Installation d'Asterisk

```
useradmin1@debian:~$ apt get install asterisk
```

L'arrêt et le redémarrage d'Asterisk peuvent se faire à l'aide de la commande service.

```
useradmin1@debian:~$ systemctl restart asterisk
```

Les clients SIP peuvent utiliser les ports TCP et UDP 5060 pour se connecter au serveur. Le port 5061 est utilisé dans le cas d'une utilisation de TLS. La commande ps permet de vérifier que notre serveur est prêt. Il est aussi possible d'utiliser la commande service asterisk status.

```
useradmin1@debian:~$ ps -ef|grep asterisk
```

Interface en ligne de commande d'Asterisk :

```
root@debian:/# sudo asterisk -r www
Asterisk 16.16.1~dfsg-1, Copyright (C) 1999 - 2018, Digium, Inc. 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 16.16.1~dfsg-1 currently running on debian (pid = 663)
debian*CLI>
```

Créer un utilisateur :

```
root@debian:/# cd /etc/asterisk
root@debian:/etc/asterisk# nano users.conf
```

```
  GNU nano 5.4                    users.conf
[general]
hasvoicemail = yes
hassip = yes

[template](!)
type = friend
host = dynamic
dtmfmode = rfc2833
disallow = all
allow = ulaw
allow = alaw

[0120](template)
fullname = John
username = JOJO
secret = 1234 ;
mailbox = 0120
context = finance

[0121](template)
                    [ Lecture de 32 lignes ]
^G Aide       ^O Écrire     ^W Chercher   ^K Couper     ^T Exécuter
^X Quitter    ^R Lire fich. ^\ Remplacer  ^U Coller     ^J Justifier
```

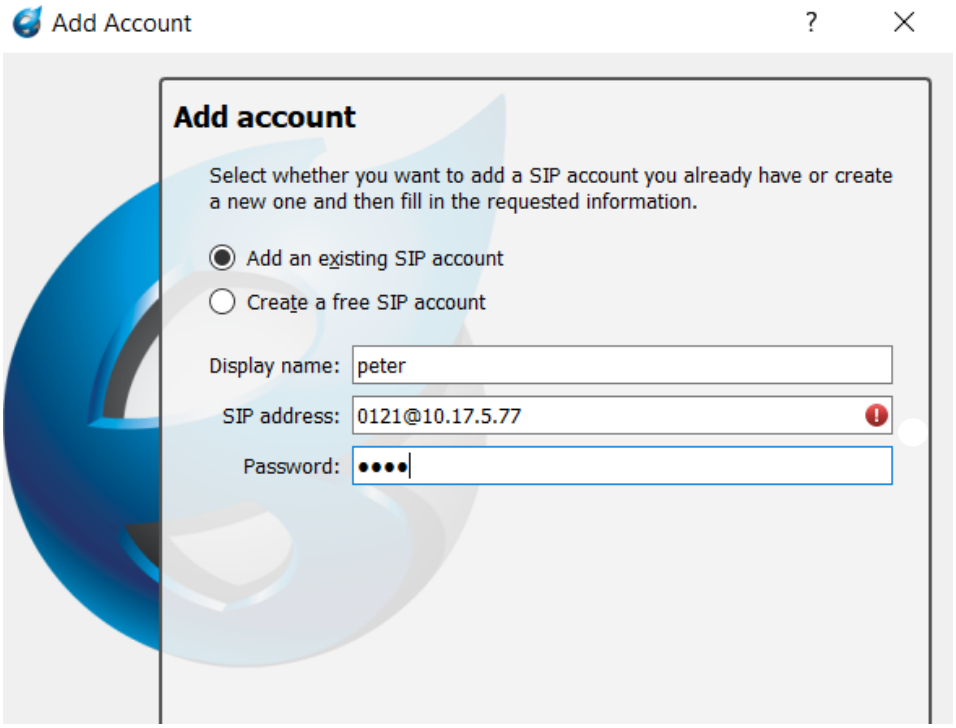
Creation du DiaPlan

```
  GNU nano 5.4                    extensions.conf
[general]
static = yes
writeprotect = yes
clearglobalvars = yes

[finance]
exten => _01XX,1,DIAL(SIP/${EXTEN},10)
exten => _01XX,2, HangUp()
```

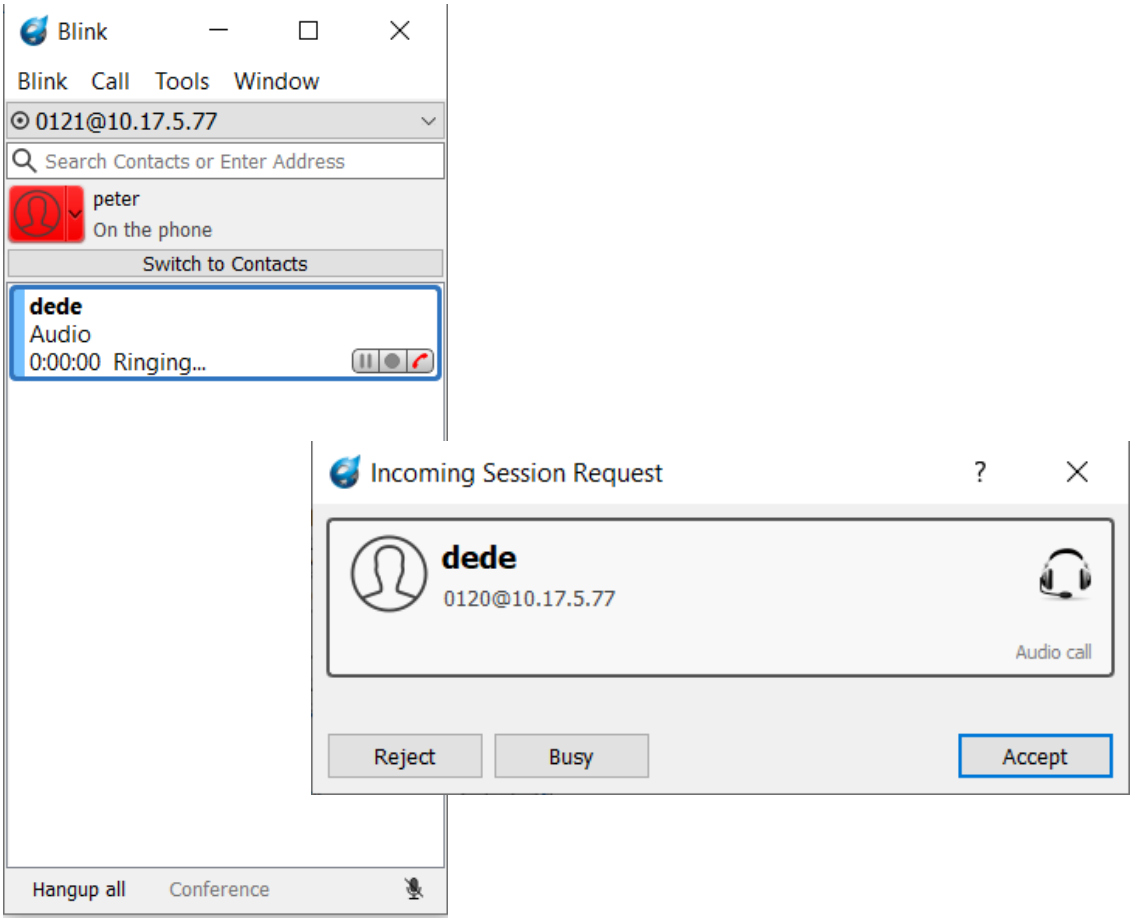
Après avoir redémarrer le service Asterisk, on peut vérifier la prise en compte du « Dial plan » :

```
debian*CLI> diaplan show
No such command 'diaplan show' (type 'core show help diaplan show' for other p
ossible commands)
debian*CLI> dialplan show
[ Context 'default' created by 'pbx_config' ]
  '0120' =>         hint: SIP/0120                                [pbx_config]
                    1. Gosub(0120,stdexten(${HINT}))              [pbx_config]
  '0121' =>         hint: SIP/0121                                [pbx_config]
                    1. Gosub(0121,stdexten(${HINT}))              [pbx_config]
  '0122' =>         hint: SIP/0122                                [pbx_config]
                    1. Gosub(0122,stdexten(${HINT}))              [pbx_config]
  '1234' =>         hint: SIP/1234                                [pbx_lua]
  Alt. Switch =>    'Lua/'                                        [pbx_lua]

[ Context 'public' created by 'pbx_lua' ]
  Alt. Switch =>    'Lua/'                                        [pbx_lua]
```

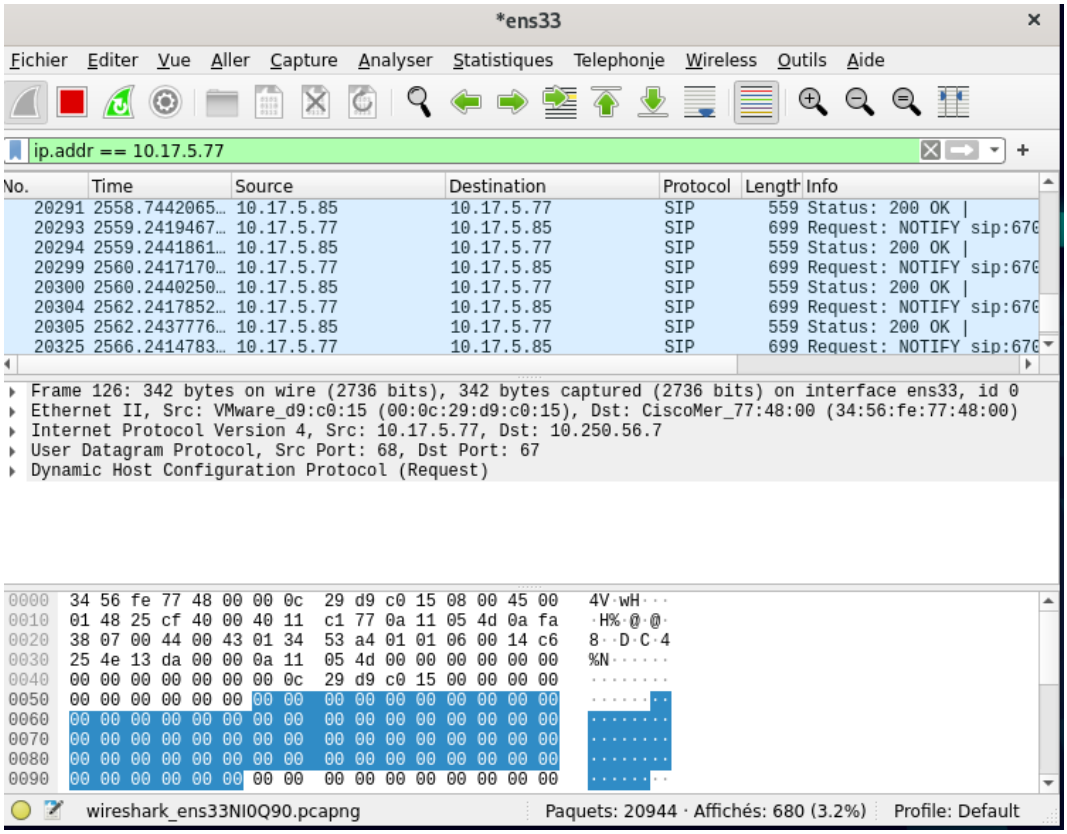
Blink :



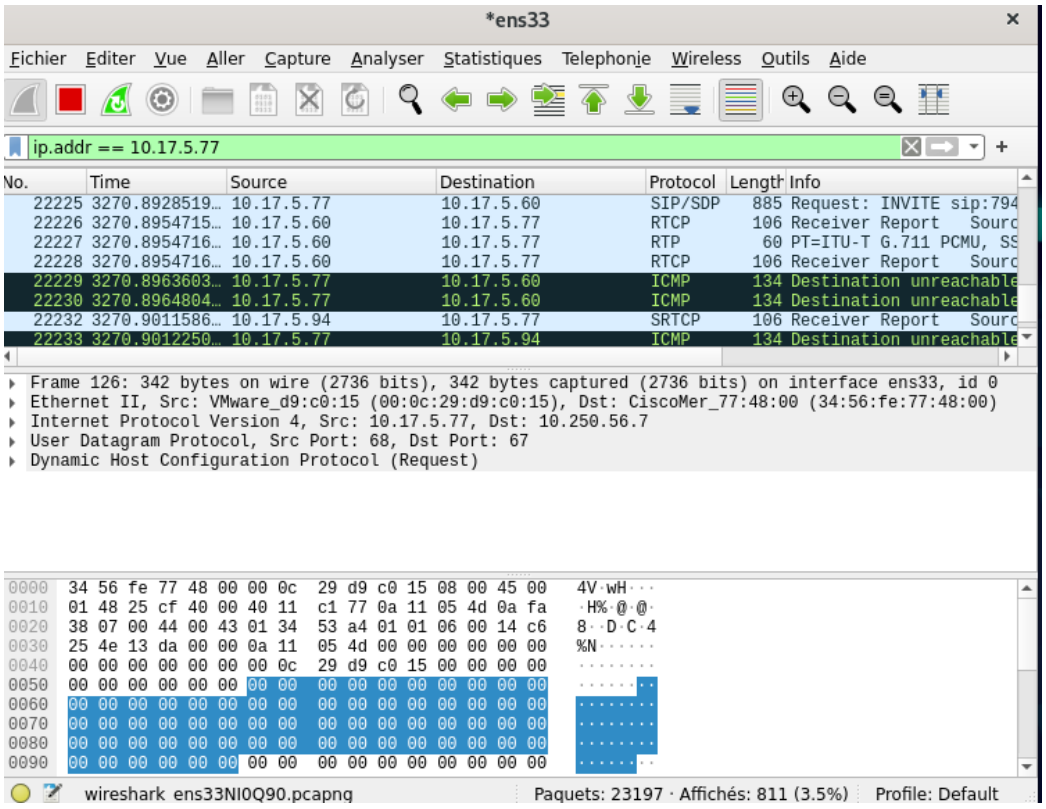
Je vais appeler mon second utilisateur et inversement :



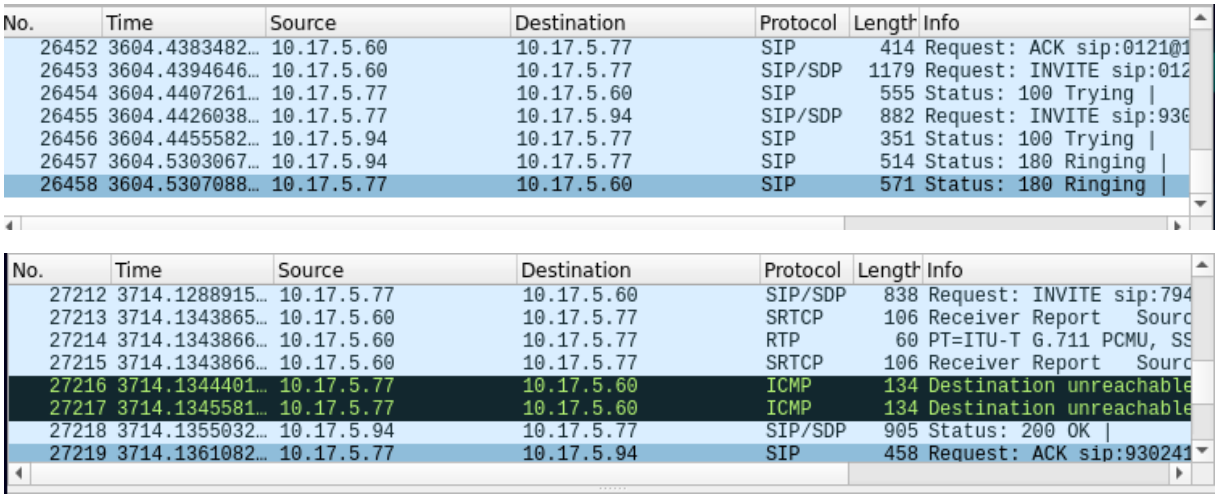
Wireshark :



Je communique avec le second utilisateur :



Le second utilisateur appel le premier :



No.	Time	Source	Destination	Protocol	Length	Info
26452	3604.4383482…	10.17.5.60	10.17.5.77	SIP	414	Request: ACK sip:0121@1
26453	3604.4394646…	10.17.5.60	10.17.5.77	SIP/SDP	1179	Request: INVITE sip:012
26454	3604.4407261…	10.17.5.77	10.17.5.60	SIP	555	Status: 100 Trying |
26455	3604.4426038…	10.17.5.77	10.17.5.94	SIP/SDP	882	Request: INVITE sip:930
26456	3604.4455582…	10.17.5.94	10.17.5.77	SIP	351	Status: 100 Trying |
26457	3604.5303067…	10.17.5.94	10.17.5.77	SIP	514	Status: 180 Ringing |
26458	3604.5307088…	10.17.5.77	10.17.5.60	SIP	571	Status: 180 Ringing |

No.	Time	Source	Destination	Protocol	Length	Info
27212	3714.1288915…	10.17.5.77	10.17.5.60	SIP/SDP	838	Request: INVITE sip:794
27213	3714.1343865…	10.17.5.60	10.17.5.77	SRTCP	106	Receiver Report Sourc
27214	3714.1343866…	10.17.5.60	10.17.5.77	RTP	60	PT=ITU-T G.711 PCMU, SS
27215	3714.1343866…	10.17.5.60	10.17.5.77	SRTCP	106	Receiver Report Sourc
27216	3714.1344401…	10.17.5.77	10.17.5.60	ICMP	134	Destination unreachable
27217	3714.1345581…	10.17.5.77	10.17.5.60	ICMP	134	Destination unreachable
27218	3714.1355032…	10.17.5.94	10.17.5.77	SIP/SDP	905	Status: 200 OK |
27219	3714.1361082…	10.17.5.77	10.17.5.94	SIP	458	Request: ACK sip:930241

INVITE : Le client demande une nouvelle session

ACK : Accusé de réception

REGISTER : Permet de s’enregistrer auprès de l’IPBX

Ringing : Entrain d’appeler le destinataire