



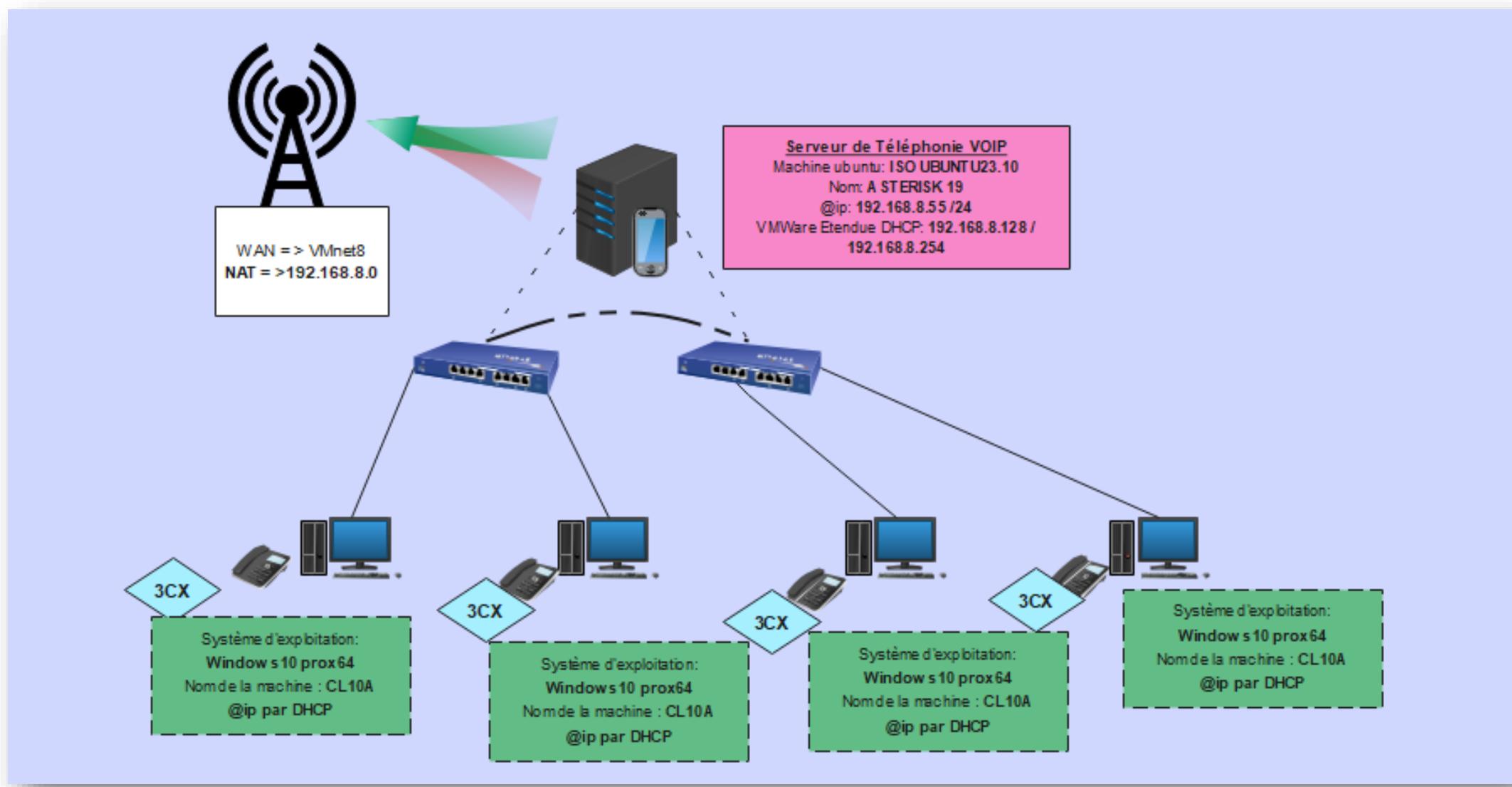
PROJET SUR LA TELEPHONIE VOIP

ТЕЛЕФОНИЕ ВОИП

Date réalisation du projet

01/12/2023

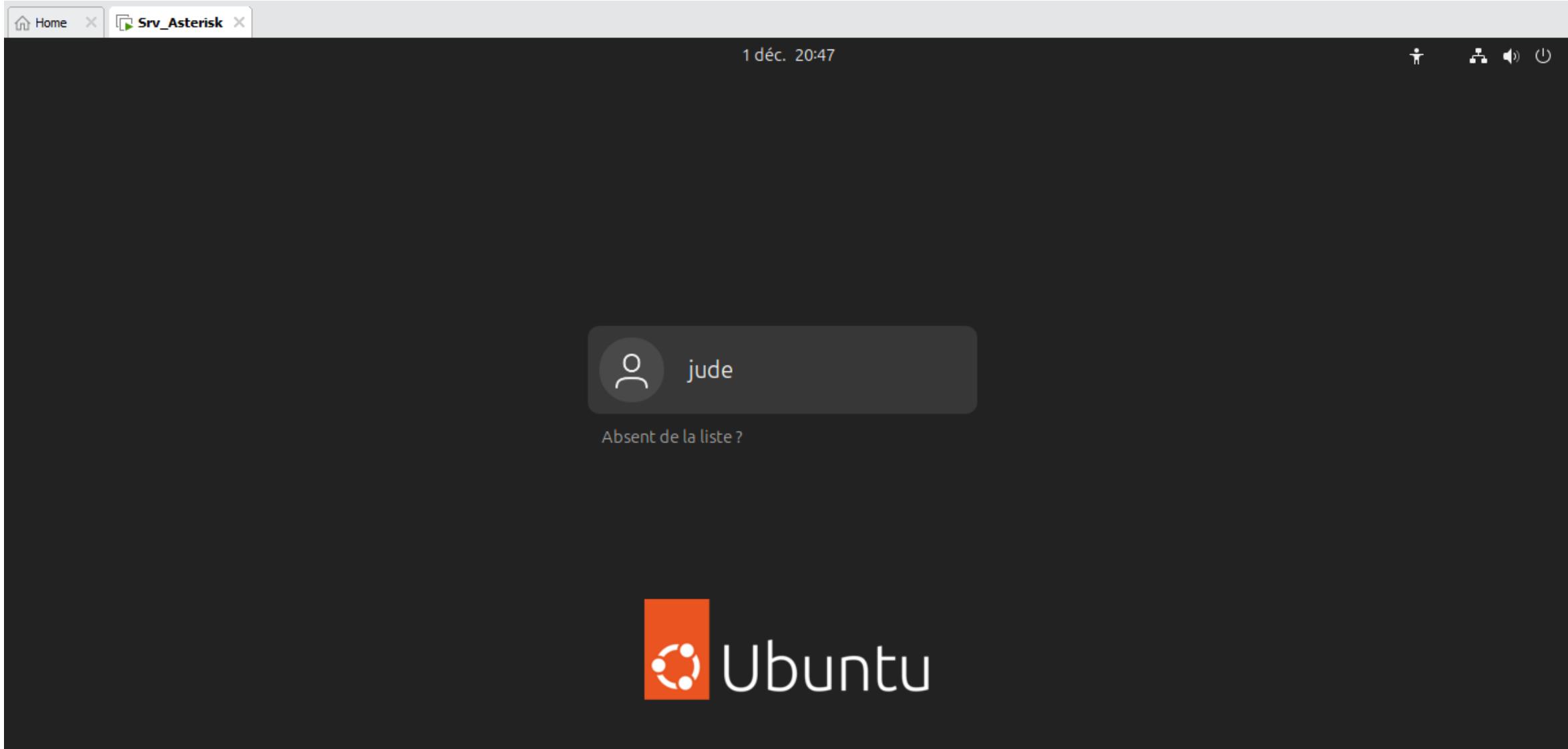
TOPOLOGIE



TRAVAUX PRATIQUE

MISE EN PLACE D'UN SERVEUR TELEPHONIE VOIP SOUS LINUX

- I. Montage d'un autocom serveur **ubuntu23.10 desktop**
 - Sur virtualisation de VMWare Workstation 17 créer des machines virtuelle pour la réalisation pratique
 - Créer la première: **Système d'exploitation ou iso ubuntu 23.10**
- I. Mettre l'application **Asterisk 19**
- II. On travaillera sur le switch **NAT** pour des raisons pratiques sur internet ou sinon le **BRIGE**
- III. Configuration des fichiers **SIP.CONF**; **EXTENSION.CONF** et **VOICEMAIL.CONF**
 - Plan de numérotation **1000 à 1004**
 - Faire la musique d'attente de votre choix
 - Faire l'**IVR** en adaptant nos réflexion dans cas de votre entreprise en prenant en compte au moins 4 services (**Informatique, Support, Gestion, Administratif**)



Screenshot of a Linux desktop environment showing the Virtual Network Editor and a DHCP Settings dialog.

The Virtual Network Editor window (highlighted with a red box) displays the following table:

Name	Type	External Connection	Host Connection	DHCP	Subnet Address
VMnet0	Bridged	Auto-bridging	-	-	-
VMnet1	Host-only	-	Connected	-	192.168.10.0
VMnet8	NAT	NAT	Connected	Enabled	192.168.8.0

Buttons at the bottom of the editor include: Add Network..., Remove Network, and Rename Network...; VMnet Information (radio buttons for Bridged, NAT, Host-only), Connect a host virtual adapter to this network (checkbox), and Use local DHCP service to distribute IP address to VMs (checkbox); and Subnet IP and Subnet mask fields (192.168.8.0 and 255.255.255.0).

The DHCP Settings dialog shows the following configuration for the vmnet8 network:

Network:	vmnet8
Subnet IP:	192.168.8.0
Subnet mask:	255.255.255.0
Starting IP address:	192.168.8.128
Ending IP address:	192.168.8.254
Broadcast address:	192.168.8.255
Default lease time:	Days: 0 Hours: 0 Minutes: 30
Max lease time:	Days: 0 Hours: 2 Minutes: 0

Buttons at the bottom of the dialog are: OK, Cancel, and Help.

❖ Se connecté en tant d'administrateur

sudo su

Mot de passe : xxxxxxxx

❖ Vérification de son adresse ip sur linux

ip a

❖ Test de connectivité

#ping 8.8.8.8

```
jude@jude:~$ sudo su
[sudo] Mot de passe de jude :
root@jude:/home/jude# ip a
1: lo: <LOOPBACK,UP,LOWER_UP> mtu 65536 qdisc noqueue state UP
    link/loopback 00:00:00:00:00:00 brd 00:00:00:00:00:00
        inet 127.0.0.1/8 scope host lo
            valid_lft forever preferred_lft forever
        inet6 ::1/128 scope host
            valid_lft forever preferred_lft forever
2: ens33: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc pf
    link/ether 00:0c:29:18:b3:cc brd ff:ff:ff:ff:ff:ff
    altname enp2s1
        inet 192.168.8.128/24 brd 192.168.8.255 scope global dynar
            valid_lft 1471sec preferred_lft 1471sec
        inet6 fe80::20c:29ff:fe18:b3cc/64 scope link
            valid_lft forever preferred_lft forever
```

```
root@jude:/home/jude# ping 8.8.8.8
PING 8.8.8.8 (8.8.8.8) 56(84) bytes of data.
64 bytes from 8.8.8.8: icmp_seq=1 ttl=128 time=15.4 ms
64 bytes from 8.8.8.8: icmp_seq=2 ttl=128 time=4.50 ms
64 bytes from 8.8.8.8: icmp_seq=3 ttl=128 time=5.48 ms
64 bytes from 8.8.8.8: icmp_seq=4 ttl=128 time=12.5 ms
64 bytes from 8.8.8.8: icmp_seq=5 ttl=128 time=4.99 ms
64 bytes from 8.8.8.8: icmp_seq=6 ttl=128 time=6.12 ms
64 bytes from 8.8.8.8: icmp_seq=7 ttl=128 time=4.79 ms
^C64 bytes from 8.8.8.8: icmp_seq=8 ttl=128 time=4.98 ms
64 bytes from 8.8.8.8: icmp_seq=9 ttl=128 time=11.3 ms
64 bytes from 8.8.8.8: icmp_seq=10 ttl=128 time=22.7 ms
64 bytes from 8.8.8.8: icmp_seq=11 ttl=128 time=15.8 ms
^C
```

❖ Mettre à jour :

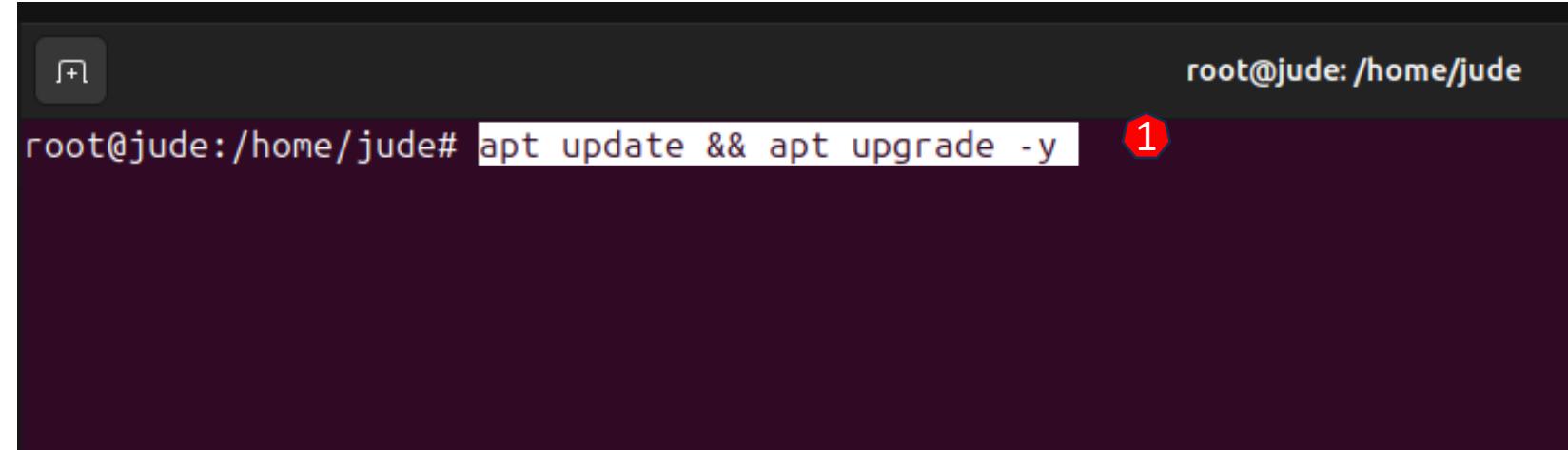
```
# apt update && apt upgrade -y
```

❖ Installer le ssh :

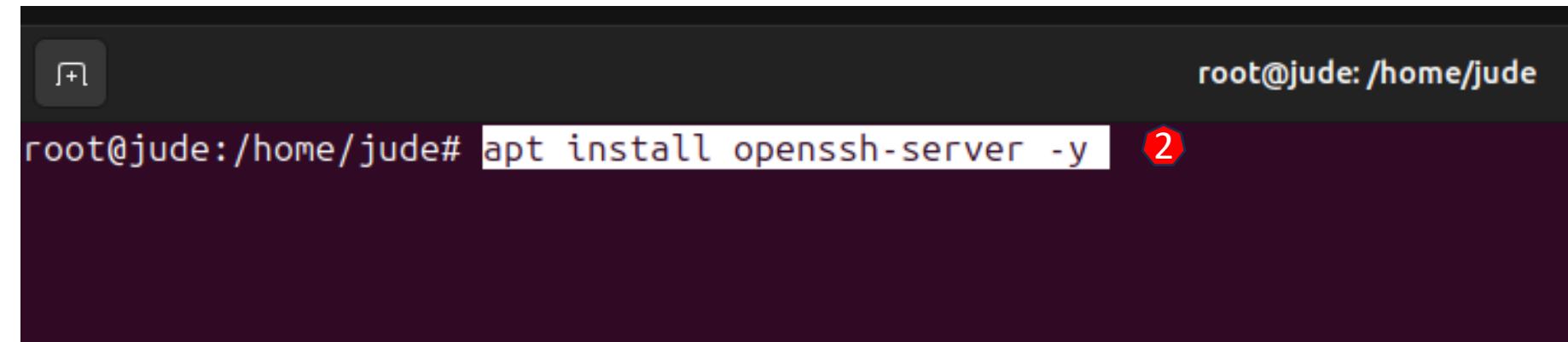
```
# apt install openssh-server -y
```

❖ Installer les tools :

```
#apt install open-vm-tools -y
```



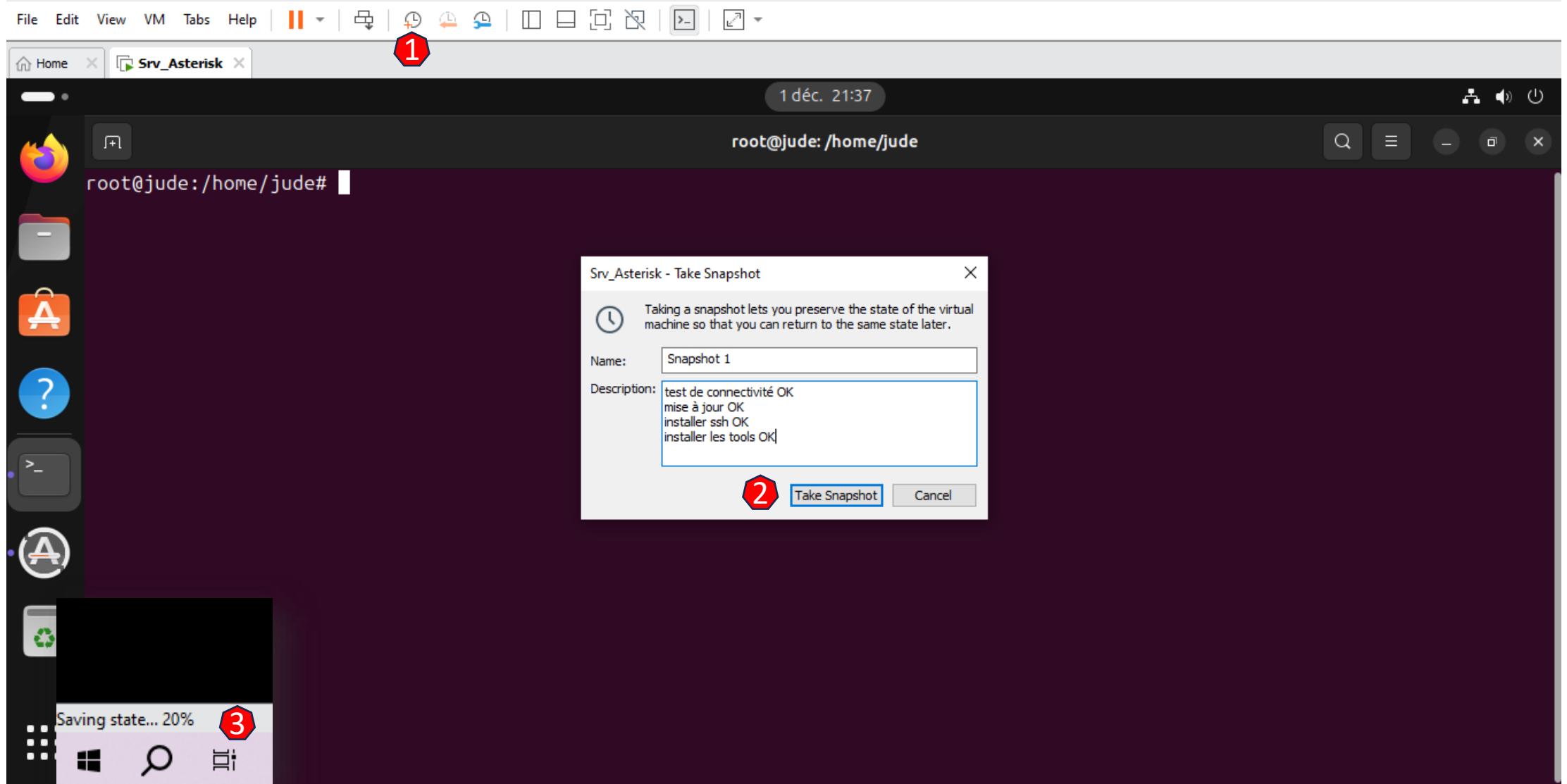
```
root@jude:/home/jude# apt update && apt upgrade -y 1
```



```
root@jude:/home/jude# apt install openssh-server -y 2
```

```
Traitement des actions différées (« triggers ») pour ufw (0.36.2-1) ...  
root@jude:/home/jude# apt install open-vm-tools -y 3
```

❖ Faire un Snapshot pour enregistrer le moment présent de la machine

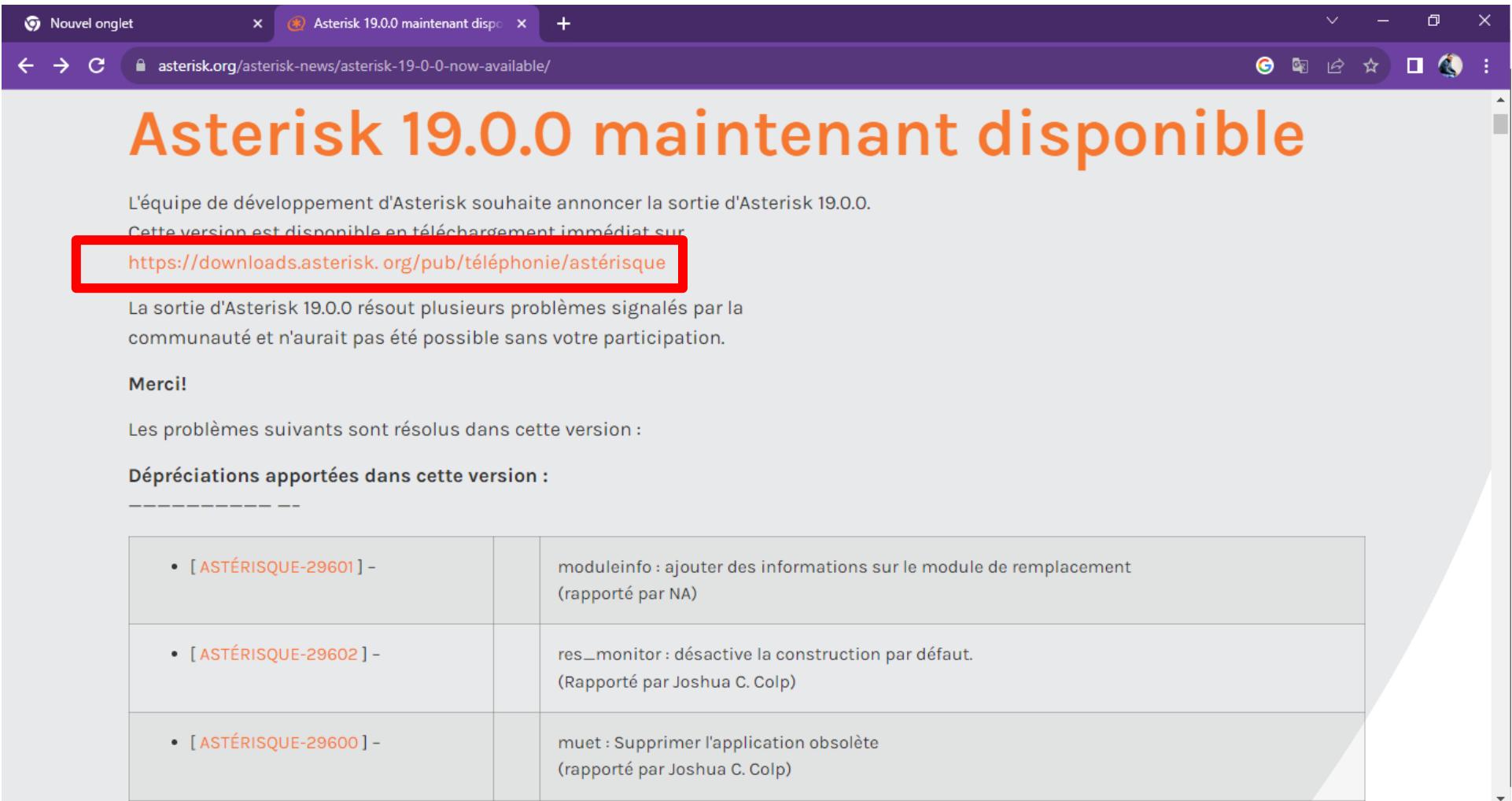


- ❖ Créer un dossier asterisk : # **mkdir -p /usr/src/asterisk**
(la commande mkdir nous permet de créer un repertoire)
- ❖ Se rendre dedans : # **cd /usr/src/astersik**
(chemin du dossier asterisk)
- ❖ On lui donne les droits maximums : # **chmod 777 -R /usr/src/asterisk**

```
root@jude: /home/jude# mkdri -p /usr/src/asterisk 1
root@jude: /home/jude# cd /usr/src/astersik
bash: cd: /usr/src/astersik: Aucun fichier ou dossier de ce type
root@jude: /home/jude# ls
Bureau Documents Images Modèles Musique Public snap Téléchargements Vidéos
root@jude: /home/jude# cd /usr/src/astersik
bash: cd: /usr/src/astersik: Aucun fichier ou dossier de ce type
root@jude: /home/jude# cd /usr/src/ 2
root@jude: /usr/src# ls
asterisk linux-headers-6.5.0-13 linux-headers-6.5.0-13-generic
root@jude: /usr/src# cd asterisk/ 3
root@jude: /usr/src/asterisk# chmod 777 -R /usr/src/asterisk 4
root@jude: /usr/src/asterisk#
```

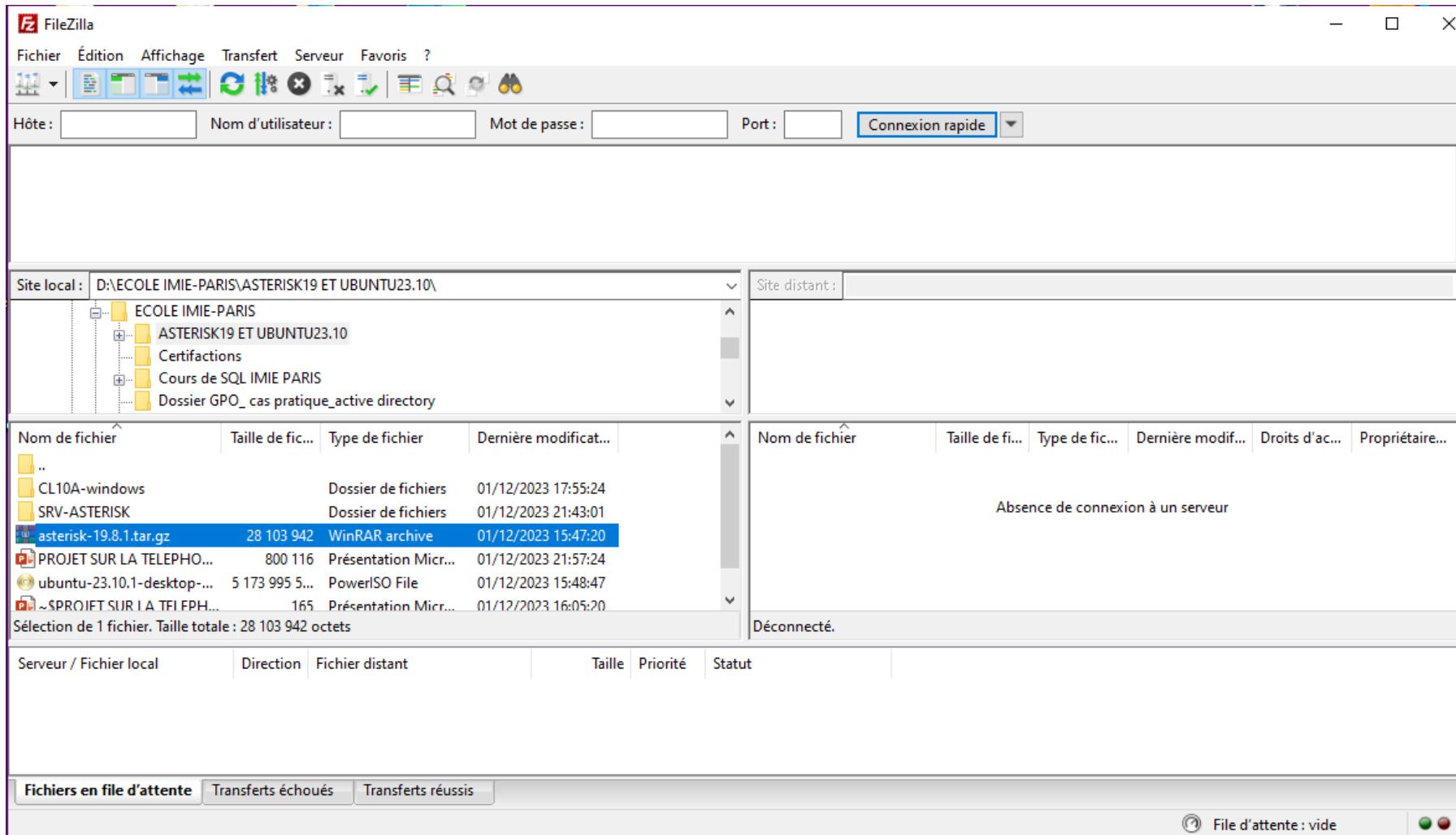
LANCER GOOGLE

Dans le site professionnel d'asterisk télécharger asterisk 19 et cliquer sur le lien en encadrer en rouge



asterisk-10.20.0-patch.sha256	18-Oct-2023 17:01	90
asterisk-18.20.0-patch.tar.gz	18-Oct-2023 17:01	63918
asterisk-18.20.0-patch.tar.gz.asc	18-Oct-2023 17:01	833
asterisk-18.20.0.md5	18-Oct-2023 17:01	58
asterisk-18.20.0.sha1	18-Oct-2023 17:01	66
asterisk-18.20.0.sha256	18-Oct-2023 17:01	90
asterisk-18.20.0.tar.gz	18-Oct-2023 17:01	28445590
asterisk-18.20.0.tar.gz.asc	18-Oct-2023 17:01	833
asterisk-19-current-patch.md5	07-Jul-2023 19:08	63
asterisk-19-current-patch.sha1	07-Jul-2023 19:08	71
asterisk-19-current-patch.sha256	07-Jul-2023 19:08	95
asterisk-19-current-patch.tar.gz	07-Jul-2023 19:08	186941
asterisk-19-current-patch.tar.gz.asc	07-Jul-2023 19:08	833
asterisk-19-current.md5	07-Jul-2023 19:08	57
asterisk-19-current.sha1	07-Jul-2023 19:08	65
asterisk-19-current.sha256	07-Jul-2023 19:08	89
asterisk-19-current.tar.gz	07-Jul-2023 19:08	28103942
asterisk-19-current.tar.gz.asc	07-Jul-2023 19:08	833
asterisk-19.8.1-patch.md5	07-Jul-2023 19:08	63
asterisk-19.8.1-patch.sha1	07-Jul-2023 19:08	71
asterisk-19.8.1-patch.sha256	07-Jul-2023 19:08	95
asterisk-19.8.1-patch.tar.gz	07-Jul-2023 19:08	186941
asterisk-19.8.1-patch.tar.gz.asc	07-Jul-2023 19:08	833
asterisk-19.8.1.md5	07-Jul-2023 19:08	57
asterisk-19.8.1.sha1	07-Jul-2023 19:08	65
asterisk-19.8.1.sha256	07-Jul-2023 19:08	89
asterisk-19.8.1.tar.gz	07-Jul-2023 19:08	28103942
asterisk-19.8.1.tar.gz.asc	07-Jul-2023 19:08	833
asterisk-20-current-patch.md5	18-Oct-2023 17:10	63
asterisk-20-current-patch.sha1	18-Oct-2023 17:10	71
asterisk-20-current-patch.sha256	18-Oct-2023 17:10	95
asterisk-20-current-patch.tar.gz	18-Oct-2023 17:10	63908
asterisk-20-current-patch.tar.gz.asc	18-Oct-2023 17:10	833
asterisk-20-current.md5	18-Oct-2023 17:10	57
asterisk-20-current.sha1	18-Oct-2023 17:10	65
asterisk-20-current.sha256	18-Oct-2023 17:10	89
asterisk-20-current.tar.gz	18-Oct-2023 17:10	28214209
asterisk-20-current.tar.gz.asc	18-Oct-2023 17:10	833
asterisk-20.5.0-patch.md5	18-Oct-2023 17:10	63
asterisk-20.5.0-patch.sha1	18-Oct-2023 17:10	71
asterisk-20.5.0-patch.sha256	18-Oct-2023 17:10	95
asterisk-20.5.0-patch.tar.gz	18-Oct-2023 17:10	63908
asterisk-20.5.0-patch.tar.gz.asc	18-Oct-2023 17:10	833
asterisk-20.5.0.md5	18-Oct-2023 17:10	57

LANCER LE FILEZILLA



LANCER LE FILEZILLA

Pour ce connecté à un serveur distant mettez l'adresse de ce serveur avec le nom de la machine et son mot de passe ainsi que le port 22



Fichier Édition Affichage Transfert Serveur Favoris ?



Hôte : sftp://192.168.8.128 Nom d'utilisateur : jude Mot de passe : **** Port : Connexion rapide

Statut : Récupération du contenu du dossier...
Statut : Listing directory /home/jude
Statut : Contenu du dossier « /home/jude » affiché avec succès
Statut : Récupération du contenu du dossier « / »...
Statut : Listing directory /
Statut : Contenu du dossier « / » affiché avec succès

Site local : D:\ECOLE IMIE-PARIS\ASTERISK19 ET UBUNTU23.10\

ECOLE IMIE-PARIS
ASTERISK19 ET UBUNTU23.10
Certifications
Cours de SQL IMIE PARIS
Dossier GPO_cas pratique_active directory

Site distant : /

bin
boot
cdrom
dev

Nom de fichier	Taille de fic...	Type de fichier	Dernière modifcat...
..			
CL10A-windows		Dossier de fichiers	01/12/2023 17:55:24
SRV-ASTERISK		Dossier de fichiers	01/12/2023 21:43:01
asterisk-19.8.1.tar.gz	28 103 942	WinRAR archive	01/12/2023 15:47:20
PROJET SUR LA TELEPHO...	800 116	Présentation Micr...	01/12/2023 21:57:24
ubuntu-23.10.1-desktop...	5 173 995 5...	PowerISO File	01/12/2023 15:48:47
~\$PROJET SUR LA TFI FPH...	165	Présentation Micr...	01/12/2023 16:05:20

Sélection de 1 fichier. Taille totale : 28 103 942 octets

Nom de fichier	Taille de fi...	Type de fic...	Dernière modif...	Droits d'ac...	Propriétaire...
..					
bin		Dossier de ...	16/10/2023 12:...	Irwxrwxrwx	root root
boot		Dossier de ...	01/12/2023 20:...	drwxr-xr-x	root root
cdrom		Dossier de ...	16/10/2023 13:...	dr-xr-xr-x	root root
dev		Dossier de ...	01/12/2023 20:...	drwxr-xr-x	root root
etc		Dossier de ...	01/12/2023 21:...	drwxr-xr-x	root root

1 fichier et 22 dossiers. Taille totale : 4 057 989 120 octets

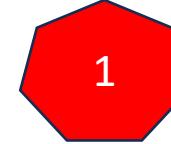
Serveur / Fichier local Direction Fichier distant Taille Priorité Statut

Fichiers en file d'attente Transferts échoués Transferts réussis

File d'attente : vide



Site distant : /



?	/					
?	bin					
?	boot					
?	cdrom					
?	dev					

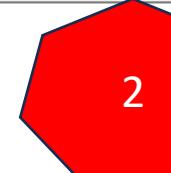
Nom de fichier	Taille de fi...	Type de fic...	Dernière modif...	Droits d'ac...	Propriétaire...
sys		Dossier de ...	01/12/2023 20:...	dr-xr-xr-x	root root
tmp		Dossier de ...	01/12/2023 21:...	drwxrwxrwt	root root
usr		Dossier de ...	16/10/2023 12:...	drwxr-xr-x	root root
var		Dossier de ...	16/10/2023 12:...	drwxr-xr-x	root root
swap.img	4 057 989 ...	Fichier d'i...	01/12/2023 20:...	-rw-----	root root

< >

Sélection de 1 dossier.

Depuis le site distant suive le chemin **/usr/scr/**

Site distant : /usr



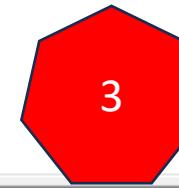
?	snap					
?	srv					
?	sys					
?	tmp					
+	usr					

Nom de fichier	Taille de fi...	Type de fic...	Dernière modif...	Droits d'ac...	Propriétaire...
libexec		Dossier de ...	01/12/2023 20:...	drwxr-xr-x	root root
local		Dossier de ...	16/10/2023 12:...	drwxr-xr-x	root root
sbin		Dossier de ...	01/12/2023 21:...	drwxr-xr-x	root root
share		Dossier de ...	01/12/2023 21:...	drwxr-xr-x	root root
src		Dossier de ...	01/12/2023 21:...	drwxr-xr-x	root root

< >

Sélection de 1 dossier.

Site distant : /usr/src



libexec
local
sbin
share
src

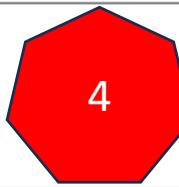
Nom de fichier Taille de fi... Type de fic... Dernière modif... Droits d'ac... Propriétaire...

..					
asterisk		Dossier de ...	01/12/2023 21:00:00	drwxrwxrwx	root root
linux-headers-6.5.0-13		Dossier de ...	01/12/2023 20:00:00	drwxr-xr-x	root root
linux-headers-6.5.0-1...		Dossier de ...	01/12/2023 20:00:00	drwxr-xr-x	root root

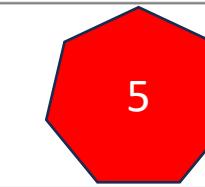
Sélection de 1 dossier.

Depuis le site distant suive le chemin /usr/src/asterisk

Site local : D:\ECOLE IMIE-PARIS\ASTERISK19 ET UBUNTU23.10\



Site distant : /usr/src/asterisk



share
src
asterisk
linux-headers-6.5.0-13
linux-headers-6.5.0-13-generic

Nom de fichier Taille de fi... Type de fichier Dernière modif...

..			
CL10A-windows		Dossier de fichiers	01/12/2023 17:55:24
SRV-ASTERISK		Dossier de fichiers	01/12/2023 21:43:01
asterisk-19.8.1.tar.gz	28 103 942	WinRAR archive	01/12/2023 15:47:20
PROJET SUR LA TELEPHONIE	800 116	Présentation Microsoft Word	01/12/2023 21:57:24
ubuntu-23.10.1-desktop-amd64.iso	5 173 995 5...	PowerISO File	01/12/2023 15:48:47
PROJET SUR LA TELEPHONIE	165	Présentation Microsoft Word	01/12/2023 16:05:20

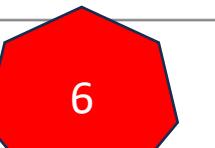
Sélection de 1 fichier. Taille totale : 28 103 942 octets

Nom de fichier Taille de fi... Type de fic... Dernière modif... Droits d'ac... Propriétaire...

..					

Ce dossier ne contient aucun élément

Dossier vide.

Site local :	D:\ECOLE IMIE-PARIS\ASTERISK19 ET UBUNTU23.10\	Site distant :	/usr/src/asterisk
	<ul style="list-style-type: none"> ... ECOLE IMIE-PARIS ASTERISK19 ET UBUNTU23.10 Certifications Cours de SQL IMIE PARIS Dossier GPO_cas pratique_active directory 		 <ul style="list-style-type: none"> ... ? share src <ul style="list-style-type: none"> asterisk ? linux-headers-6.5.0-13 ? linux-headers-6.5.0-13-generic
Nom de fichier	Taille de fic...	Type de fichier	Dernière modif...
..			
CL10A-windows		Dossier de fichiers	01/12/2023 17:55:24
SRV-ASTERISK		Dossier de fichiers	01/12/2023 21:43:01
asterisk-19.8.1.tar.gz	28 103 942	WinRAR archive	01/12/2023 15:47:20
PROJET SUR LA TELEPHO...	800 116	Présentation Micr...	01/12/2023 21:57:24
ubuntu-23.10.1-desktop...	5 173 995 ...	PowerISO File	01/12/2023 15:48:47
~SPROJET SUR LA TFI FPH...	165	Présentation Micr...	01/12/2023 16:05:20
Sélection de 1 fichier. Taille totale : 28 103 942 octets			
Sources / Fichiers local	Direction	Fichiers distants	Taille

ls

- ❖ On décomprime cette archive avec la commande :

```
# tar -xvzf asterisk-13.38.3.tar.gz
```

```
root@jude:/usr/src/asterisk# ls 1  
asterisk-19.8.1.tar.gz  
root@jude:/usr/src/asterisk# tar-xvzf asterisk-19.8.1.tar.gz  
tar-xvzf : commande introuvable  
root@jude:/usr/src/asterisk# tar -xvzf asterisk-19.8.1.tar.gz 2
```

```
# ls
```

- ❖ On se rend dans le dossier script du nouveau dossier
astersisk: # cd asterisk-13.38.3/contrib/scripts

```
root@jude:/usr/src/asterisk# ls ①
asterisk-19.8.1  asterisk-19.8.1.tar.gz
root@jude:/usr/src/asterisk# ll
total 27460
drwxrwxrwx  3 root root    4096 déc.   1 22:23 ②/
drwxr-xr-x  5 root root    4096 déc.   1 21:45 ../
drwxrwxr-x 32 root root    4096 juil.  7 21:07 asterisk-19.8.1/
-rw-rw-r--  1 jude jude 28103942 déc.   1 22:13 asterisk-19.8.1.tar.gz
root@jude:/usr/src/asterisk# cd asterisk-19.8.1/contrib/scripts/ ②
```

- ❖ On exécute le script fourni pour la préfabrication d'astersik : # ./install_prereq install

```
root@jude:/usr/src/asterisk/asterisk-19.8.1/contrib/scripts# ls ①
agents.php          ast_loggrabber      get_mp3_source.sh    README.messages-expire
astcli              ast_tls_cert       get_swagger_ui.sh  refcounter.py
ast_coredumper      astversion        import-cdr-csv-mysql.pl  reflocks.py
asterisk.ldap-schema autosupport      install_prereq     refstats.py
asterisk.ldif        autosupport.8    live_ast           retrieve_extensions_from_mysql.pl
asterisk.logrotate   clang-scan-build  loadtest.tcl      retrieve_extensions_from_sql.pl
astgenkey           dahdi_span_config_hook lookup.agi        retrieve_sip_conf_from_mysql.pl
astgenkey.8          dbsep.cgi         managerproxy.pl  safe_asterisk
ast_grab_core       file.convert.sh   messages-expire.pl safe_asterisk.8
ast_logescalator   get_ilbc_source.sh qview.pl        safe_asterisk_restore
root@jude:/usr/src/asterisk/asterisk-19.8.1/contrib/scripts# ./install_prereq install ②
```

sip_nat_settings
sipp-sendfax.xml
sip_to_pjsip
spandspflow2pcap.log
spandspflow2pcap.py
valgrind_compare
vmail.cgi
voicemailpwcheck.py

On retourne au niveau du dossier asterisk 19.8.1 en faisant la commande :

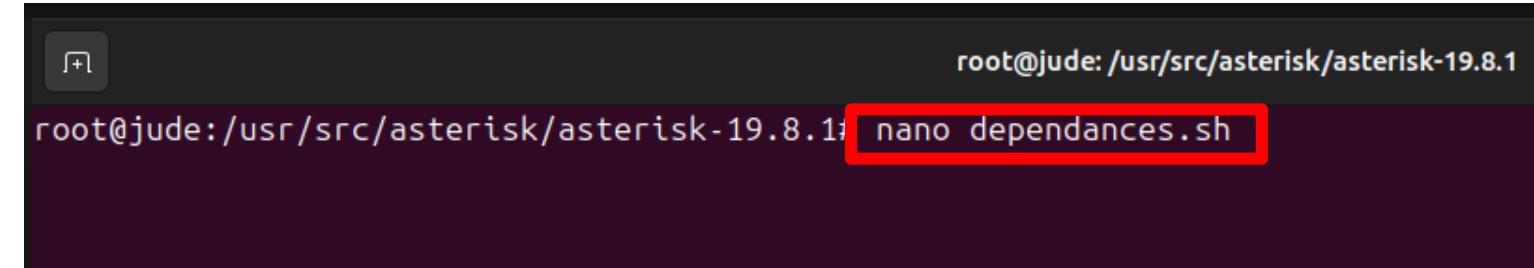
```
# cd.. (2 fois)  
# ls
```

```
#####
## install completed successfully
#####
root@jude:/usr/src/asterisk/asterisk-19.8.1/contrib/scripts# cd ..
root@jude:/usr/src/asterisk/asterisk-19.8.1/contrib# cd ..
root@jude:/usr/src/asterisk/asterisk-19.8.1# ls
addons      cel          contrib        LICENSE
agi         ChangeLogs   COPYING        main
apps         CHANGES.md   CREDITS       Makefile
autoconf    channels     default.exports Makefile.moddir_rules
bootstrap.sh codecs      doc           Makefile.rules
bridges     config.guess formats       makeopts.in
BSDmakefile configs     funcs         menuselect
BUGS        config.sub   images        missing
build_tools configure   include       mkinstalldirs
cdr          configure.ac install-sh pbx
root@jude:/usr/src/asterisk/asterisk-19.8.1#
```

On créer un fichier dépendance.sh : # **touch dependances.sh**

On lui donne des droits pour pouvoir l'exécuter : # **chmod 777 dependances.sh**

```
root@jude:/usr/src/asterisk/asterisk-19.8.1
root@jude:/usr/src/asterisk/asterisk-19.8.1# touch dependances.sh ①
root@jude:/usr/src/asterisk/asterisk-19.8.1# chmod 777 dependances.sh ②
root@jude:/usr/src/asterisk/asterisk-19.8.1# ls ③
addons      cel          contrib        install-sh      pbx
agi         ChangeLogs   COPYING        LICENSE        phoneprov
apps         CHANGES.md   CREDITS       main           README-addons.txt
autoconf    channels     default.exports Makefile        README.md
bootstrap.sh codecs      dependances.sh Makefile.moddir_rules README-SERIOUSLY.bestpractices.md
bridges     config.guess  doc           Makefile.rules  res
BSDmakefile  configs     formats       makeopts.in   rest-api
BUGS        config.sub   funcs        menuselect    rest-api-templates
build_tools  configure   images       missing        sample.call
cdr         configure.ac include     mkinstalldirs sounds
root@jude:/usr/src/asterisk/asterisk-19.8.1#
```



```
root@jude:/usr/src/asterisk/asterisk-19.8.1# nano dependances.sh
```

❖ Ensuite on l'édite : # **nano dependances.sh**

❖ Pour sauvegarder il faut maintenir
la touche Ctrl + la touche O
Ensuite sur la touche **Entrée**
Et pour quitter il faut maintenir la
touche Ctrl + la touche X



```
GNU nano 7.2                                     dependances.sh *
```

```
apt-get install linux-headers-4.9.0-6 -y
apt-get install build-essential -y
apt-get install libxml2-dev -y
apt-get install libncurses5-dev -y
apt-get install libreadline-dev -y
apt-get install libreadline6-dev -y
apt-get install libssl-dev -y
apt-get install uuid-dev -y
Aide apt-get install libjansson-dev -y
apt-get install libsdl1.2debian -y
apt-get install pkg-config -y
apt-get install perl -y
apt-get install libwww-perl -y
apt-get install sox -y
apt-get install mpg123 -y
apt-get install libedit-dev -y
apt-get install libedit2 -y
```

❖ Ensuite on l'édite : # nano dependances.sh

```
apt-get install linux-headers-4.9.0-6 -y
apt-get install build-essential -y
apt-get install libxml2-dev -y
apt-get install libncurses5-dev -y
apt-get install libreadline-dev -y
apt-get install libreadline6-dev -y
apt-get install libssl-dev -y
apt-get install uuid-dev -y
apt-get install libjansson-dev -y
apt-get install libsdl1.2-dev -y
apt-get install pkg-config -y
apt-get install perl -y
apt-get install libwww-perl -y
apt-get install sox -y
apt-get install mpg123 -y
apt-get install libedit-dev -y
apt-get install libedit2 -y
```

Une fois notre script fini on l'exécute : **# ./dependances.sh**
ls

Ensuite on exécute le script de configuration : **# ./configure**

```
root@jude: /usr/src/asterisk/asterisk-19.8.1
root@jude:/usr/src/asterisk/asterisk-19.8.1# ./dependances.sh
```

```
root@jude: /usr/src/asterisk/asterisk-19.8.1
root@jude:/usr/src/asterisk/asterisk-19.8.1# ls
addons      cel          contrib      install-sh      pbx
agi         ChangeLogs   COPYING      LICENSE        phoneprov
apps        CHANGES.md   CREDITS     main           README-addons.t
autoconf    channels     default.exports Makefile       README.md
bootstrap.sh codecs      dependances.sh 1 Makefile.moddir_rules README-SERIOUSL
bridges     config.guess doc          formats       Makefile.rules
BSDmakefile configs     func        funcs        makeopts.in
BUGS        config.sub   images      include      menuselect
build_tools configure   missing     mkinstalldirs
cdr         configure.ac include
```

root@jude:/usr/src/asterisk/asterisk-19.8.1# ./configure

Un logo devrait apparaître : il faut être en plein écran

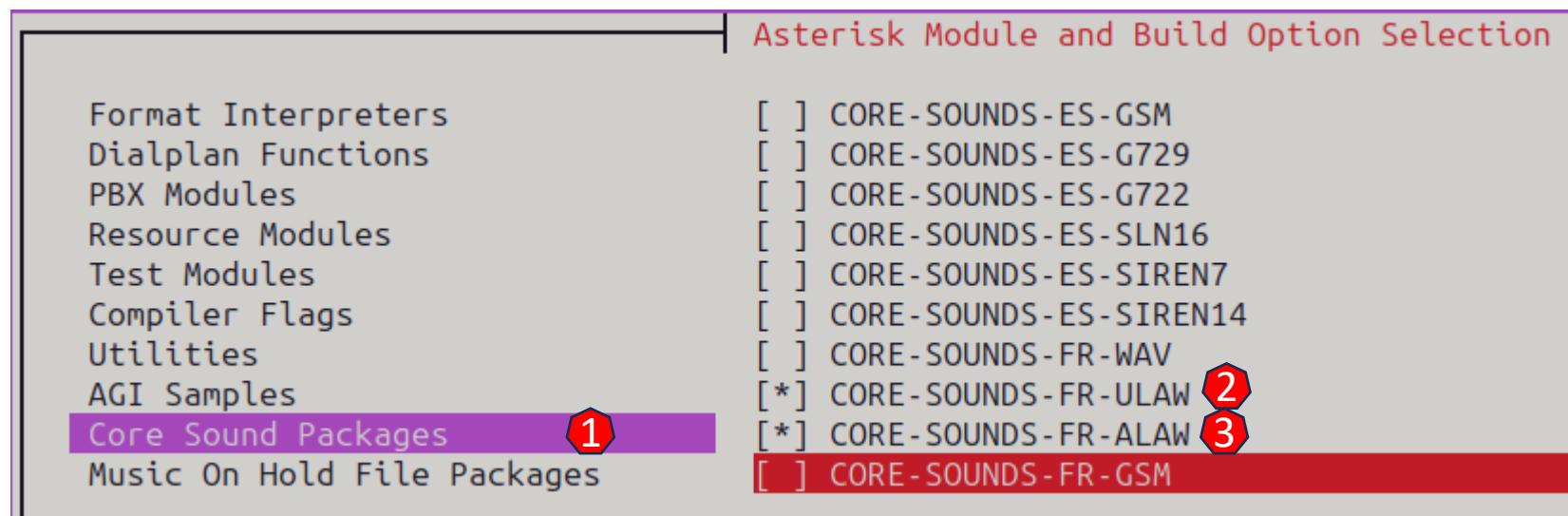
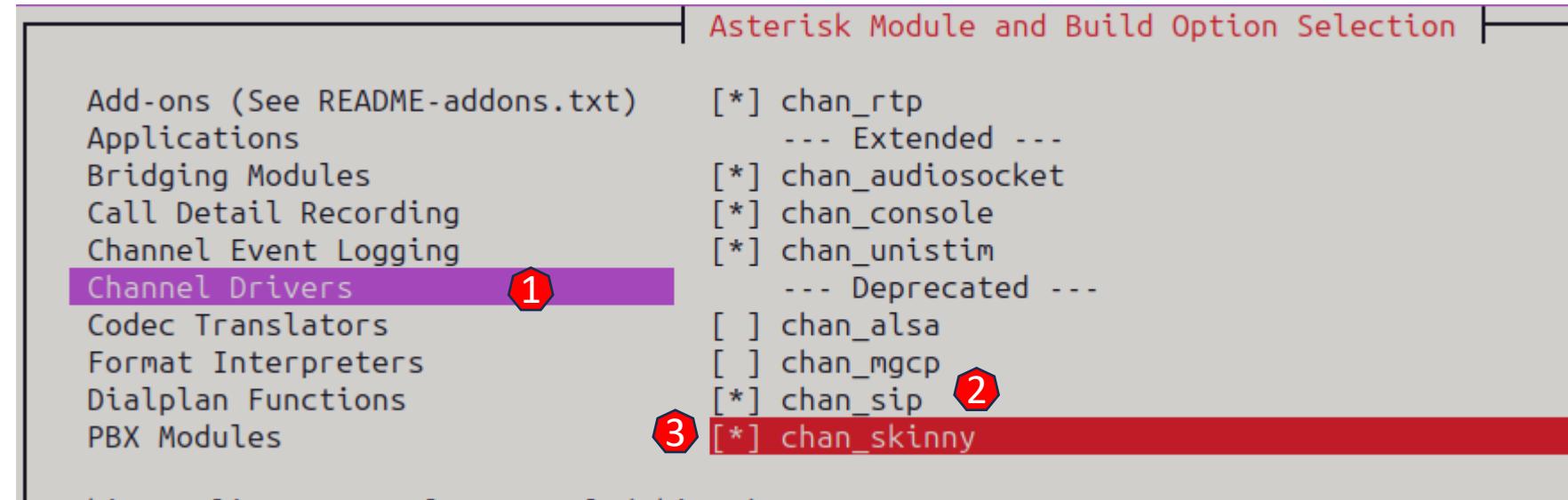
root@jude: /usr/src/asterisk/asterisk-19.8.1

```
config.status: creating autoconfig.h
configure: Menuselect build configuration successfully completed
```

```
.$$$$$$$$$$$$$=..
.$7$7.. .7$7:.
.$$:.. ,$7.7
.$7. 7$$$$ .$$77
..$$. $$$$ .$$$7
..7$ .?. $$$$ .?.. 7$$$. 
$.$. .$$$7. $$$7 .7$$. .$$$. 
.777. .$$$$$77$$$77$$$$$. $$$,
$$$~ .7$$$$$$$$$7. .$$$. 
.$$7 .7$$$$$7: ?$$. 
$$$ ?7$$$$$$$$$I .$$$7
$$$ .7$$$$$$$$$7. :$$. 
$$$ $$$7$$$$$7. .$$$. 
$$$ $$$ 7$$$7 .$$$ .$$$. 
$$$$ $$$7 .$$$ .$$$. 
7$$$7 7$$$ 7$$$
$$$$$ $$$
$$$$7. $$ (TM)
$$$$$$. .7$$$$$ $$
$$$$$7$$$$$7$$$$$7. $$$$.
$$$$$7$$$$$7. $$$$.
```

Nous allons ensuite configurer les options de son avec la commande :
make menuselect

```
root@jude:/usr/src/asterisk/asterisk-19.8.1# make menuselect
```



Music On Hold File Packages (on va décocher MOH-OP SOUND-S- WAV)

Asterisk Module and Build Option Selection

Format Interpreters	---	Core ---
Dialplan Functions	[]	MOH-OP SOUND-WAV
PBX Modules	[*]	MOH-OP SOUND-ULAW
Resource Modules	[*]	MOH-OP SOUND-ALAW
Test Modules	[]	MOH-OP SOUND-GSM
Compiler Flags	[]	MOH-OP SOUND-G729
Utilities	[]	MOH-OP SOUND-G722
AGI Samples	[]	MOH-OP SOUND-SLN16
Core Sound Packages	[]	MOH-OP SOUND-SIREN7
Music On Hold File Packages	[]	MOH-OP SOUND-SIREN14

opsound.org Music On Hold Files, a-Law format

Asterisk Module and Build Option Selection

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

French, GSM format

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

Save & Exit **Exit**

```
menuselect changes saved.
```

```
make[1] : on quitte le répertoire « /usr/src/asterisk/asterisk-19.8.1 »
root@jude:/usr/src/asterisk/asterisk-19.8.1# make install
```

❖Ensuite, pour compiler asterisk, on tape la commande : make

Puis on commence l'installation avec la commande : # **make install**

```
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
+
+----- OR -----
+
+ You can go ahead and install the asterisk
+ program documentation now or later run:
+
+         make progdocs
+
+ **Note** This requires that you have
```

Puis la commande : # **make config**

Puis la commande : # **make samples**

Et enfin la commande : # **make install-logrotate**

On se rend ensuite dans le répertoire d'asterisk où se trouvent les fichiers de configurations :

cd /etc/asterisk

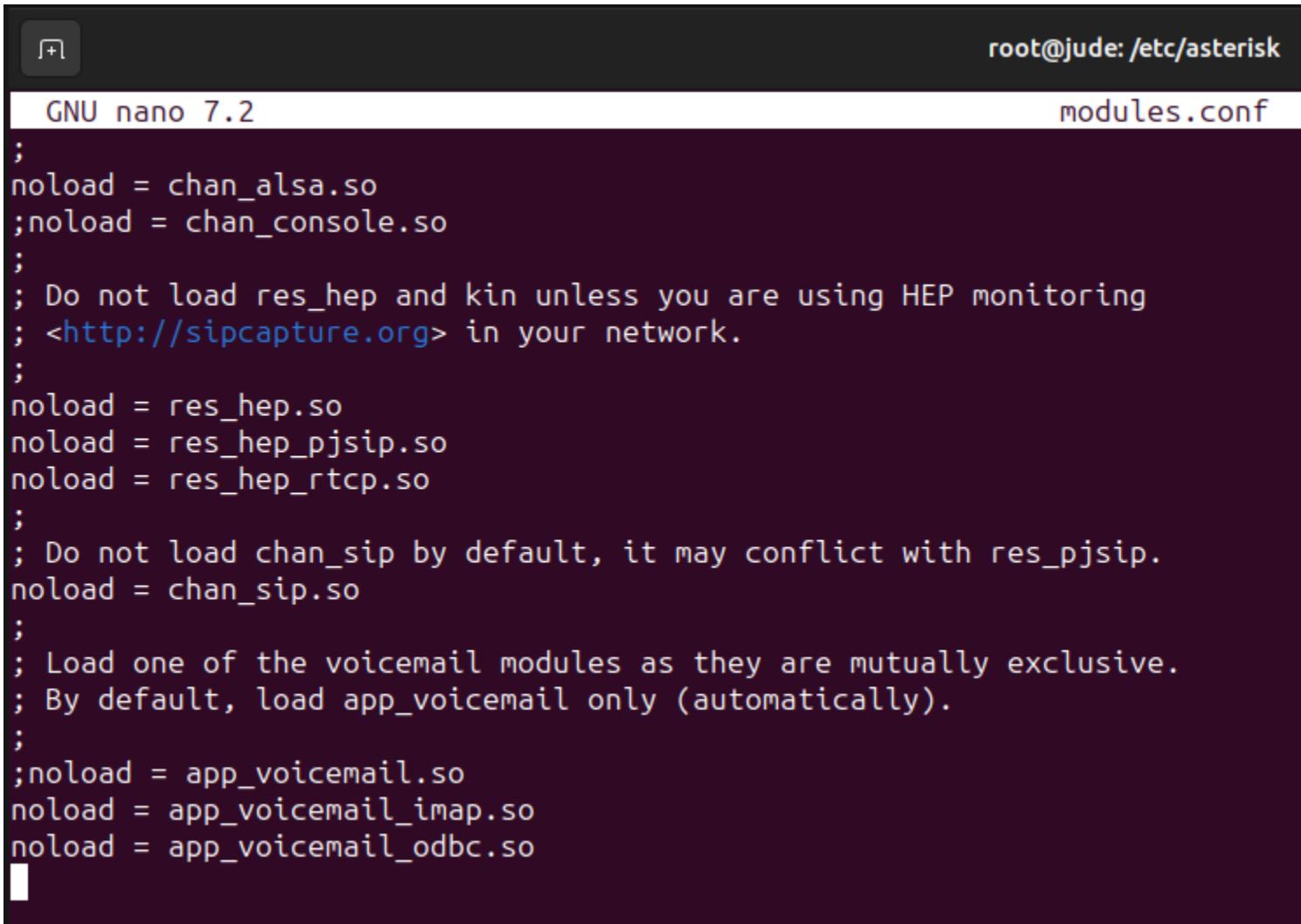
```
+ **Note** This requires that you have      +
+ doxygen installed on your local system      +
+-----+
root@jude:/usr/src/asterisk/asterisk-19.8.1# make config
```

```
Installing file phoneprov/000000000000.cfg
Installing file phoneprov/000000000000-directory.xml
Installing file phoneprov/000000000000-phone.cfg
Installing file phoneprov/polycom_line.xml
Installing file phoneprov/polycom.xml
Installing file phoneprov/snom-mac.xml
root@jude:/usr/src/asterisk/asterisk-19.8.1# make samples
```

```
root@jude:/usr/src/asterisk/asterisk-19.8. # cd /etc/asterisk/  
root@jude:/etc/asterisk# ls  
acl.conf          cdr_tds.conf      extconfig.conf    musiconhold.conf  res_snmp.conf  
adsi.conf         cel_beanstalkd.conf extensions.ael  ooh323.conf     res_stun_monitor.conf  
aeap.conf         cel.conf        extensions.conf  osp.conf       rtp.conf  
agents.conf       cel_custom.conf   extensions.lua   phoneprov.conf say.conf  
alarmreceiver.conf cel_odbc.conf   extensions_minivm.conf pjproject.conf sip.conf  
alsa.conf         cel_pgsql.conf  features.conf   pjsip.conf    sip_notify.conf  
amd.conf          cel_sqlite3_custom.conf festival.conf followme.conf skinny.conf  
app_skel.conf    cel_tds.conf    func_odbc.conf geolocation.conf sla.conf  
ari.conf          chan_dahdi.conf cli_aliases.conf hep.conf     smdi.conf  
ast_debug_tools.conf chan_mobile.conf cli.conf      http.conf    sorcery.conf  
asterisk.adsi    cli_permissions.conf codecs.conf  iax.conf      ss7.timers  
asterisk.conf    configbridge.conf confbridge.conf iaxprov.conf stasis.conf  
calendar.conf   config_test.conf  config_test.conf indications.conf statsd.conf  
ccss.conf         console.conf    dbsep.conf     logger.conf  stir_shaken.conf  
cdr_adaptive_odbc.conf config_test.conf  dnsmgr.conf  manager.conf telcordia-1.adsi  
cdr_beanstalkd.conf config_test.conf  dsp.conf     meetme.conf test_sorcery.conf  
cdr.conf          config_test.conf  dundi.conf    mgcp.conf    udptl.conf  
cdr_custom.conf   config_test.conf  enum.conf     modules.conf unistim.conf  
cdr_manager.conf  config_test.conf  motit.conf    motit.conf  users.conf  
cdr_odbc.conf    config_test.conf  motit.conf    motit.conf  voicemail.conf  
cdr_pgsql.conf   config_test.conf  motit.conf    motit.conf  xmpp.conf  
cdr_sqlite3_custom.conf config_test.conf  motit.conf    motit.conf  
root@jude:/etc/asterisk#
```

```
root@jude:/etc/asterisk# nano modules.conf
```

- ❖ On édite ensuite le fichier module.conf : # nano modules.conf
- ❖ On ajoute ces deux lignes à la fin du fichier : « load => chan_sip »
« load => chan_sip.so »



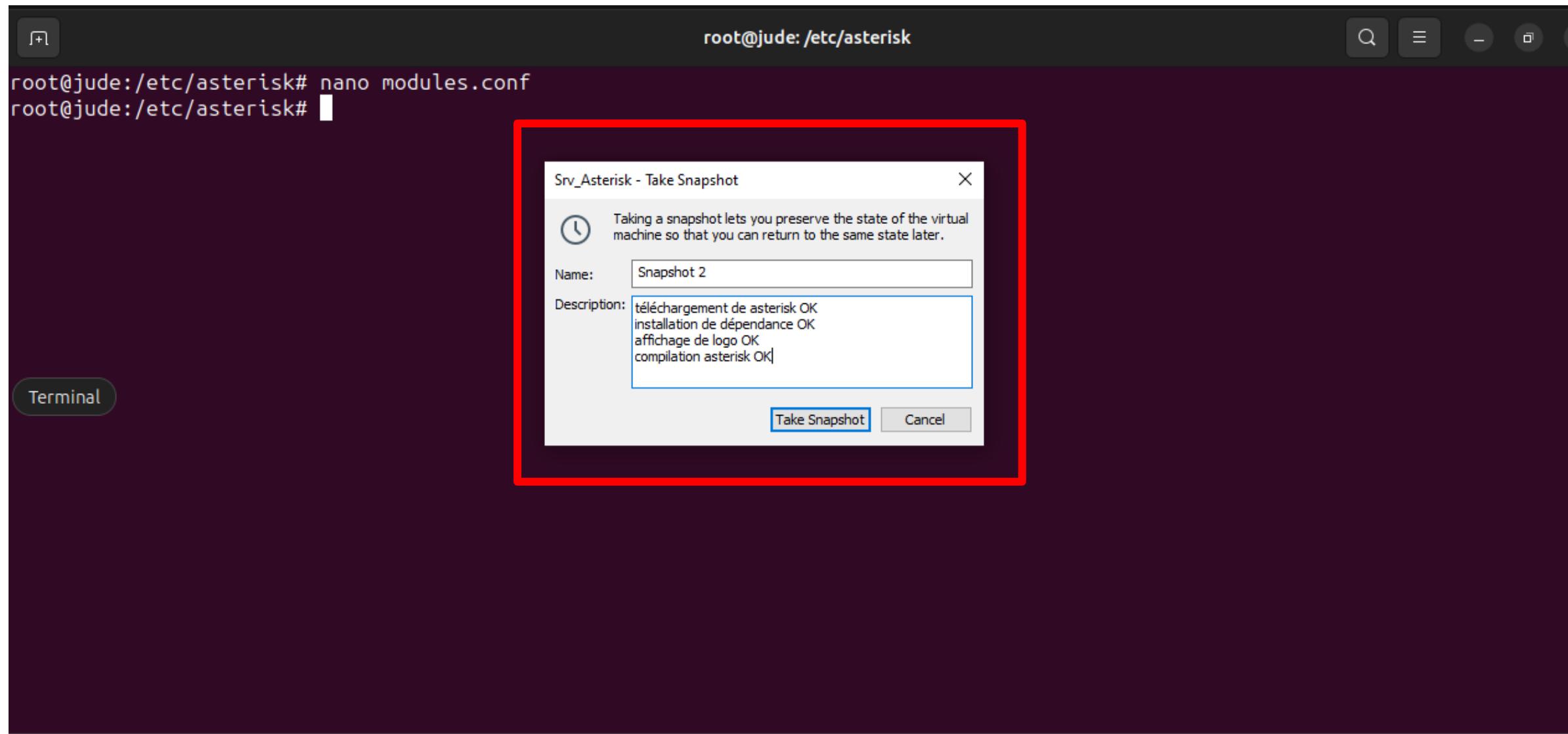
The screenshot shows a terminal window with the title "GNU nano 7.2" and the path "root@jude: /etc/asterisk". The file being edited is "modules.conf". The content of the file is as follows:

```
;;
noload = chan_alsa.so
;noload = chan_console.so
;
; Do not load res_hep and kin unless you are using HEP monitoring
; <http://sipcapture.org> in your network.
;
noload = res_hep.so
noload = res_hep_pjsip.so
noload = res_hep_rtcp.so
;
; Do not load chan_sip by default, it may conflict with res_pjsip.
noload = chan_sip.so
;
; Load one of the voicemail modules as they are mutually exclusive.
; By default, load app_voicemail only (automatically).
;
;noload = app_voicemail.so
noload = app_voicemail_imap.so
noload = app_voicemail_odbc.so
```

- ❖ On édite ensuite le fichier module.conf : # **nano modules.conf**
- ❖ On ajoute ces deux lignes à la fin du fichier : « load => chan_sip » et « load => chan_sip.so »
- ❖ Enregistrer **CTRL+O Entrer / Quitter CTRL+X**

```
root@jude: /etc/asterisk
GNU nano 7.2                               modules.conf *
;
; Do not load res_hep and kin unless you are using HEP monitoring
; <http://sipcapture.org> in your network.
;
noload = res_hep.so
noload = res_hep_pjsip.so
noload = res_hep_rtcp.so
.
load => chan_sip
load => chan_sip.so
;
; Do not load chan_sip by default, it may conflict with res_pjsip.
noload = chan_sip.so
;
; Gestionnaire de mises à jour cemail modules as they are mutually exclusive.
; by default, load app_voicemail only (automatically).
;
;noload = app_voicemail.so
noload = app_voicemail_imap.so
noload = app_voicemail_odbc.so
```

❖ Faire un Snapshot pour enregistrer le moment présent de la machine



Nous allons créer un utilisateur et un groupe disposant des droits nécessaires à l'exécution d'Asterisk avec les commandes suivantes :

- ❖ **groupadd asterisk**
- ❖ **useradd -r -d /var/lib/asterisk -g asterisk asterisk**
- ❖ **usermod -aG audio,dialout asterisk**
- ❖ **chown -R asterisk /etc/asterisk**
- ❖ **chown -R asterisk /var/{lib,log,spool}/asterisk**
- ❖ **chown -R asterisk /usr/lib/asterisk**
- ❖ **chmod -R 777 /var/{lib,log,run,spool}/asterisk /usr/lib/asterisk /etc/asterisk/**

```
root@jude: /etc/asterisk# groupadd asterisk
root@jude: /etc/asterisk# useradd -r -d /var/lib/asterisk -g asterisk asterisk
root@jude: /etc/asterisk# usermod -aG audio,dialout asterisk
root@jude: /etc/asterisk# chown -R asterisk /etc/asterisk
root@jude: /etc/asterisk# chown -R asterisk /var/{lib,log,spool}/asterisk
root@jude: /etc/asterisk# chown -R asterisk /usr/lib/asterisk
root@jude: /etc/asterisk# chmod -R 777 /var/{lib,log,run,spool}/asterisk /usr/lib/asterisk /etc/asterisk/
chmod: impossible d'accéler à '/usr/lib/asterisk/etc/asterisk': Aucun fichier ou dossier de ce type
root@jude: /etc/asterisk# chmod -R 777 /var/{lib,log,run,spool}/asterisk /usr/lib/asterisk /etc/asterisk/
root@jude: /etc/asterisk#
```

- ❖ Editer le fichier asterisk
- # **nano /etc/default/asterisk**

- ❖ Et on décommente les lignes suivantes :
- AST_USER= « asterisk »
- AST_GROUP= « asterisk »

```
root@jude:/etc/asterisk: nano /etc/default/asterisk
```

```
GNU nano 7.2 /etc/default/asterisk
# Startup configuration for the Asterisk daemon

# Uncomment the following and set them to the user/groups that you
# want to run Asterisk as. NOTE: this requires substantial work to
# be sure that Asterisk's environment has permission to write the
# files required for its operation, including logs, its comm
# socket, the asterisk database, etc.
#AST_USER="asterisk"
#AST_GROUP="asterisk"
```

```
GNU nano 7.2 /etc/default/asterisk
# Startup configuration for the Asterisk daemon

# Uncomment the following and set them to the user/groups that you
# want to run Asterisk as. NOTE: this requires substantial work to
# be sure that Asterisk's environment has permission to write the
# files required for its operation, including logs, its comm
# socket, the asterisk database, etc.

AST_USER="asterisk"
AST_GROUP="asterisk"
```

- ❖ On édite également le fichier asterisk.conf : **nano /etc/asterisk/asterisk.conf**

- ❖ Et on décommente les lignes suivantes :

runuser = asterisk (Enlever la virgule devant)

rungroup = asterisk (Enlever la virgule devant)

```
root@jude:/etc/asterisk# nano asterisk.conf  
root@jude:/etc/asterisk# nano asterisk.conf
```

1

GNU nano 7.2

;record_cache_dir = /tmp

;transmit_silence = yes

;transcode_via_sln = yes

runuser = asterisk

rungroup = asterisk

;lightbackground = yes

;forceblackbackground = yes

root@jude:/etc/asterisk

asterisk.conf *

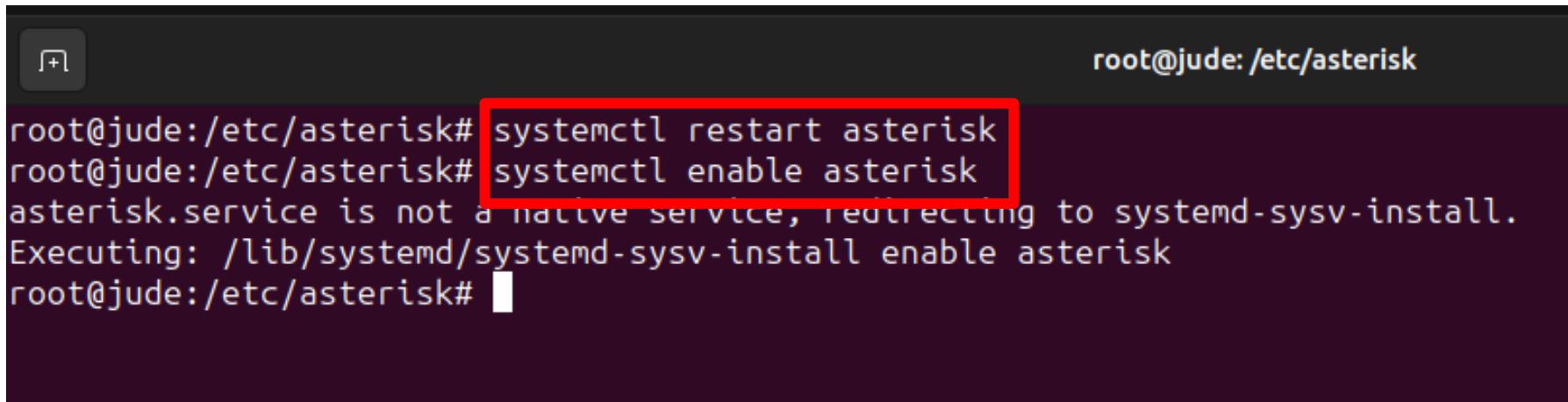
; directory during recording.
; Specify cache directory (used in conjunction
; with cache_record_files).
; Transmit silence while a channel is in a
; waiting state, a recording only state, or
; when DTMF is being generated. Note that the
; silence internally is generated in raw signed
; linear format. This means that it must be
; transcoded into the native format of the
; channel before it can be sent to the device.
; It is for this reason that this is optional,
; as it may result in requiring a temporary
; codec translation path for a channel that may
; not otherwise require one.
; Build transcode paths via SLINEAR, instead of
; directly.
; The user to run as.
; The group to run as.
; If your terminal is set for a light-colored
; background.
; Force the background of the terminal to be

2

❖ Pour redémarrer le service faire la commande suivante :

systemctl restart asterisk

systemctl enable asterisk



A screenshot of a terminal window titled "root@jude: /etc/asterisk". The terminal shows the following command sequence:

```
root@jude:/etc/asterisk# systemctl restart asterisk
root@jude:/etc/asterisk# systemctl enable asterisk
asterisk.service is not a native service, redirecting to systemd-sysv-install.
Executing: /lib/systemd/systemd-sysv-install enable asterisk
root@jude:/etc/asterisk#
```

The first two commands, "systemctl restart asterisk" and "systemctl enable asterisk", are highlighted with a red rectangular box.

PROCHAINE ETAPE ACTIVER LA CONSOLE ASTERISK

On active la console asterisk, pour vérifier que tout fonctionne bien avec la commande :

asterisk -rvvvv

(il faut parfois plusieurs essais ou un redémarrage des services asterisk)

Ensuite on fait : > **exit** pour quitter la console.

```
root@jude:/etc/asterisk# asterisk -rvvvv
Asterisk 19.8.1, Copyright (c) 1999 - 2022, 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.
=====
Running as user 'asterisk'
Running under group 'asterisk'
Connected to Asterisk 19.8.1 currently running on jude (pid = 41239)
jude*CLI> exit|
```

Nous allons ensuite faire un copie des fichier de configuration de base d'asterisk avant de les modifier avec les commandes suivantes :

- ❖ **mv sip.conf sip.conf.old**
- ❖ **mv extensions.conf exentions.conf.old**
- ❖ **mv voicemail.conf voicemail.conf.old**

```
Connected to Asterisk 19.8.1 currently running on jude (pid = 41239)
jude*CLI> exit
Asterisk cleanly ending (0).
Executing last minute cleanups
root@jude:/etc/asterisk# mv sip.conf sip.conf.old
root@jude:/etc/asterisk# mv extensions.conf exentions.conf.old
root@jude:/etc/asterisk# mv voicemail.conf voicemail.conf.old
root@jude:/etc/asterisk#
```

Il faut ensuite modifier les trois fichiers en fonction des besoins de l'entreprise.



extensions.conf
(1).txt



SIP.CONF.txt



voicemail.conf.tx
t



SIP.CONF.txt

nano sip.conf

```
[general]
context=default
bindport=5060
bindaddr=0.0.0.0
language=fr
disallow=all
allow=ulaw
allow=alaw

#register => B_A:azerty@192.168.8.134

[1000]
secret=1000
type=friend
host=dynamic
callerid="Jude" <1000>
mailbox=1000@default

[1001]
secret=1001
type=friend
host=dynamic
callerid=<< Vincent" <1001>
mailbox=1001@default

[1002]
secret=1002
type=friend
host=dynamic
callerid=<< Nerimene" <1002>
mailbox=1002@default

[1003]
secret=1003
type=friend
host=dynamic
callerid=<< Yassine" <1003>
mailbox=1003@default

[1004]
secret=1004
type=friend
host=dynamic
callerid=<< Alamindi" <1004>
mailbox=1004@default

#[A_B]
#type=friend
#secret=azerty
#context=default
#host=dynamic
#insecure=port,invite
```



root@jude: /etc/asterisk

root@jude:/etc/asterisk# nano sip.conf



root@jude: /etc/asterisk



GNU nano 7.2

sip.conf *

[1000]
secret=1000
type=friend
host=dynamic
callerid="Jude" <1000>
mailbox=1000@default

[1001]
secret=1001
type=friend
host=dynamic
callerid="Vincent" <1001>
mailbox=1001@default

[1002]
secret=1002
type=friend
host=dynamic
callerid="Nerimene" <1002>
mailbox=1002@default

^G Aide
^X Quitter

^O Écrire
^R Lire fich.

^W Chercher
^\\ Remplacer

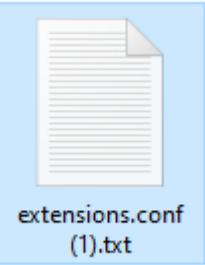
^K Couper
^U Coller

^T Exécuter
^J Justifier

^C Emplacement
^/ Aller ligne

M-U Annuler
M-E Refaire

M-A Marquer
M-6 Copier



[default]

```
exten => 1000,1,Answer()
exten => 1000, n,Dial(Sip/1000,10)
exten => 1000,n,VoiceMail(1000)
exten => 1000,n,Hangup()
```

nano extensions.conf

```
exten => 1001,1,Answer()
exten => 1001,n,Dial(Sip/1001,10)
exten => 1001,n,VoiceMail(1001)
exten => 1001,n,Hangup()
```

```
exten => 1002,1,Answer()
exten => 1002,n,Dial(Sip/1002,10)
exten => 1002,n,VoiceMail(1002)
exten => 1002,n,Hangup()
```

```
exten => 1003,1,Answer()
exten => 1003,n,Dial(Sip/1003,10)
exten => 1003,n,VoiceMail(1003)
exten => 1003,n,Hangup()
```

```
exten => 1004,1,Answer()
exten => 1004,n,Dial(Sip/1004,10)
exten => 1004,n,VoiceMail(1004)
exten => 1004,n,Hangup()
```

```
#exten => _2XXX,1,Dial(Sip/A_B/${EXTEN})
exten => 500,1,VoiceMailMain()
```

root@jude: /etc/asterisk

```
root@jude:/etc/asterisk# nano sip.conf  
root@jude:/etc/asterisk# nano extensions.conf
```

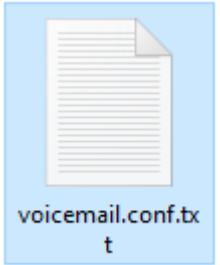
root@jude: /etc/asterisk

```
GNU nano 7.2                                         extensions.conf *
```

[default]

```
exten => 1000,1,Answer()  
exten => 1000, n,Dial(Sip/1000,10)  
exten => 1000,n,VoiceMail(1000)  
exten => 1000,n,Hangup()  
  
exten => 1001,1,Answer()  
exten => 1001,n,Dial(Sip/1001,10)  
exten => 1001,n,VoiceMail(1001)  
exten => 1001,n,Hangup()  
  
exten => 1002,1,Answer()  
exten => 1002,n,Dial(Sip/1002,10)  
exten => 1002,n,VoiceMail(1002)  
exten => 1002,n,Hangup()  
  
exten => 1003,1,Answer()  
exten => 1003,n,Dial(Sip/1003,10)  
exten => 1003,n,VoiceMail(1003)  
exten => 1003,n,Hangup()
```

^G Aide ^O Écrire ^W Chercher ^K Couper ^T Exécuter ^C Emplacement M-U Annuler M-A Marquer
^X Quitter ^R Lire fich. ^\ Remplacer ^U Coller ^J Justifier ^/ Aller ligne M-E Refaire M-6 Copier



voicemail.conf.txt

```
# nano voicemail.conf
```

[default]

```
1000 => 1000,Jude,1000@192.168.8.128
1001 => 1001,Vincent,1001@192.168.8.128
1002 => 1002,Nerimene,1002@192.168.8.128
1003 => 1003,Yassine,1003@192.168.8.128
1004 => 1004,Alamindi,1004@192.168.8.128
```

L'adresse ip
192.168.8.128
appartient au
serveur
téléphonie

L'ID 1001 est
l'identifiant
attribuer un
autocom

Nerimene est le
nom attribuer à
l'autocom pour
l'utilisateur

root@jude: /etc/asterisk

```
root@jude:/etc/asterisk# nano sip.conf
root@jude:/etc/asterisk# nano extensions.conf
root@jude:/etc/asterisk# nano voicemail.conf
```

root@jude: /etc/asterisk

```
GNU nano 7.2                                     voicemail.conf *
[default]
1000 => 1000,Jude,1000@192.168.8.128
1001 => 1001,Vincent,1001@192.168.8.128
1002 => 1002,Nerimene,1002@192.168.8.128
1003 => 1003,Yassine,1003@192.168.8.128
1004 => 1004,Alamindi,1004@192.168.8.128
```

^G Aide
^X Quitter

^O Écrire
^R Lire fich.

^W Chercher
^V Remplacer

^K Couper
^U Coller

^T Exécuter
^J Justifier

^C Emplacement
^/ Aller ligne

M-U Annuler
M-E Refaire

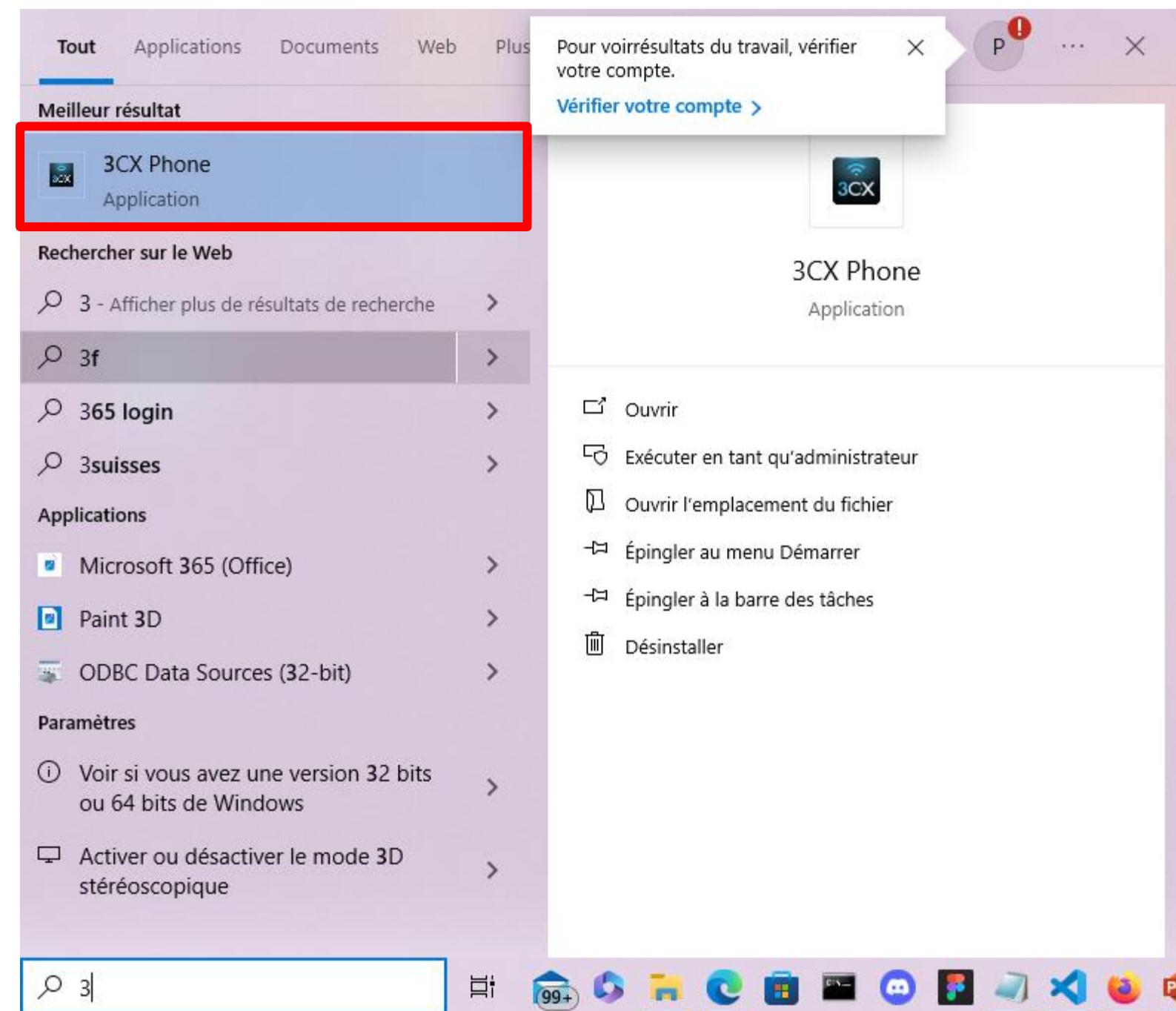
M-A Marquer
M-6 Copier

- ❖ On redémarre asterisk : /etc/init.d/asterisk restart
- ❖ On rouvre la console asterisk : **astersik -rvvvv**

```
root@jude:/etc/asterisk
root@jude:/etc/asterisk# /etc/init.d/asterisk restart
Restarting asterisk (via systemctl): asterisk.service.
root@jude:/etc/asterisk# asterisk -rvvvv
Unable to connect to remote asterisk (does /var/run/asterisk/asterisk.ctl exist?)
root@jude:/etc/asterisk# systemctl restart asterisk
root@jude:/etc/asterisk# systemctl enable asterisk
asterisk.service is not a native service, redirecting to systemd-sysv-install.
Executing: /lib/systemd/systemd-sysv-install enable asterisk
root@jude:/etc/asterisk# asterisk -rvvvv
Asterisk 19.8.1, Copyright (C) 1999 - 2022, 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.
=====
Running as user 'asterisk'
Running under group 'asterisk'
Connected to Asterisk 19.8.1 currently running on jude (pid = 42032)
jude*CLI>
```

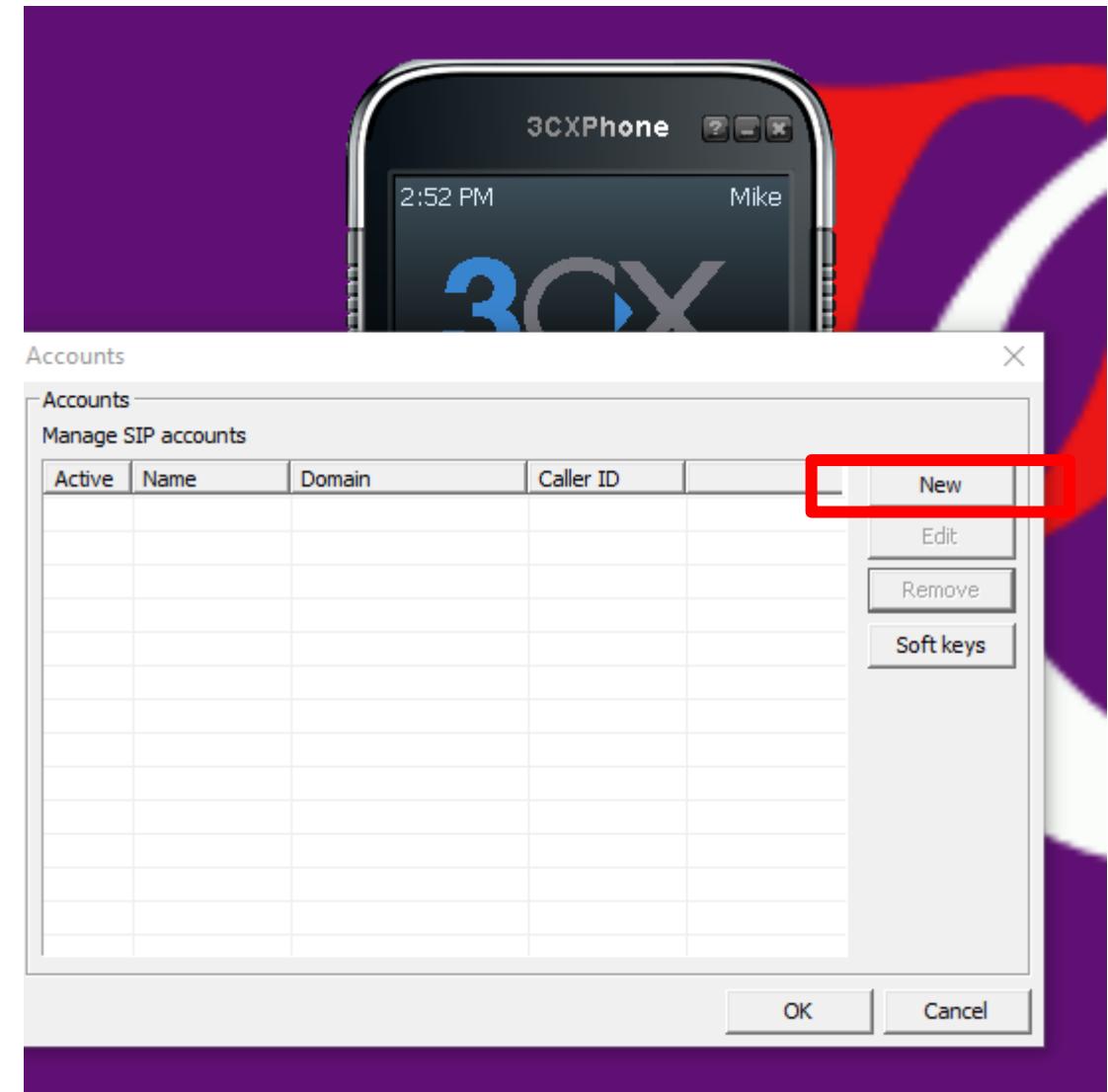
LANCER L'APPLICATION 3CX Phone

Si vous avez pas l'application veuillez le télécharger via internet



REGISTRE DES UTILISATEUR

A ce niveau, depuis le 3CXPhone, nous allons enregistrer tous les utilisateurs en le configurant à ce qu'il soit opérationnel.



The screenshot shows the 3CXPhone software interface. In the foreground, a modal dialog titled "Account settings" is open, overlaid on the main "Accounts" window. The "Accounts" window displays a list of SIP accounts with columns for Active, Name, Domain, and Caller ID. Three accounts are listed: Jude (Active, Name: Jude, Domain: 1000@192.168.8.128, Caller ID: 1000), Vincent (Active, Name: Vincent, Domain: 1001@192.168.8.128, Caller ID: 1001), and Nerimene (Active, Name: Nerimene, Domain: 1002@192.168.8.128, Caller ID: 1002). The "Account settings" dialog contains fields for Account name (Jude), Caller ID (1000), Extension (1000), ID (1000), Password (****), and My location (IP of PBX/SIP server set to 192.168.8.128). It also includes checkboxes for "Use 3CX Tunnel" (unchecked) and "Use Outbound Proxy server" (unchecked), and fields for Local IP of remote PBX (192.168.8.128) and Tunnel password (***). At the bottom of the dialog are "Advanced settings", "OK", and "Cancel" buttons.

3CXPhone

Accounts

Manage SIP accounts

Active	Name	Domain	Caller ID
<input checked="" type="checkbox"/>	Jude	1000@192.168.8.128	1000
<input checked="" type="checkbox"/>	Vincent	1001@192.168.8.128	1001
<input checked="" type="checkbox"/>	Nerimene	1002@192.168.8.128	1002

New

Edit

Remove

Soft keys

OK Cancel

Account settings

Account name: Jude

Caller ID: 1000

Credentials

Enter your SIP account credentials

Extension: 1000

ID: 1000

Password: ****

My location

Specify the IP of your PBX/SIP server

I am in the office - local IP 192.168.8.128 of PBX

I am out of the office - external IP

Use 3CX Tunnel

Eliminates firewall configuration. Requires 3CX Phone System for Windows

Local IP of remote PBX: 192.168.8.128

Tunnel password: *** Port: 5090

Use Outbound Proxy server

Required by some VoIP Providers. Specify IP or name.

Perform provisioning from following URL:

http://

Advanced settings OK Cancel

- ❖ Sur cette capture d'écran un message d'erreur s'affiche ce qui n'est pas SUPER pour le bon fonctionnement.
- ❖ Après des recherches, afin de comprendre pourquoi le redémarrage n'a pas marcher
Ensuite on redémarre les services asterisk : # /etc/init.d/asterisk restart
On ouvre la console astersik : # asterisk - rvvvv
Puis : sip reload

```
root@jude:/etc/asterisk
+--- Registered custom function 'QUEUE_MEMBER'
+--- Registered custom function 'QUEUE_MEMBER_COUNT'
+--- Registered custom function 'QUEUE_MEMBER_LIST'
+--- Registered custom function 'QUEUE_GET_CHANNEL'
+--- Registered custom function 'QUEUE_WAITING_COUNT'
+--- Registered custom function 'QUEUE_MEMBER_PENALTY'
+--- app_queue.so => (True Call Queueing)
[Dec  2 15:16:20] WARNING[3845]: loader.c:2398 load_modules: Some non-required modules failed to load.
[Dec  2 15:16:20] WARNING[3845]: loader.c:2492 load_modules: Module 'chan_skinny' has been loaded but was deprecated in Asterisk version 19 and will be removed in Asterisk version 21.
[Dec  2 15:16:20] WARNING[3845]: loader.c:2492 load_modules: Module 'res_adsi' has been loaded but may be removed in a future release.
[Dec  2 15:16:20] WARNING[3845]: loader.c:2492 load_modules: Module 'app_getcpeid' has been loaded but may be removed in a future release.
[Dec  2 15:16:20] WARNING[3845]: loader.c:2492 load_modules: Module 'app_adsiprogs' has been loaded but may be removed in a future release.
[Dec  2 15:16:20] ERROR[3845]: loader.c:2513 load_modules: cdr_tds declined to load.
[Dec  2 15:16:20] ERROR[3845]: loader.c:2513 load_modules: cel_sqlite3_custom declined to load.
[Dec  2 15:16:20] ERROR[3845]: loader.c:2513 load_modules: cel_tds declined to load.
[Dec  2 15:16:20] ERROR[3845]: loader.c:2513 load_modules: cdr_radius declined to load.
[Dec  2 15:16:20] ERROR[3845]: loader.c:2513 load_modules: cel_radius declined to load.
[Dec  2 15:16:20] ERROR[3845]: loader.c:2513 load_modules: cdr_sqlite3_custom declined to load.
[Dec  2 15:16:20] ERROR[3845]: loader.c:2513 load_modules: cdr_pgsql declined to load.
ASTERISK Ready.
jude*CLI>
```

Chaque fois que je redémarre Asterisk, tous mes téléphones logiciels ne se connectent pas.

Lorsque je vérifie mes pairs avec la commande `sip show peers` OU `sip reload`, j'obtiens des erreurs : -

```
No such command 'sip show peers'
```

ou.

```
No such command 'sip reload'
```

J'ai trouvé la solution temporaire mais lorsque je redémarre mon astérisque je rencontre à nouveau le même problème

Ensuite on redémarre les services asterisk : **/etc/init.d/asterisk restart**

On ouvre la console astersik : **asterisk – rvvvv**

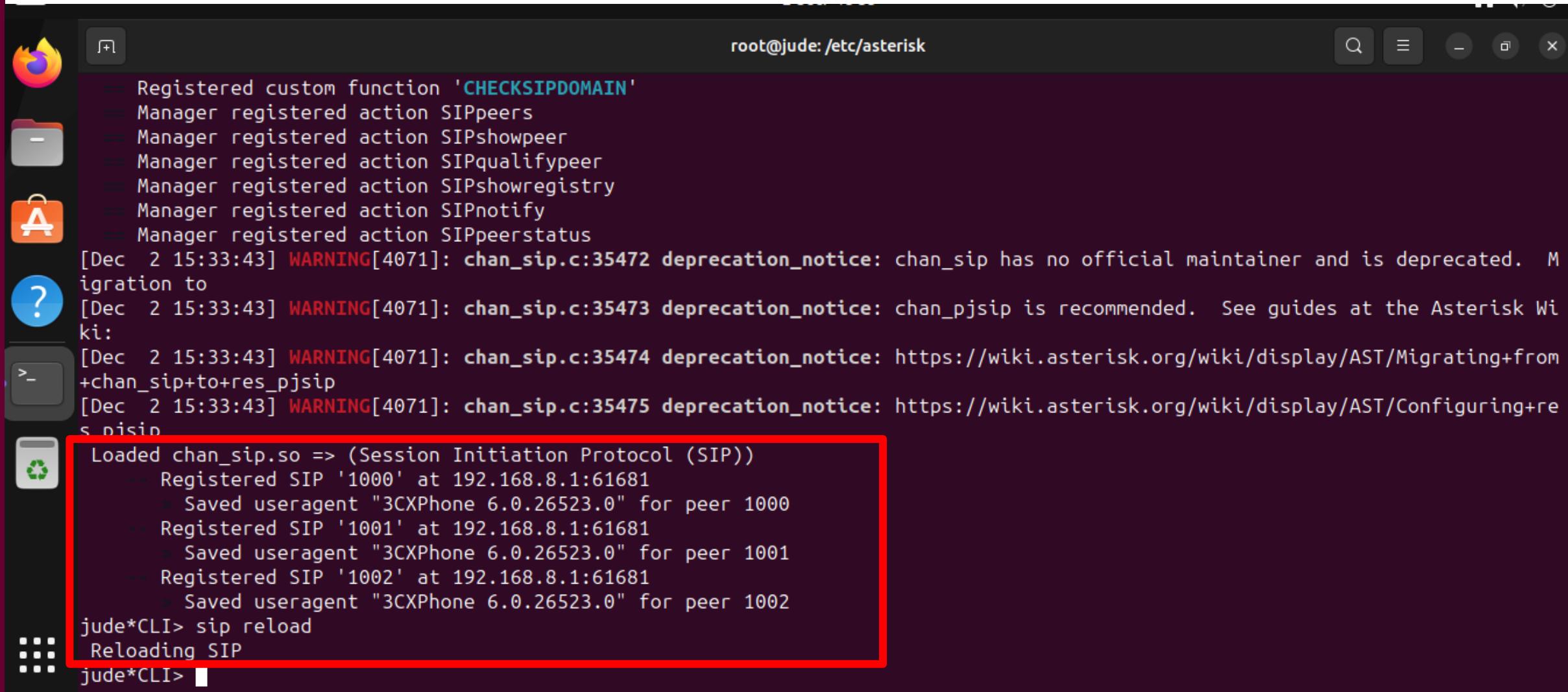
: > module load chan_sip.so

Puis : **sip reload**

À l'intérieur de la CLI astérisque :

```
module load chan_sip.so
```

Le problème est résolu et le message d'erreur à disparue, le redémarrage à pris en compte tous les enregistrements apporter dans l'application 3CXPhone



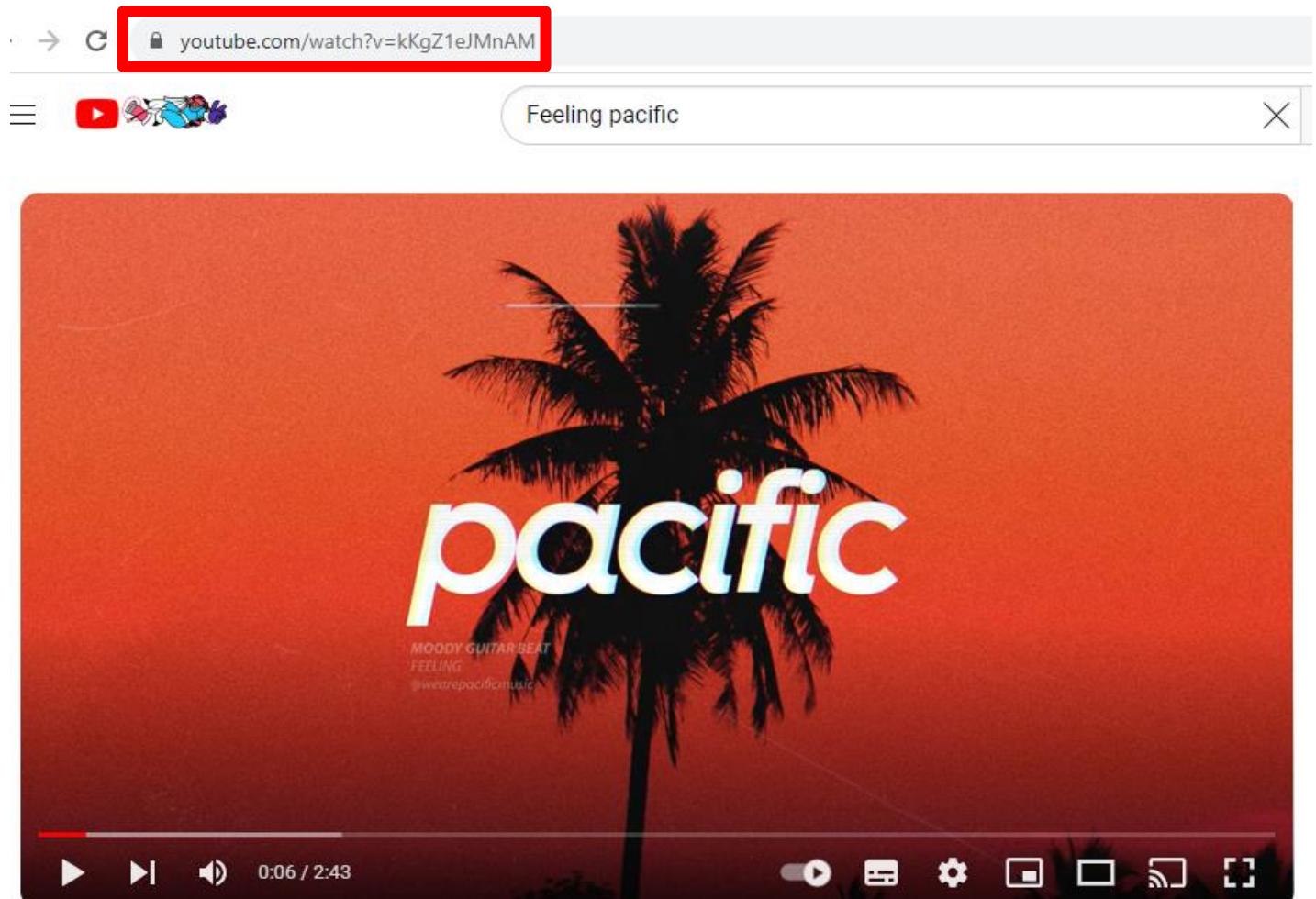
```
root@jude: /etc/asterisk
--- Registered custom function 'CHECKSIPDOMAIN'
--- Manager registered action SIPpeers
--- Manager registered action SIPshowpeer
--- Manager registered action SIPqualifypeer
--- Manager registered action SIPshowregistry
--- Manager registered action SIPnotify
--- Manager registered action SIPpeerstatus
[Dec 2 15:33:43] WARNING[4071]: chan_sip.c:35472 deprecation_notice: chan_sip has no official maintainer and is deprecated. Migration to
[Dec 2 15:33:43] WARNING[4071]: chan_sip.c:35473 deprecation_notice: chan_pjsip is recommended. See guides at the Asterisk Wiki:
[Dec 2 15:33:43] WARNING[4071]: chan_sip.c:35474 deprecation_notice: https://wiki.asterisk.org/wiki/display/AST/Migrating+from+chan_sip+to+res_pjsip
[Dec 2 15:33:43] WARNING[4071]: chan_sip.c:35475 deprecation_notice: https://wiki.asterisk.org/wiki/display/AST/Configuring+res_pjsip
Loaded chan_sip.so => (Session Initiation Protocol (SIP))
-- Registered SIP '1000' at 192.168.8.1:61681
> Saved useragent "3CXPhone 6.0.26523.0" for peer 1000
-- Registered SIP '1001' at 192.168.8.1:61681
> Saved useragent "3CXPhone 6.0.26523.0" for peer 1001
-- Registered SIP '1002' at 192.168.8.1:61681
> Saved useragent "3CXPhone 6.0.26523.0" for peer 1002
jude*CLI> sip reload
Reloading SIP
jude*CLI>
```



```
root@jude: /etc/asterisk
-- Registered SIP '1001' at 192.168.8.1:61681
  > Saved useragent "3CXPhone 6.0.26523.0" for peer 1001
-- Registered SIP '1002' at 192.168.8.1:61681
  > Saved useragent "3CXPhone 6.0.26523.0" for peer 1002
jude*CLI> sip reload
Reloading SIP
== Using SIP RTP CoS mark 5
  > 0x7f203c022220 -- Strict RTP learning after remote address set to: 192.168.8.1:40006
-- Executing [1000@default:1] Answer("SIP/1000-00000000", "") in new stack
  > 0x7f203c022220 -- Strict RTP switching to RTP target address 192.168.8.1:40006 as source
-- Executing [1000@default:2] Dial("SIP/1000-00000000", "Sip/1000,10") in new stack
== Using SIP RTP CoS mark 5
  -- Called Sip/1000
  -- SIP/1000-00000001 is ringing
    > 0x7f203c022220 -- Strict RTP learning complete - Locking on source address 192.168.8.1:40006
    > 0x7f20440176f0 -- Strict RTP learning after remote address set to: 192.168.8.1:40012
  -- SIP/1000-00000001 answered SIP/1000-00000000
  -- Channel SIP/1000-00000001 joined 'simple_bridge' basic-bridge <d79e9917-a6bb-4de1-8234-5aaa884157bc>
  -- Channel SIP/1000-00000000 joined 'simple_bridge' basic-bridge <d79e9917-a6bb-4de1-8234-5aaa884157bc>
    > Bridge d79e9917-a6bb-4de1-8234-5aaa884157bc: switching from simple_bridge technology to native_rtp
    > Remotely bridged 'SIP/1000-00000000' and 'SIP/1000-00000001' - media will flow directly between them
  -- Started music on hold, class 'default', on channel 'SIP/1000-00000001'
    > 0x7f20440176f0 -- Strict RTP switching to RTP target address 192.168.8.1:40012 as source
    > 0x7f20440176f0 -- Strict RTP learning complete - Locking on source address 192.168.8.1:40012
  -- Started music on hold, class 'default', on channel 'SIP/1000-00000000'
```

MUSIQUE D'ATTENTE SUR ASTERISK

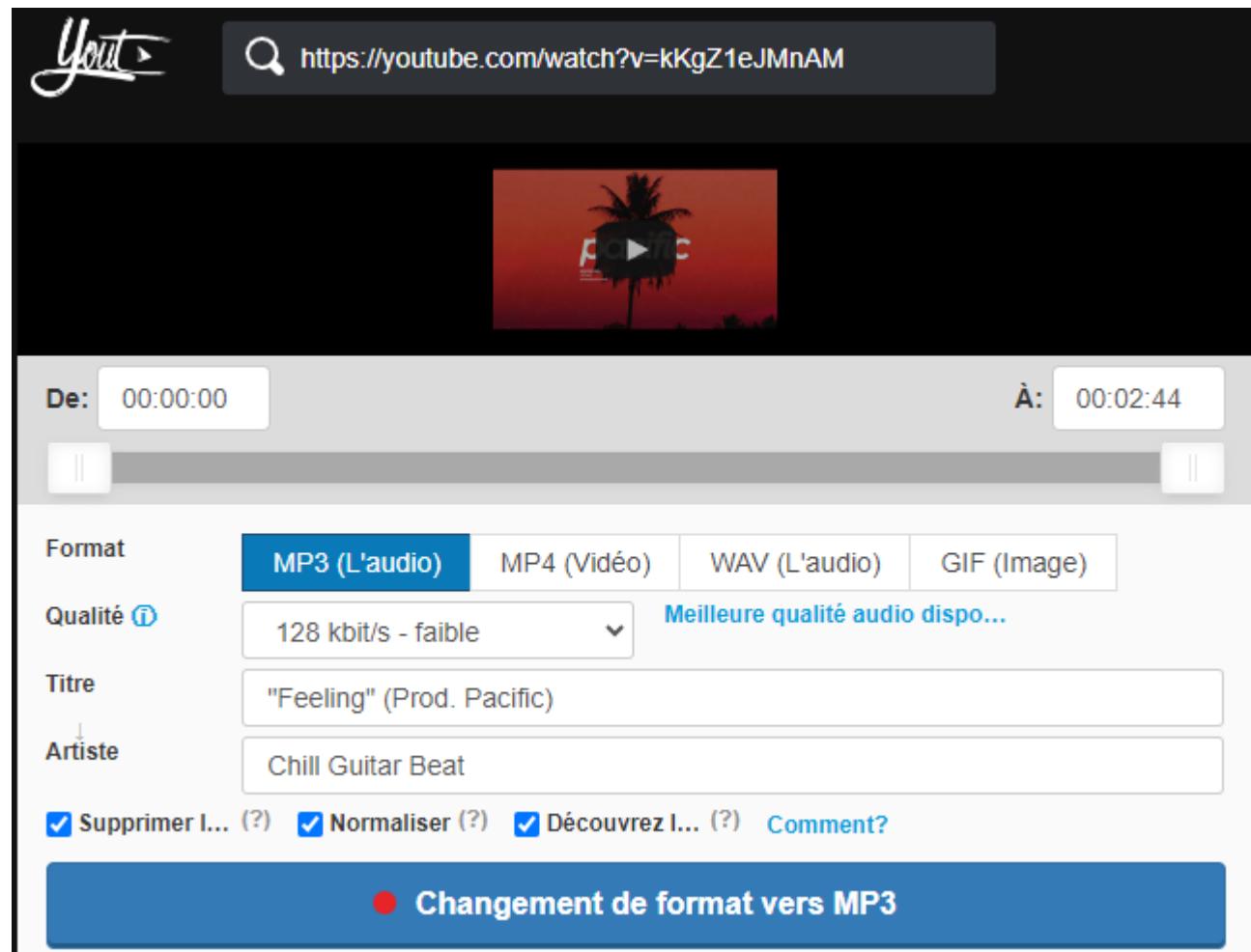
- ❖ Commençons par télécharger une musique qui vous plaît et de la transformer en mp3.
- ❖ Aller sur YouTube choisissez votre musique, puis effacer le « **ube** » de **youtube** dans la barre d'adresse pour avoir accès à un convertisseur mp3.



Chill Guitar Beat - "Feeling" (Prod. Pacific)

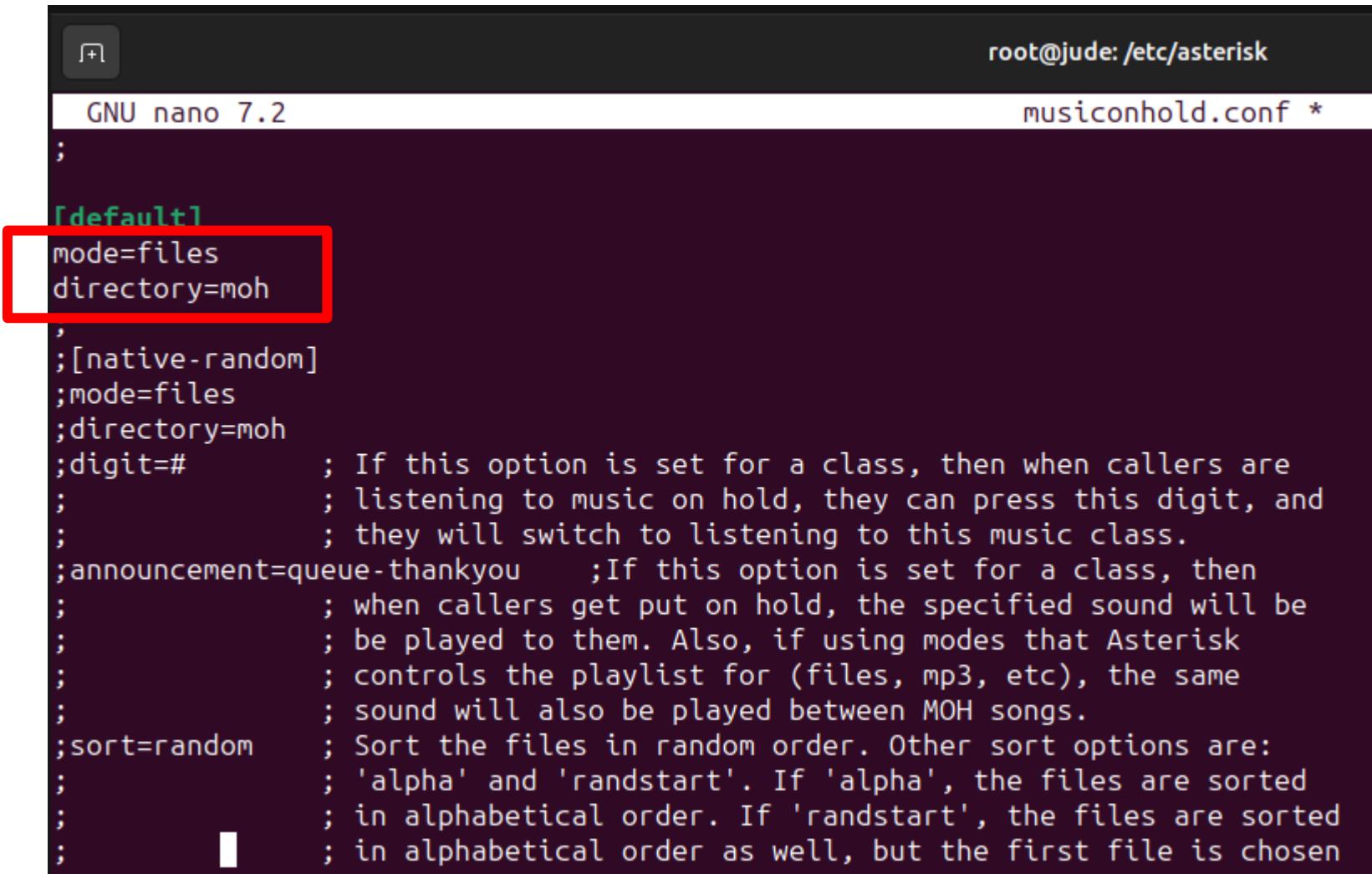
MUSIQUE D'ATTENTE SUR ASTERISK

- ❖ Commençons par télécharger une musique qui vous plaît et de la transformer en mp3.
- ❖ Aller sur YouTube choisissez votre musique, puis effacer le « **ube** » de **youtube** dans la barre d'adresse pour avoir accès à un convertisseur mp3.



```
root@jude:/etc/asterisk# ls
acl.conf          cel_beanstalkd.conf    extensions.ael      osp.conf          say.conf
adsi.conf         cel.conf             extensions.conf    phoneprov.conf   sip.conf
aeap.conf         cel_custom.conf     extensions.lua     pjproject.conf  sip.conf.old
agents.conf       cel_odbc.conf      extensions_minivm.conf pjsip.conf        sip_notify.conf
alarmreceiver.conf cel_pgsql.conf    features.conf     pjsip_notify.conf skinny.conf
alsa.conf         cel_sqlite3_custom.conf festival.conf   pjsip_wizard.conf sla.conf
amd.conf          cel_tds.conf       followme.conf    prometheus.conf smdi.conf
app_skel.conf     chan_dahdi.conf   func_odbc.conf  queues.conf      sorcery.conf
ari.conf          chan_mobile.conf  geolocation.conf res_config_mysql.conf ss7.timers
ast_debug_tools.conf cli_aliases.conf hep.conf        res_config_sqlite3.conf stasis.conf
asterisk.adsi     cli.conf          http.conf       res_corosync.conf statsd.conf
asterisk.conf     cli_permissions.conf iax.conf        res_curl.conf    stir_shaken.conf
calendar.conf    codecs.conf       iaxprov.conf    res_fax.conf    telcordia-1.adsi
ccss.conf         confbridge.conf  indications.conf logger.conf    test_sorcery.conf
cdr_adaptive_odbc.conf config_test.conf  iaxprov.conf   manager.conf   udptl.conf
cdr_beanstalkd.conf console.conf    indications.conf logger.conf    unistim.conf
cdr.conf          dbsep.conf       iaxprov.conf   meetme.conf   users.conf
cdr_custom.conf   dnsmgr.conf     indications.conf logger.conf    voicemail.conf
cdr_manager.conf  dsp.conf        iaxprov.conf   mgcp.conf     voicemail.conf.old
cdr_odbc.conf    dundi.conf      indications.conf manager.conf   xmpp.conf
cdr_pgsql.conf   enum.conf       iaxprov.conf   minivm.conf
cdr_sqlite3_custom.conf exentions.conf.old  indications.conf modules.conf
cdr_tds.conf      extconfig.conf  iaxprov.conf   motifs.conf
root@jude:/etc/asterisk#
```

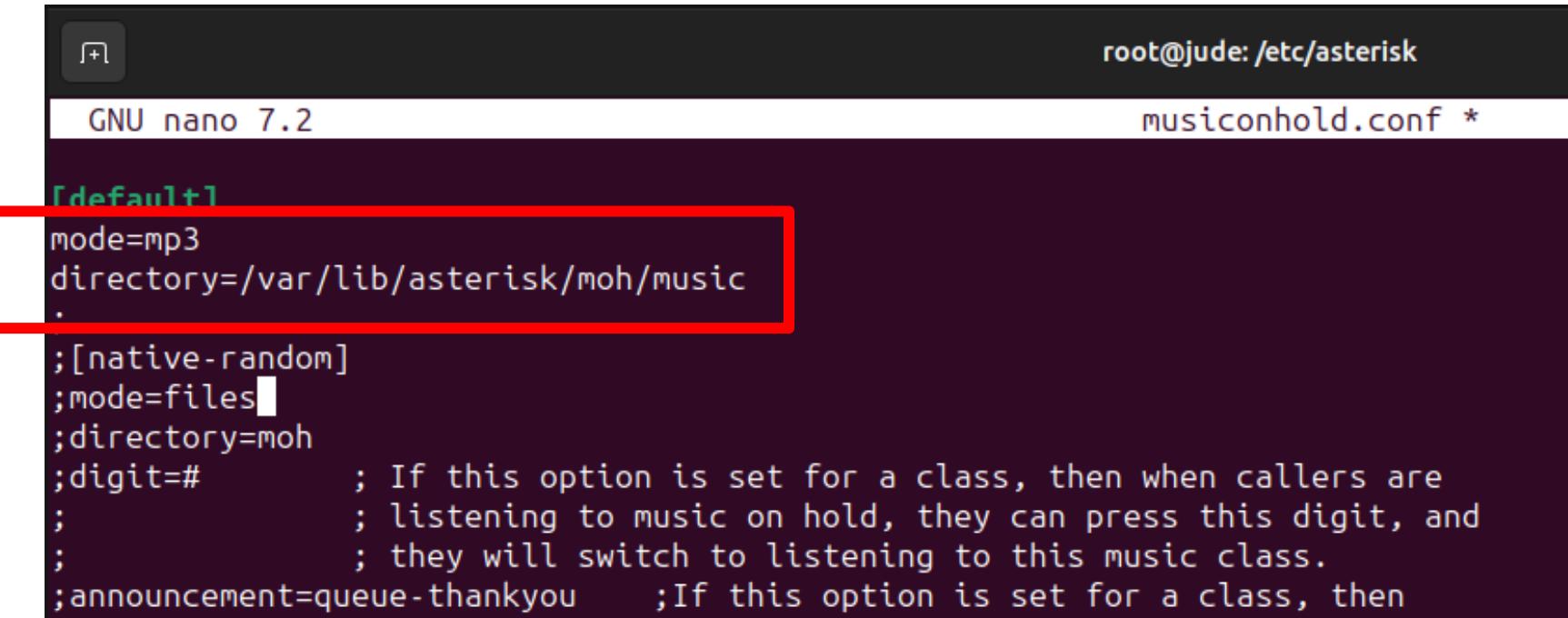
Puis on va modifier le fichier musiconhold.conf : # nano musiconhold.conf
La vous devez changer la ligne « **mode=files** » par « **mode=mp3** » et la ligne en dessous « **directory=moh** » par « **directory=le chemin vers notre musique** ».



```
root@jude: /etc/asterisk
GNU nano 7.2                         musiconhold.conf *
;
[default]
mode=files
directory=moh
,[native-random]
;mode=files
;directory=moh
;digit=#          ; If this option is set for a class, then when callers are
;                  ; listening to music on hold, they can press this digit, and
;                  ; they will switch to listening to this music class.
;announcement=queue-thankyou    ;If this option is set for a class, then
;                                ; when callers get put on hold, the specified sound will be
;                                ; be played to them. Also, if using modes that Asterisk
;                                ; controls the playlist for (files, mp3, etc), the same
;                                ; sound will also be played between MOH songs.
;sort=random      ; Sort the files in random order. Other sort options are:
;                  ; 'alpha' and 'randstart'. If 'alpha', the files are sorted
;                  ; in alphabetical order. If 'randstart', the files are sorted
;                  ; in alphabetical order as well, but the first file is chosen
```

Puis on va modifier le fichier musiconhold.conf : # nano musiconhold.conf

La vous devez changer la ligne « **mode=files** » par « **mode=mp3** » et la ligne en dessous « **directory=moh** » par « **directory=le chemin vers notre musique** ».



```
root@jude: /etc/asterisk
GNU nano 7.2                         musiconhold.conf *
[default]
mode=mp3
directory=/var/lib/asterisk/moh/music
.
;[native-random]
;mode=files
;directory=moh
;digit=#          ; If this option is set for a class, then when callers are
;                  ; listening to music on hold, they can press this digit, and
;                  ; they will switch to listening to this music class.
;announcement=queue-thankyou    ;If this option is set for a class, then
```

- ❖ On modifie ensuite notre fichier « extensions.conf » : # **nano extensions.conf**
- ❖ On ajoute la ligne suivante à la toute fin :
exten => 6,1,Musiconhold(/var/lib/asterisk/moh/music)

```
GNU nano 7.2                                         root@jude: /etc/asterisk
exten => 1001,n,Hangup()

exten => 1002,1,Answer()
exten => 1002,n,Dial(Sip/1002,10)
exten => 1002,n,VoiceMail(1002)
exten => 1002,n,Hangup()

exten => 1003,1,Answer()
exten => 1003,n,Dial(Sip/1003,10)
exten => 1003,n,VoiceMail(1003)
exten => 1003,n,Hangup()

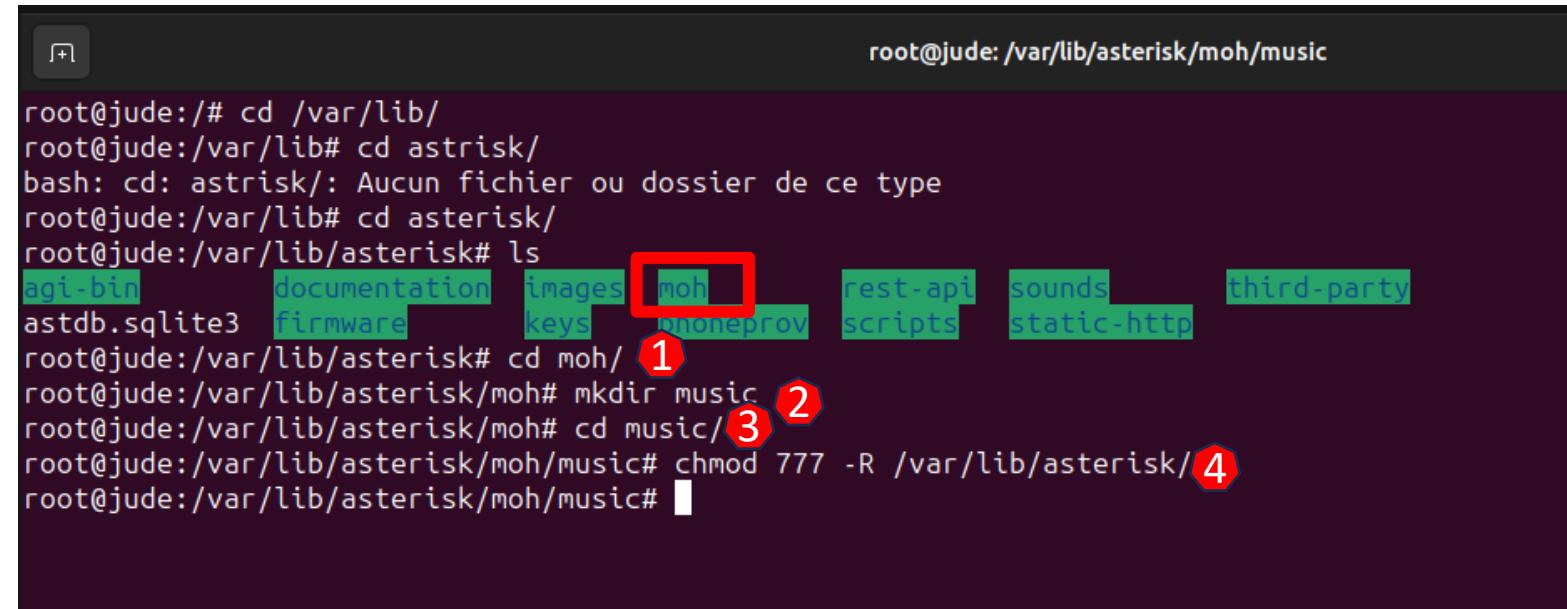
exten => 1004,1,Answer()
exten => 1004,n,Dial(Sip/1004,10)
exten => 1004,n,VoiceMail(1004)
exten => 1004,n,Hangup()

#exten => _2XXX,1,Dial(Sip/A_B/${EXTEN})
exten => 500,1,VoiceMailMain()

exten => 6,1,Musiconhold(/var/lib/asterisk/moh/music)
```

- ❖ Crée un dossier **music** dont lequel on va stocker la musique d'attente en ligne de commande et attribuer à ce dossier music tous les droits.
- ❖ On se rend ensuite dans le dossier « moh » : **# cd /var/lib/asterisk/moh**
- ❖ On créer un dossier music :
mkdir music
- ❖ On retourne dans /var/lib/asterisk/ :
cd music/
- ❖ On donne les droits :
chmod 777 -R /var/lib/asterisk

On copie ensuite notre musique mp3 téléchargé dans le dossier music soit via **Filezilla** soit moba.



The screenshot shows a terminal window with a dark background and light-colored text. The title bar says "root@jude: /var/lib/asterisk/moh/music". The terminal shows the following command sequence:

```
root@jude:/# cd /var/lib/
root@jude:/var/lib# cd asterisk/
bash: cd: asterisk/: Aucun fichier ou dossier de ce type
root@jude:/var/lib# cd asterisk/
root@jude:/var/lib/asterisk# ls
agi-bin documentation images moh ①
astdb.sqlite3 firmware keys phoneprov ②
rest-api sounds static-http third-party
scripts
root@jude:/var/lib/asterisk# cd moh/
root@jude:/var/lib/asterisk/moh# mkdir music ③ ②
root@jude:/var/lib/asterisk/moh# cd music/
root@jude:/var/lib/asterisk/moh/music# chmod 777 -R /var/lib/asterisk/ ④
```

Red boxes and numbers highlight specific parts of the terminal output:

- A red box surrounds the "moh" directory in the first "ls" command, with a red number "1" above it.
- A red box surrounds the "moh" directory in the "cd moh/" command, with a red number "2" above it.
- A red box surrounds the "music" directory in the "mkdir music" command, with a red number "3" above it.
- A red box surrounds the "asterisk" directory in the final "chmod" command, with a red number "4" above it.

Site local :	D:\ECOLE IMIE-PARIS\extensions voicemail sip\	Site distant :	/var/lib/asterisk/moh/music		
<ul style="list-style-type: none"> +... Cours de SQL IMIE PARIS +... Dossier GPO_cas pratique_active directory +... EXPOSITION TECH SHOW PARIS +... extensions voicemail sip +... ITIL_IMIE-PARIS 		<ul style="list-style-type: none"> firmware ? images ? keys moh <ul style="list-style-type: none"> music 			
Nom de fichier	Taille de fichier	Type de fichier	Dernière modification		
..					
cherie.mp3	3 017 818	Fichier MP3	02/11/2023 16:20:03		
extensions.conf (1).txt	698	Document texte	02/11/2023 14:26:32		
feeling-prod-pacific.mp3	2 608 897	Fichier MP3	02/12/2023 16:35:38		
SIP.CONF.txt	741	Document texte	02/11/2023 14:26:30		
voicemail.conf.txt	205	Document texte	02/11/2023 14:59:52		
5 fichiers. Taille totale : 5 628 359 octets					
Nom de fichier	Taille de fichier	Type de fichier	Dernière modification	Droits d'accès	Propriétaire
..					
cherie.mp3	3 017 818	Fichier MP3	02/11/2023 16:20:03		
extensions.conf (1).txt	698	Document texte	02/11/2023 14:26:32		
feeling-prod-pacific.mp3	2 608 897	Fichier MP3	02/12/2023 16:35:38	-rw-rw-r--	jude jude
SIP.CONF.txt	741	Document texte	02/11/2023 14:26:30		
voicemail.conf.txt	205	Document texte	02/11/2023 14:59:52		
Dossier vide.					

Nom de fichier	Taille de fichier	Type de fichier	Dernière modification		
..					
cherie.mp3	3 017 818	Fichier MP3	02/11/2023 16:20:03		
extensions.conf (1).txt	698	Document texte	02/11/2023 14:26:32		
feeling-prod-pacific.mp3	2 608 897	Fichier MP3	02/12/2023 16:35:38		
SIP.CONF.txt	741	Document texte	02/11/2023 14:26:30		
voicemail.conf.txt	205	Document texte	02/11/2023 14:59:52		
Sélection de 1 fichier. Taille totale : 2 608 897 octets					
Nom de fichier	Taille de fichier	Type de fichier	Dernière modification	Droits d'accès	Propriétaire
..					
feeling-prod-pacific.mp3	2 608 897	Fichier MP3	02/12/2023 17:00:00	-rw-rw-r--	jude jude
1 fichier. Taille totale : 2 608 897 octets					

Serveur / Fichier local	Direction	Fichier distant	Taille	Priorité	Statut
Fichiers en file d'attente	Transferts échoués	Transferts réussi (2)			

 FileZilla

Transferts réussis

Tous les fichiers ont été transférés avec succès

- ❖ Profiter pour ajouter les droits à votre musique en faisant un clic droit dessus et en cochant les cases :
- Clique droit sur music feeling-prod-pacific en bas droit (**TOUS COCHER**)

The screenshot shows a file synchronization interface with two main panes and a dialog box.

Left Pane (Site local): Displays the local file structure at `D:\ECOLE IMIE-PARIS\extensions voicemail sip\`. It includes folders like `Cours de SQL IMIE PARIS`, `Dossier GPO_ cas pratique_active directory`, `EXPOSITION TECH SHOW PARIS`, `extensions voicemail sip`, and `ITIL_IMIE-PARIS`. Below this is a table listing files:

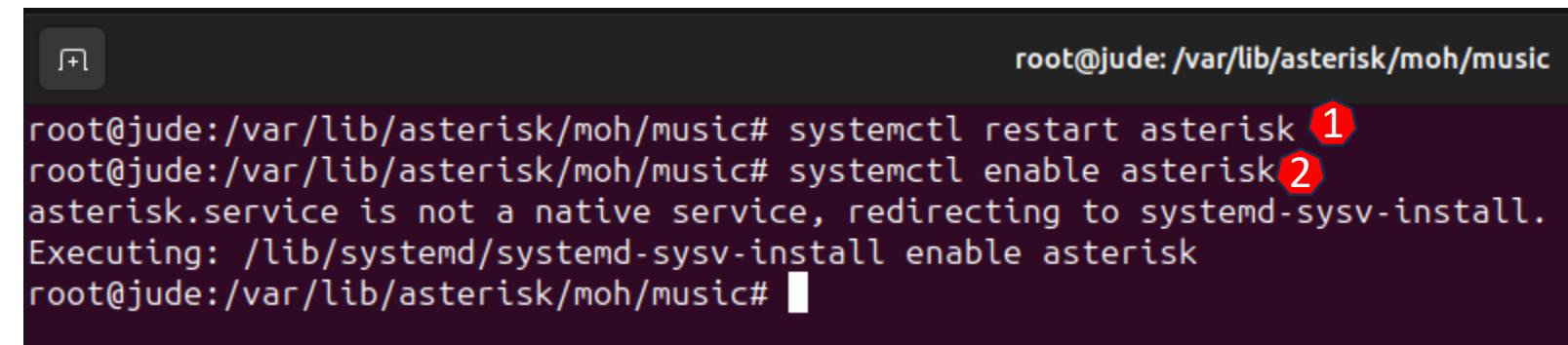
Nom de fichier	Taille de fic...	Type de fichier	Dernière modificat...
..			
cherie.mp3	3 017 818	Fichier MP3	02/11/2023 16:20:03
extensions.conf (1).txt	698	Document texte	02/11/2023 14:26:32
feeling-prod-pacific.mp3	2 608 897	Fichier MP3	02/12/2023 16:35:38
SIP.CONF.txt	741	Document texte	02/11/2023 14:26:30
voicemail.conf.txt	205	Document texte	02/11/2023 14:59:52

Right Pane (Site distant): Displays the remote file structure at `/var/lib/asterisk/moh/music`. It includes subfolders `firmware`, `images`, `keys`, `moh`, and `music`.

Dialog Box (Modification des attributs du fichier): A modal window for changing file attributes. It contains three sections of checkboxes for owner, group, and public permissions, all of which are checked (Lire, Écrire, Exécuter). A red box highlights the checkboxes for the public section. Below the checkboxes is a numeric value field set to `777`. At the bottom are `OK` and `Annuler` buttons.

- ❖ Pour redémarrer le service faire la commande suivante :

```
# systemctl restart asterisk  
# systemctl enable asterisk
```

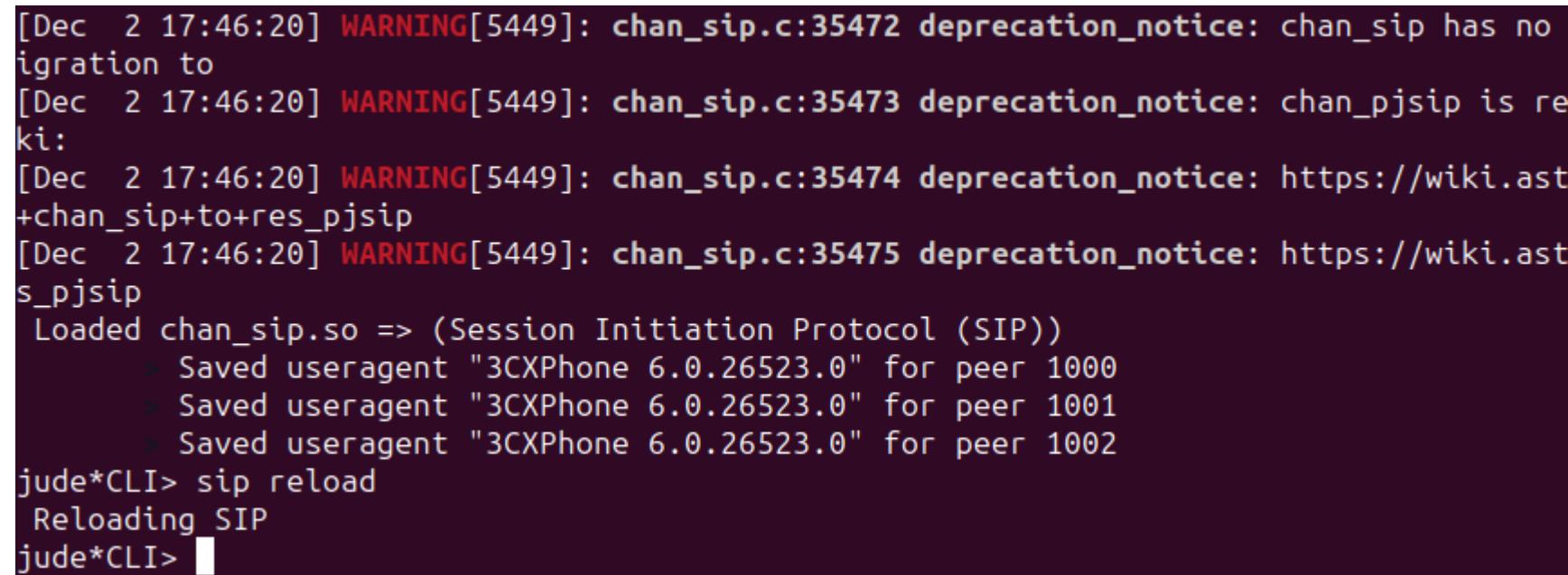


```
root@jude:/var/lib/asterisk/moh/music# systemctl restart asterisk ①  
root@jude:/var/lib/asterisk/moh/music# systemctl enable asterisk ②  
asterisk.service is not a native service, redirecting to systemd-sysv-install.  
Executing: /lib/systemd/systemd-sysv-install enable asterisk  
root@jude:/var/lib/asterisk/moh/music#
```

- ❖ Enfin on redémarre asterisk :

```
# /etc/init.d/asterisk restart
```

- On ouvre la console asterisk :> **asterisk -rvvvv**
- De là on tape la commande :> **sip reload**



```
[Dec 2 17:46:20] WARNING[5449]: chan_sip.c:35472 deprecation_notice: chan_sip has no  
igration to  
[Dec 2 17:46:20] WARNING[5449]: chan_sip.c:35473 deprecation_notice: chan_pjsip is re  
ki:  
[Dec 2 17:46:20] WARNING[5449]: chan_sip.c:35474 deprecation_notice: https://wiki.ast  
+chan_sip+to+res_pjsip  
[Dec 2 17:46:20] WARNING[5449]: chan_sip.c:35475 deprecation_notice: https://wiki.ast  
s_pjsip  
Loaded chan_sip.so => (Session Initiation Protocol (SIP))  
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1000  
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1001  
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1002  
jude*CLI> sip reload  
Reloading SIP  
jude*CLI>
```



```
Reloading SIP
-- Using SIP RTP CoS mark 5
  > 0x7f95f00223e0 -- Strict RTP learning after remote address set to: 192.168.8.1:40038
-- Executing [1000@default:1] Answer("SIP/1002-00000000", "") in new stack
  > 0x7f95f00223e0 -- Strict RTP switching to RTP target address 192.168.8.1:40038 as source
-- Executing [1000@default:2] Dial("SIP/1002-00000000", "Sip/1000,10") in new stack
-- Using SIP RTP CoS mark 5
-- Called Sip/1000
-- SIP/1000-00000001 is ringing
-- Call on SIP/1000-00000001 placed on hold
  > 0x7f95f80162b0 -- Strict RTP learning after remote address set to: 192.168.8.1:40040
-- Started music on hold, class 'default', on channel 'SIP/1000-00000001'
-- SIP/1000-00000001 answered SIP/1002-00000000
-- Channel SIP/1000-00000001 joined 'simple_bridge' basic-bridge <8ce211d8-0439-4c04-b4bd-7801dc61c902>
-- Channel SIP/1002-00000000 joined 'simple_bridge' basic-bridge <8ce211d8-0439-4c04-b4bd-7801dc61c902>
  > Bridge 8ce211d8-0439-4c04-b4bd-7801dc61c902: switching from simple_bridge technology to native_rtp
  > Remotely bridged 'SIP/1002-00000000' and 'SIP/1000-00000001' - media will flow directly between them
  > 0x7f95f80162b0 -- Strict RTP switching to RTP target address 192.168.8.1:40040 as source
  > 0x7f95f80162b0 -- Strict RTP learning complete - Locking on source address 192.168.8.1:40040
```

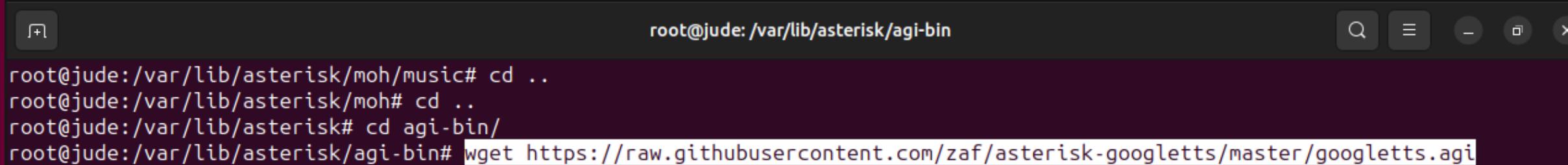
REDIRECTION VOCALE

- ❖ Pour la redirection vocale on va devoir se rendre dans le dossier agi-bin :

```
# cd /var/lib/asterisk/agi_bin
```

- ❖ La on télécharge le paquet suivant :

```
# wget https://raw.githubusercontent.com/zaf/asterisk-googletts/master/googletts.agi
```



A screenshot of a terminal window titled "root@jude: /var/lib/asterisk/agi-bin". The terminal shows a command history and the current command being typed:

```
root@jude:/var/lib/asterisk/moh/music# cd ..  
root@jude:/var/lib/asterisk/moh# cd ..  
root@jude:/var/lib/asterisk# cd agi-bin/  
root@jude:/var/lib/asterisk/agi-bin# wget https://raw.githubusercontent.com/zaf/asterisk-googletts/master/googletts.agi
```

- ❖ On lui donne les droits : **# chmod 777 googletts.agi**
- ❖ On édite ensuite le fichier extensions.conf : **# nano /etc/asterisk/extensions.conf**

```
root@jude: /var/lib/asterisk/agi-bin
root@jude:/var/lib/asterisk/moh/music# cd ..
root@jude:/var/lib/asterisk/moh# cd ..
root@jude:/var/lib/asterisk# cd agi-bin/
root@jude:/var/lib/asterisk/agi-bin# wget https://raw.githubusercontent.com/zaf/asterisk-googletts/master/googletts.agi
--2023-12-02 18:03:41--  https://raw.githubusercontent.com/zaf/asterisk-googletts/master/googletts.agi
Résolution de raw.githubusercontent.com (raw.githubusercontent.com)... 185.199.111.133, 185.199.108.133, 185.199.109.133, ...
Connexion à raw.githubusercontent.com (raw.githubusercontent.com)|185.199.111.133|:443... connecté.
requête HTTP transmise, en attente de la réponse... 200 OK
Taille : 10224 (10,0K) [text/plain]
Enregistre : 'googletts.agi'

googletts.agi          100%[=====] 9,98K  --. KB/s  ds 0,008s

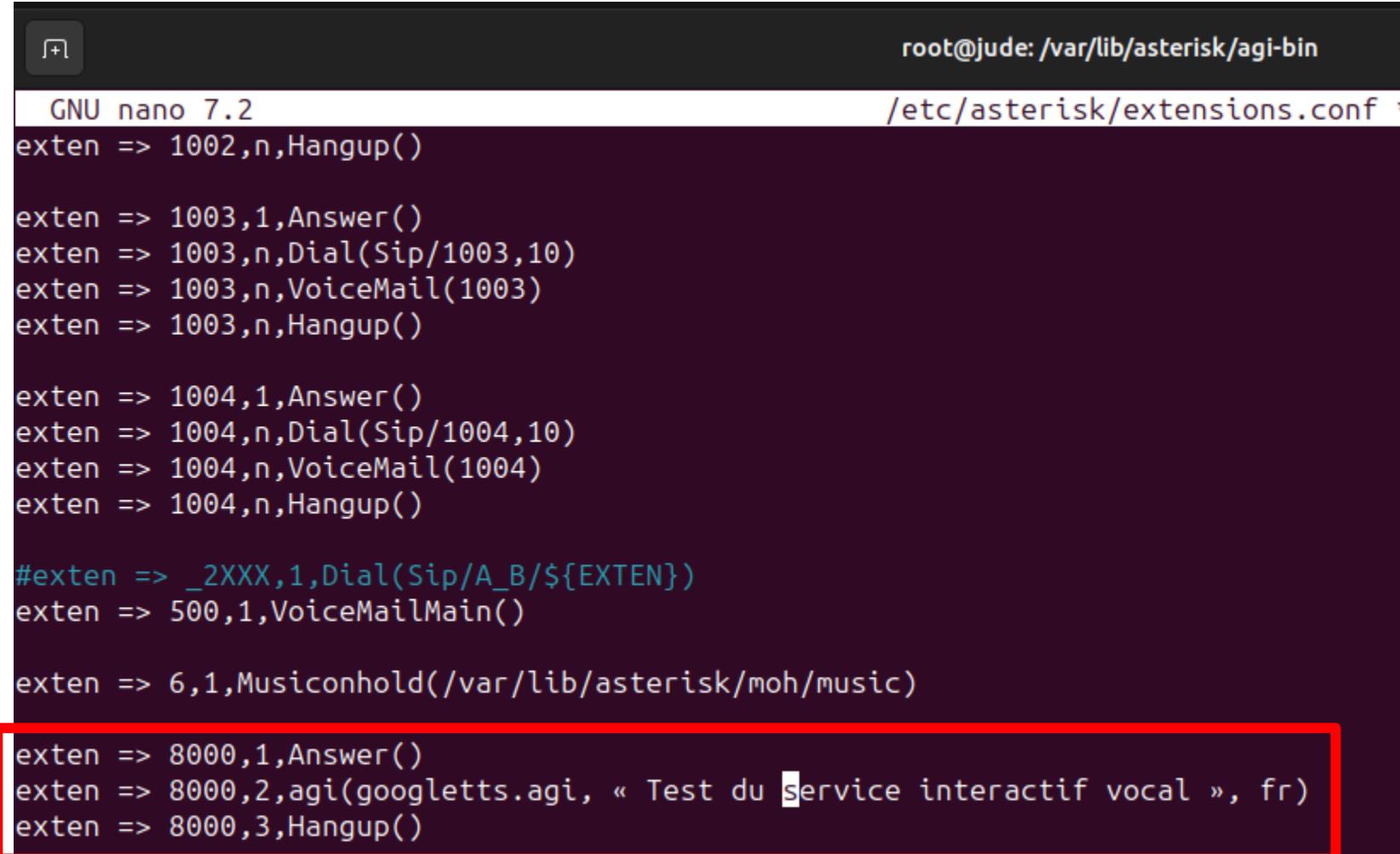
2023-12-02 18:03:43 (1,24 MB/s) - 'googletts.agi' enregistré [10224/10224]

root@jude:/var/lib/asterisk/agi-bin# chmod 777 googletts.agi
root@jude:/var/lib/asterisk/agi-bin# nano /etc/asterisk/extensions.conf
```

On ajoute les lignes suivantes à la fin du fichier :

#IVR Test

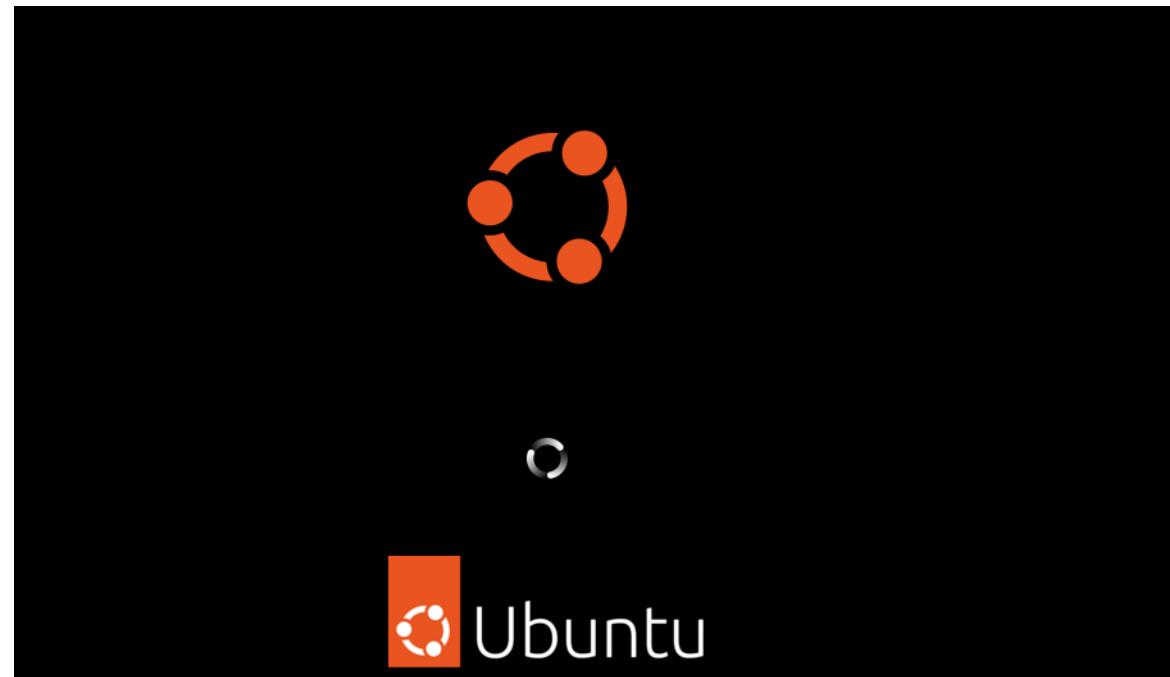
```
exten => 8000,1,Answer()  
exten => 8000,2,agi(googletts.agi, « Test du service interactif vocal », fr)  
exten => 8000,3,Hangup()
```



```
root@jude: /var/lib/asterisk/agi-bin  
GNU nano 7.2 /etc/asterisk/extensions.conf *  
exten => 1002,n,Hangup()  
  
exten => 1003,1,Answer()  
exten => 1003,n,Dial(Sip/1003,10)  
exten => 1003,n,VoiceMail(1003)  
exten => 1003,n,Hangup()  
  
exten => 1004,1,Answer()  
exten => 1004,n,Dial(Sip/1004,10)  
exten => 1004,n,VoiceMail(1004)  
exten => 1004,n,Hangup()  
  
#exten => _2XXX,1,Dial(Sip/A_B/${EXTEN})  
exten => 500,1,VoiceMailMain()  
  
exten => 6,1,Musiconhold(/var/lib/asterisk/moh/music)  
  
exten => 8000,1,Answer()  
exten => 8000,2,agi(googletts.agi, « Test du service interactif vocal », fr)  
exten => 8000,3,Hangup()
```

- ❖ Ensuite on redémarre les services asterisk :# /etc/init.d/asterisk restart
- ❖ On ouvre la console astersik :> **asterisk -rvvvv**
 # reboot

```
root@jude: /var/lib/asterisk/agi-bin
root@jude:/var/lib/asterisk/agi-bin# /etc/init.d/asterisk restart
Restarting asterisk (via systemctl): asterisk.service.
root@jude:/var/lib/asterisk/agi-bin# asterisk -rvvvv
Unable to connect to remote asterisk (does '/var/run/asterisk/asterisk.ctl' exist?)
root@jude:/var/lib/asterisk/agi-bin# reboot
```

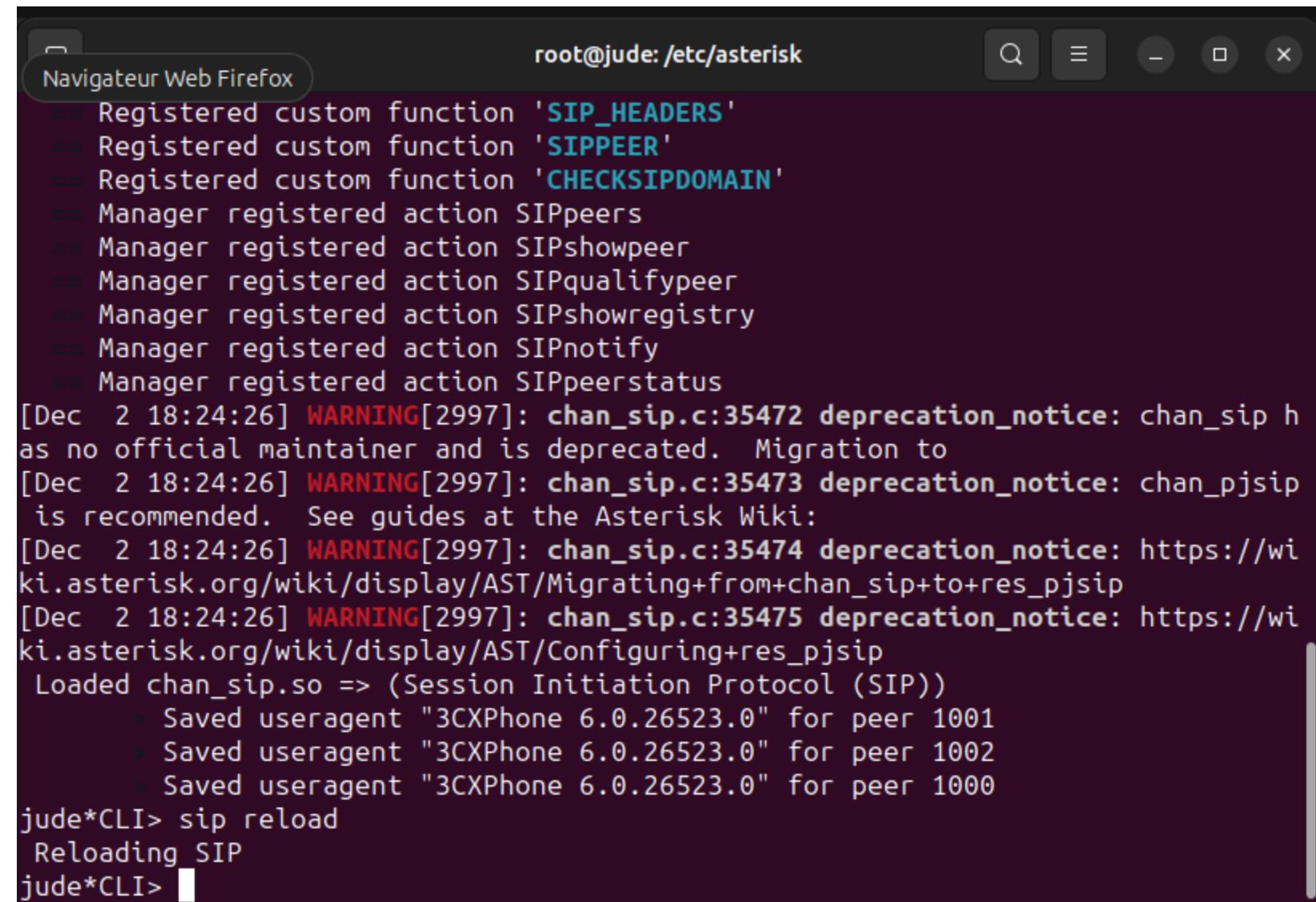


- ❖ On ouvre la console astersik :

> asterisk – rvvvv

Puis : sip reload

- ❖ Vous pouvez tester cela avec votre 3CX en composant le 8000.



The screenshot shows a Firefox browser window titled "Navigateur Web Firefox" with the URL "root@jude: /etc/asterisk". The content of the window is the Asterisk CLI output:

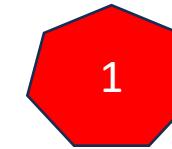
```
root@jude: /etc/asterisk
== Registered custom function 'SIP_HEADERS'
== Registered custom function 'SIPPEER'
== Registered custom function 'CHECKSIPDOMAIN'
== Manager registered action SIPpeers
== Manager registered action SIPshowpeer
== Manager registered action SIPqualifypeer
== Manager registered action SIPshowregistry
== Manager registered action SIPnotify
== Manager registered action SIPpeerstatus
[Dec 2 18:24:26] WARNING[2997]: chan_sip.c:35472 deprecation_notice: chan_sip has no official maintainer and is deprecated. Migration to
[Dec 2 18:24:26] WARNING[2997]: chan_sip.c:35473 deprecation_notice: chan_pjsip is recommended. See guides at the Asterisk Wiki:
[Dec 2 18:24:26] WARNING[2997]: chan_sip.c:35474 deprecation_notice: https://wiki.asterisk.org/wiki/display/AST/Migrating+from+chan_sip+to+res_pjsip
[Dec 2 18:24:26] WARNING[2997]: chan_sip.c:35475 deprecation_notice: https://wiki.asterisk.org/wiki/display/AST/Configuring+res_pjsip
Loaded chan_sip.so => (Session Initiation Protocol (SIP))
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1001
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1002
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1000
jude*CLI> sip reload
    Reloading SIP
jude*CLI>
```



```
root@jude: /etc/asterisk
ki.asterisk.org/wiki/display/AST/Configuring+res_pjsip
Loaded chan_sip.so => (Session Initiation Protocol (SIP))
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1001
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1002
    > Saved useragent "3CXPhone 6.0.26523.0" for peer 1000
jude*CLI> sip reload
Reloading SIP
== Using SIP RTP CoS mark 5
    > 0x7fee100238a0 -- Strict RTP learning after remote address set to: 192.168.8.1:40042
    -- Executing [8000@default:1] Answer("SIP/1000-00000000", "") in new stack
        > 0x7fee100238a0 -- Strict RTP switching to RTP target address 192.168.8.1:40042 as source
        -- Executing [8000@default:2] AGI("SIP/1000-00000000", "googletts.agi, « Test du service interactif vocal », fr") in new stack
            -- Launched AGI Script /var/lib/asterisk/agi-bin/googletts.agi
            -- <SIP/1000-00000000> Playing '/tmp/ggl_d_QcXaDC.slin' (escape_digits=) (sample_offset 0) (language 'fr')
                > 0x7fee100238a0 -- Strict RTP learning complete - Locking on source address 192.168.8.1:40042
            -- <SIP/1000-00000000>AGI Script googletts.agi completed, returning 0
            -- Executing [8000@default:3] Hangup("SIP/1000-00000000", "") in new stack
== Spawn extension (default, 8000, 3) exited non-zero on 'SIP/1000-00000000'
jude*CLI>
```

Service vocale ++

Une fois tester on peut améliorer notre service vocal : on va y ajouter la redirection vocale en ajoutant toute ces lignes à la fin du fichier extensions.conf : # **nano extensions.conf**

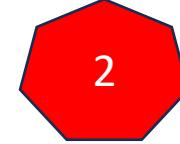


```
#extensions des IVR
#lorsque le numéro 400 est composé, le dialplan indique au serveur qu'il doit se rendre au contexte
;nommé « ivr » pour la suite de la communication.
exten => 400,1,Goto(ivr,s,1)
```

```
[ivr]
#La première étape de l'appel est le fait que le serveur répond.
exten => s,1,Answer()
#On met un timeout de 10 secondes pour le que l'appelant fasse un choix parmi ceux proposés.
exten => s,n,Set(TIMEOUT(response)=20)
#On annonce les différents choix. "agi" : est l'application utilisé, "googletts.agi" est le script qui lira le ;texte, "fr"
est la langue, "any" : l'appelant peut appuyer sur n'importe quelle touche pour arrêter
```

Service vocale ++

Une fois tester on peut améliorer notre service vocal : on va y ajouter la redirection vocale en ajoutant toute ces lignes à la fin du fichier extensions.conf : # **nano extensions.conf**

A red octagonal button with a black border and a white center, containing the number 2.

```
#la lecture du texte, "1.3" est la vitesse de lecture du texte, elle est par défaut à 1 .
exten => s,n,agi(googletts.agi,"Bienvenue au Greta!nous sommes ouvert du lundi au vendredi de 9h à 17h" mais
on mange très beaucoup pendant des heures,fr,any,1.3)
#Wait permet de mettre un temps d'attente avant la lecture de la ligne suivante
exten => s,n,Wait(1)
exten => s,n,agi(googletts.agi,"Pour accéder au service informatique, tapez 1",fr,any,1.3)
exten => s,n,Wait(1)
exten => s,n,agi(googletts.agi,"Pour accéder au service administratif, tapez 2",fr,any,1.3)
exten => s,n,Wait(1)
exten => s,n,agi(googletts.agi,"Pour accéder au service de support, tapez 3",fr,any,1.3)

exten => s,n,Wait(1)
exten=> s,n,agi(googletts.agi,"Appuyez sur dièse, si vous souhaitez réécouter ce message",fr,any,1.3)
```

Service vocale ++

Une fois tester on peut améliorer notre service vocal : on va y ajouter la redirection vocale en ajoutant toute ces lignes à la fin du fichier extensions.conf : # nano extensions.conf

#On attend que l'utilisateur appuis sur une touche pour faire son choix

exten => s,n,WaitExten()

#Si l'utilisateur appuis sur 1 on va à la priorité 1 du contexte informatique

exten => 1,1,Goto(informatique,s,1)

#Si l'utilisateur appuis sur 2 on va à la priorité 1 du contexte administratif

exten => 2,1,Goto(administratif,s,1)

#Si l'utilisateur appuis sur 3 on va à la priorité 1 du contexte support

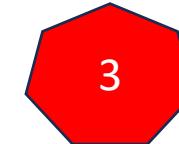
exten => 3,1,Goto(support,s,1)

#Si l'utilisateur tape un numéro compris entre 3 et 9 et # il retourne à l'étape 3 de l'IVR

exten => _[4-9#],1,Goto(ivr,s,3)

#Si l'utilisateur ne fais rien il retourne à l'étape 3 de l'IVR au bout de 10 secondes

exten => t,1,Goto(ivr,s,3)

A red octagonal button with a black border and a white number '3' in the center.

Service vocale ++

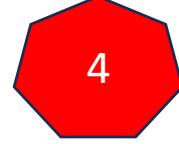
Une fois tester on peut améliorer notre service vocal : on va y ajouter la redirection vocale en ajoutant toute ces lignes à la fin du fichier extensions.conf : # **nano extensions.conf**

[informatique]

```
exten => s,1,Dial(SIP/1000,5)  
exten => s,n,Voicemail(1000)  
exten => s,n,Hangup()
```

[administratif]

```
exten => s,1,Dial(SIP/1001,5)  
exten => s,n,Voicemail(1001)  
exten => s,n,Hangup()
```

A red octagonal button with a black border and a white center, containing the number 4.

[support]

```
exten => s,1,Dial(SIP/1002,5)  
exten => s,n,Voicemail(1002)  
exten => s,n,Hangup()
```

Service vocale ++

Une fois tester on peut améliorer notre service vocal : on va y ajouter la redirection vocale en ajoutant toute ces lignes à la fin du fichier extensions.conf : # **nano extensions.conf**

```
root@jude: /etc/asterisk
GNU nano 7.2                                         extensions.conf *
exten => 8000,2,agi(googletts.agi, « Test du service interactif vocal », fr)
exten => 8000,3,Hangup()

#extensions des IVR
#lorsque le numéro 400 est composé, le dialplan indique au serveur qu'il doit se rendre au contexte ;nommé « ivr »
exten => 400,1,Goto(ivr,s,1)

[ivr]
#La première étape de l'appel est le fait que le serveur répond.
exten => s,1,Answer()
#On met un timeout de 10 secondes pour le que l'appelant fasse un choix parmi ceux proposés.
exten => s,n,Set(TIMEOUT(response)=20)
#On annonce les différents choix. "agi" : est l'application utilisé, "googletts.agi" est le script qui lira le ;texte, "fr" es>
#la lecture du texte, "1.3" est la vitesse de lecture du texte, elle est par défaut à 1 .
exten => s,n,agi(googletts.agi,"Bienvenue dans l'entreprise Charli nofile!nous sommes ouvert du lundi au vendredi de 9h à 17h">
#Wait permet de mettre un temps d'attente avant la lecture de la ligne suivante
exten => s,n,Wait(1)
exten => s,n,agi(googletts.agi,"Pour accéder au service informatique, tapez 1",fr,any,1.3)
exten => s,n,Wait(1)
```

```
root@jude:/etc/asterisk# nano extensions.conf
root@jude:/etc/asterisk# systemctl restart asterisk
root@jude:/etc/asterisk# systemctl enable asterisk
asterisk.service is not a native service, redirecting to systemd-sysv-install.
Executing: /lib/systemd/systemd-sysv-install enable asterisk
root@jude:/etc/asterisk# /etc/init.d/asterisk restart
Restarting asterisk (via systemctl): asterisk.service.
root@jude:/etc/asterisk# █
```

```
Recording the message
- x=0, open writing: /var/spool/asterisk/voicemail/default/1000/tmp/jAgC02 format: wav, 0x7fc6b801b140
- User hung up
Spawn extension (informatique, s, 2) exited non-zero on 'SIP/1000-00000001'
Using SIP RTP CoS mark 5
  > 0x7fc6a8017c80 -- Strict RTP learning after remote address set to: 192.168.8.1:40024
- Executing [1000@default:1] Answer("SIP/1000-00000003", "") in new stack
  > 0x7fc6a8017c80 -- Strict RTP switching to RTP target address 192.168.8.1:40024 as source
- Executing [1000@default:2] Dial("SIP/1000-00000003", "Sip/1000,10") in new stack
Using SIP RTP CoS mark 5
- Called Sip/1000
- SIP/1000-00000004 is ringing
- Got SIP response 486 "Busy Here" back from 192.168.8.1:52088
- SIP/1000-00000004 is busy
Everyone is busy/congested at this time (1:1/0/0)
- Executing [1000@default:3] VoiceMail("SIP/1000-00000003", "1000") in new stack
- <SIP/1000-00000003> Playing 'vm-intro.ulaw' (language 'fr')
  > 0x7fc6a8017c80 -- Strict RTP learning complete - Locking on source address 192.168.8.1:40024
- <SIP/1000-00000003> Playing 'beep.ulaw' (language 'fr')
- Recording the message
- x=0, open writing: /var/spool/asterisk/voicemail/default/1000/tmp/28i6yL format: wav, 0x7fc6cc0260f0
- User hung up
Spawn extension (default, 1000, 3) exited non-zero on 'SIP/1000-00000003'
```

