with //
# - blocks enclosed in between /* */
# Note: the config performs symmetric SIP signaling
# - it sends the reply to the source address of the request
# - remove the use of force_rport() for asymmetric SIP signaling
#
# Several features can be enabled using 'Wdefine WITH_FEATURE' directives:
#
# *** To run in debug mode:
# - define WITH_DEBUG
# - debug level increased to 3, logs still sent to syslog
# - debugger module loaded with cgttrace enabled
#
# *** To enable mysql:
# - define WITH_MYSQL
#
# *** To enable authentication execute:
# - enable mysql
# - define WITH_AUTH
# - add users using 'kamctl' or 'kamcli'
#
#
# Aide quitter    # Écrire    # Chercher    # Couper    # Exécuter    # Emplacement    # Annuler    # Marquer    # Ctrl-> Crochet    # Précédent
# Quitter        # Lire fich.  # Remplacer   # Colier    # Justifier   # Aller ligne    # Refaire    # Copier     # Retrouver    # Suivant
```

Etape 79 : Editer votre fichier tls.cfg : nano /etc/kamailio/tls.cfg



```
root@debian:/etc/asterisk/keys# nano /etc/kamailio/kamailio.cfg
```

Etape 80 : Configurer votre fichier tls.cfg

```
GNU nano 7.2 /etc/kamailio/tls.cfg *
#
# Example Kamailio TLS Configuration File
#
# ---
# This is the default server domain profile.
# Settings in this domain will be used for all incoming
# connections that do not match any other server
# domain in this configuration file.
#
# We do not enable anything else than TLSv1.2+
# over the public Internet. Clients do not have
# to present client certificates by default.
#
[server:default]
method = TLSv1.2+
verify_certificate = no
require_certificate = no
private_key = /etc/kamailio/certs/server.key
certificate = /etc/kamailio/certs/server.pem
#ca_list = /etc/kamailio/tls/cacert.pem
#cri = /etc/kamailio/tls/crl.pem

# ---
# This is the default client domain profile.
# Settings in this domain will be used for all outgoing
# TLS connections that do not match any other
# client domain in this configuration file.
# We require that server's present valid certificate.
#
[client:default]
Method = TLSv1.2+
verify_certificate = yes
require_certificate = yes

# ---
# This is an example server domain for TLS connections
# received from the loopback interface. We allow
# the use of TLSv1.2+ protocols here, we do
# not require that clients present client certificates
# but if they present it it must be valid. We also use
# a special certificate and CA list for loopback
# interface.
#
[server:5.6.7.8:5061]
Method = TLSv1.2+
```

Etape 81 : Router les appels SIP vers Asterisk dans votre fichier kamailio.cfg : nano /etc/kamailio/kamailio.cfg

```
GNU nano 7.2 /etc/kamailio/kamailio.cfg *
#KAMAILIO

modparam("tls", "config", "/etc/kamailio/tls.cfg")

# Configuration des interfaces d'ecoute

listen=udp:192.168.185.150:5065
listen=tls:192.168.185.150:5061

# Routage des appels SIP vers Asterisk

if ($nd == "192.168.185.150") {
    forward("192.168.185.150:5060");
    exit;
}

#
# Kamailio SIP Server v5.6 - default configuration script
# - web: https://www.kamailio.org
# - git: https://github.com/kamailio/kamailio
#
# Direct your questions about this file to: <en-users@lists.kamailio.org>
#
# Refer to the Core Cookbook at https://www.kamailio.org/wiki/
# for an explanation of possible statements, functions and parameters.
#
# Note: the comments can be:
# - lines starting with #, but not the pre-processor directives,
#   which start with #!, like #define, #ifdef, #endif, #else, #ifndef,
#   #subst, #substdef, ...
# - lines starting with //
# - blocks enclosed in between /* */
# Note: the config performs symmetric SIP signaling
# - it sends the reply to the source address of the request
# - remove the use of force_rport() for asymmetric SIP signaling
#
# Several features can be enabled using '#define WITH_FEATURE' directives:
#
# *** To run in debug mode:
# - define WITH_DEBUG
# - debug level increased to 3, logs still sent to syslog
# - debugger module loaded with cfgtrace enabled
#
# *** To enable mysql:
# - define WITH_MYSQL
#
```

Etape 72 : Editer le fichier kamailio.cfg : nano /etc/kamailio/kamailio.cfg

Etape 73 : Editer le fichier tls.cfg : nano /etc/kamailio/tls.cfg

Etape 74 : Redémarrer Kamailio : systemctl restart kamailio

Etape 75 : Faire un backup de votre fichier pjsip.conf :

Etape 76 : Editer votre fichier pjsip.conf : nano /etc/asterisk/pjsip.conf

Etape 77 : Redémarrer Asterisk : systemctl restart asterisk