TP Réseaux Multimédia

Téléphonie sur IP avec le protocole SIP

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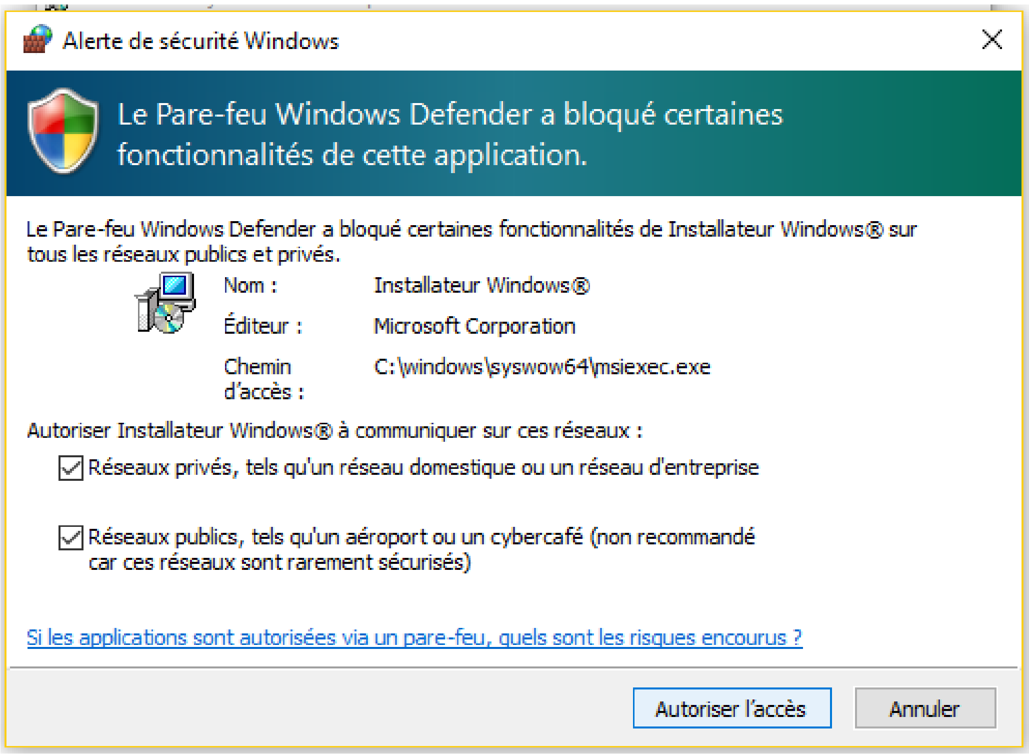
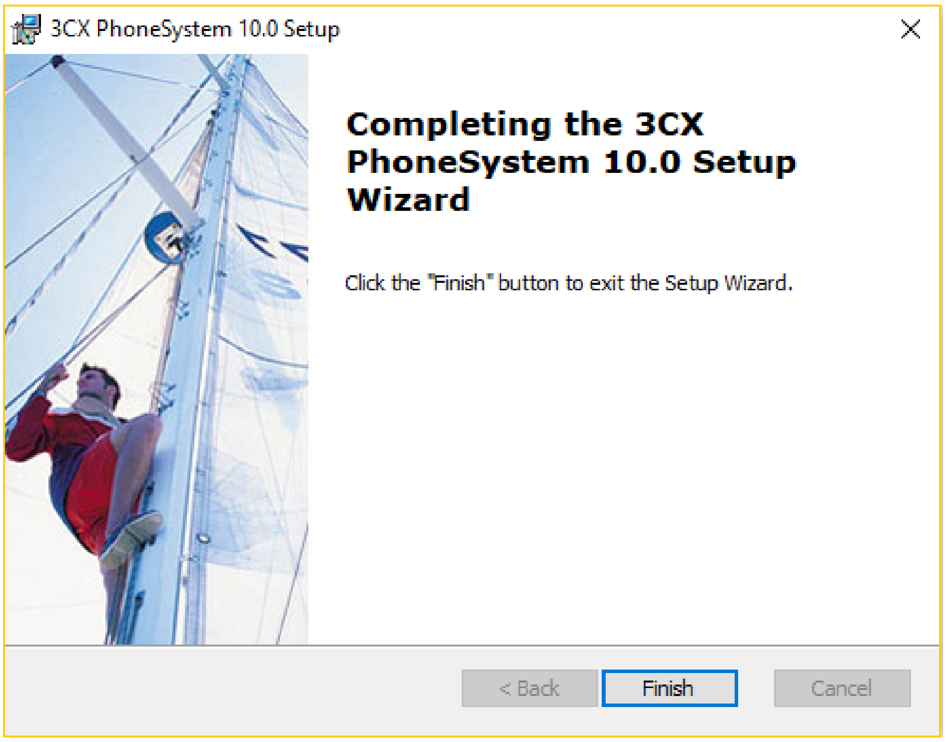
Objectif : installer, configurer et analyser le fonctionnement d'un système de téléphonie sur IPv4 compatible avec le protocole SIP (Session Initiation Protocol) et couplant Téléphonie, Vidéoconference et messagerie éléctronique.

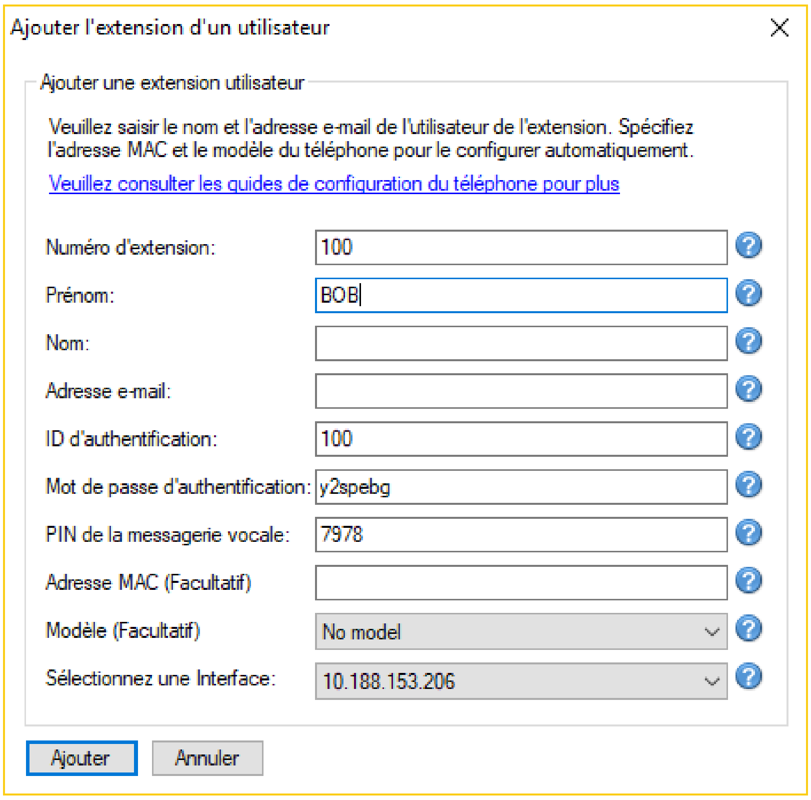
- Répondre aux questions et reportez vos réponses sur ce document. Les captures d'écran sont appréciées pour illustrer/valider vos réponses. Convertir votre document en PDF si possible avant envoi par email : master2srs.dir@gmail.com en précisant dans le sujet du mail : ***« M1 Cyber – TP3 ToIP »***.
- Date limite de remise de votre rapport par email : 21 avril 17h00
- Pour répondre aux questions suivantes veuillez installer les logiciels fournis et utiliser la documentation associée. Aidez vous si besoin d'Internet.

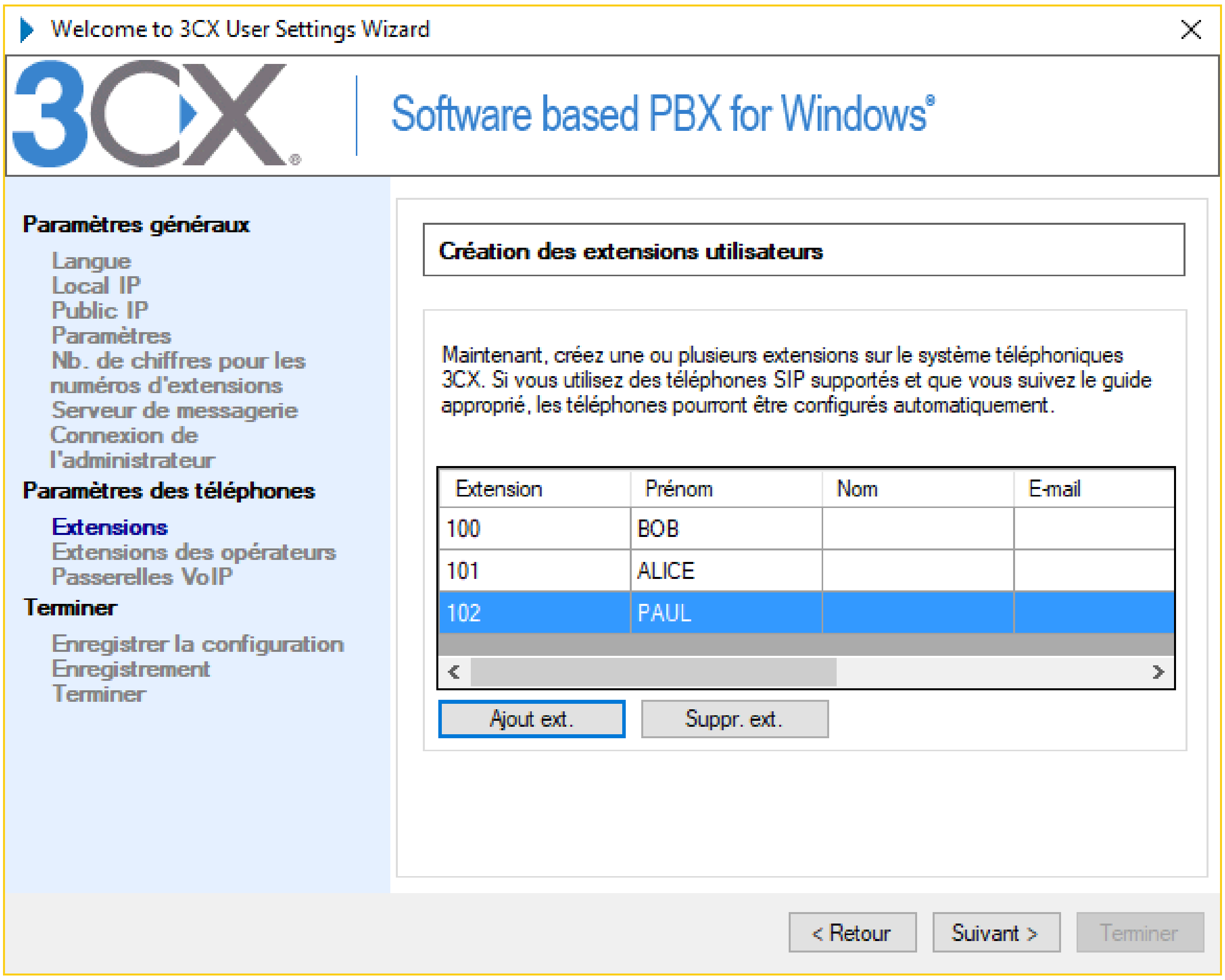
- **Partie 1 : Téléphonie sur IP avec le protocole SIP en mode de communication via un serveur téléphonique (SIP proxy) et authentification des clients**

10. **install** the 3CX SIP proxy server. **Create** three (3) user accounts (BOB (100), ALICE (101) and PAUL (102)) in the SIP server administration console to authenticate each SIP client during the calling process. https://www.3cx.fr



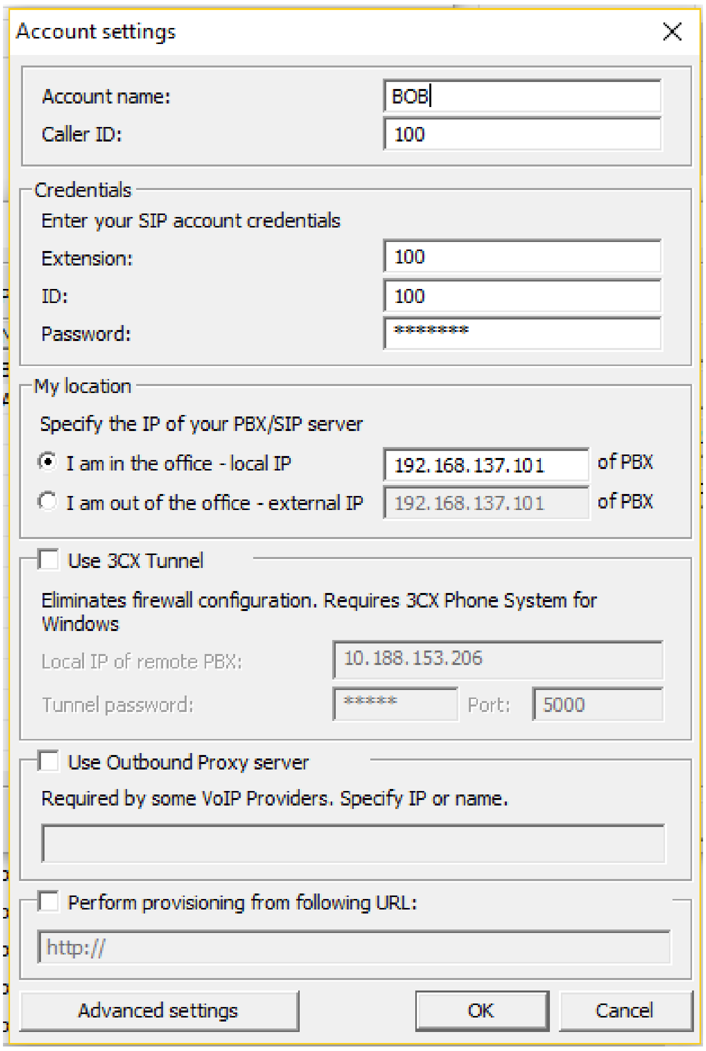

3CX PhoneSystem 10.0 Setup
Completing the 3CX PhoneSystem 10.0 Setup Wizard
Click the "Finish" button to exit the Setup Wizard.
< Back
Finish
Cancel

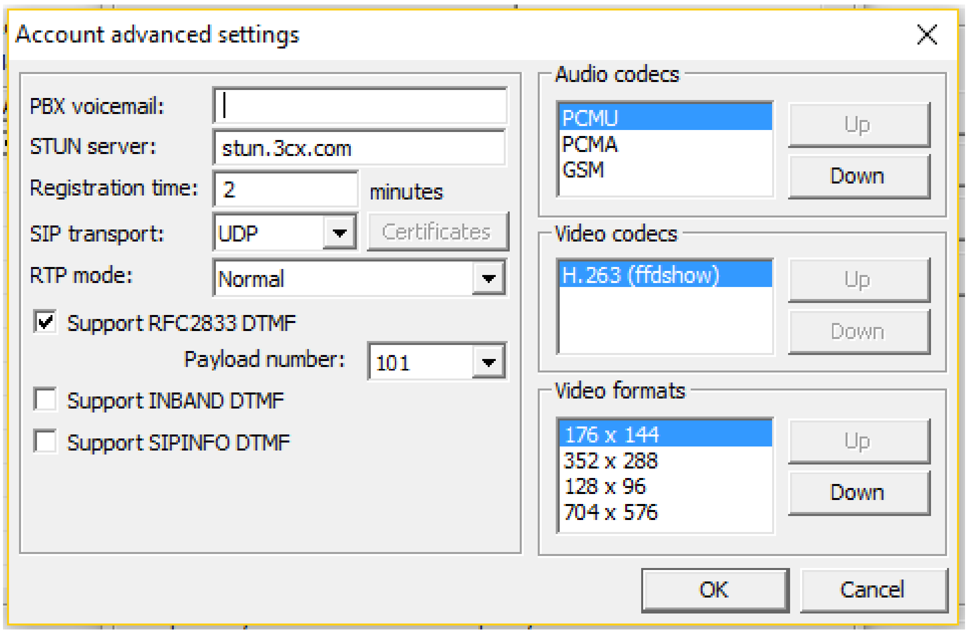

Ajouter l'extension d'un utilisateur
Ajouter une extension utilisateur
Veuillez saisir le nom et l'adresse e-mail de l'utilisateur de l'extension. Spécifiez l'adresse MAC et le modèle du téléphone pour le configurer automatiquement.
Veuillez consulter les guides de configuration du téléphone pour plus
Numéro d'extension: 100
Prénom: BOB
Nom:
Adresse e-mail:
ID d'authentification: 100
Mot de passe d'authentification: y2spebg
PIN de la messagerie vocale: 7978
Adresse MAC (Facultatif)
Modèle (Facultatif) No model
Sélectionnez une Interface: 10.188.153.206
Ajouter
Annuler

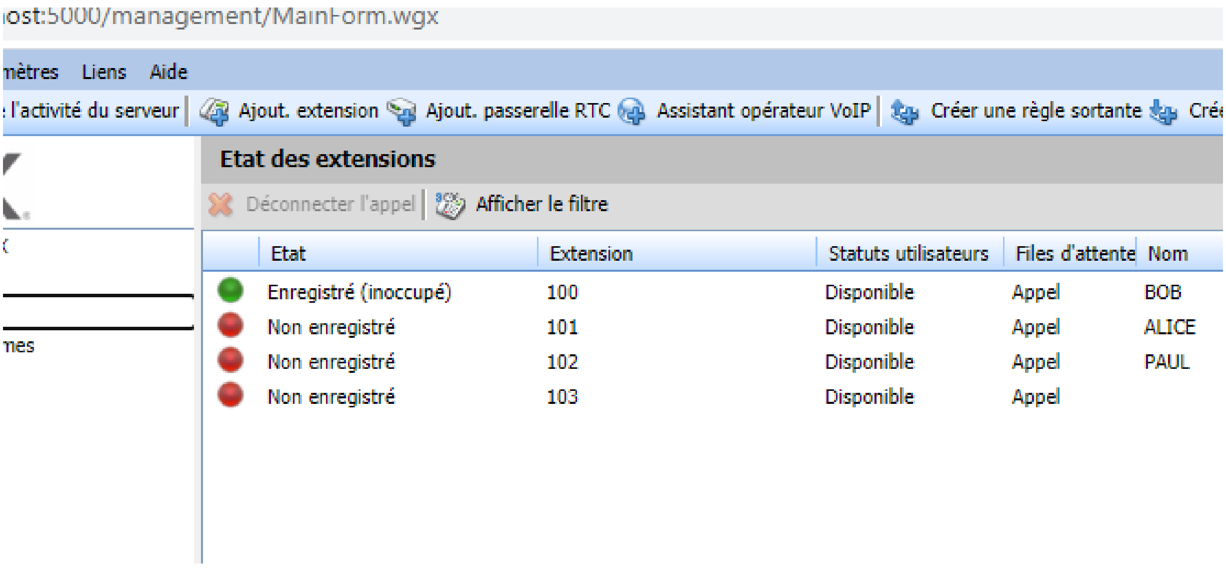


11. **Configure** your first SIP softphone clients to create a profile using BOB account to log on to your SIP proxy server. **Configure** your second SIP 3CX VoIP softphone to create another profile using ALICE account to log on to the SIP proxy server of your student mate. You may use a laptop or your mobile/Tablet devices to setup your lab.

- Beware : we may need to update your hosts file on server and client systems
- Beware : use PCMA or PCMU audio codec in your VoIP clients.
- Beware : use only numbers (no letters or special characters) for the password
- Beware : desactivate or reconfigure your windows firewall to open the voip tcp/udp ports.


Account settings
Account name: BOB
Caller ID: 100
Credentials
Enter your SIP account credentials
Extension: 100
ID: 100
Password: ********
My location
Specify the IP of your PBX/SIP server
I am in the office - local IP 192.168.137.101 of PBX
I am out of the office - external IP 192.168.137.101 of PBX
Use 3CX Tunnel
Eliminates firewall configuration. Requires 3CX Phone System for Windows
Local IP of remote PBX: 10.188.153.206
Tunnel password: ***** Port: 5000
Use Outbound Proxy server
Required by some VoIP Providers. Specify IP or name.
Perform provisioning from following URL:
http://
Advanced settings
OK
Cancel

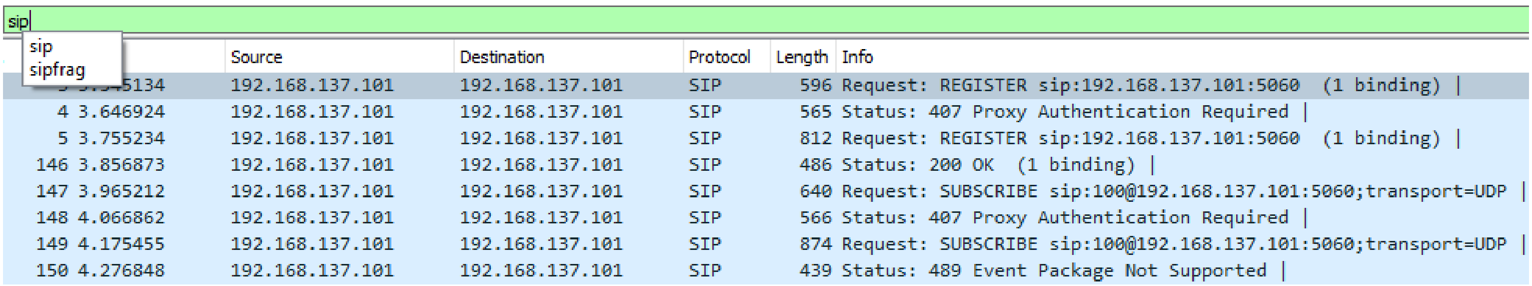

Account advanced settings
PBX voicemail:
STUN server: stun.3cx.com
Registration time: 2 minutes
SIP transport: UDP Certificates
RTP mode: Normal
Support RFC2833 DTMF
Payload number: 101
Support INBAND DTMF
Support SIPINFO DTMF
Audio codecs
PCMU
PCMA
GSM
Up
Down
Video codecs
H.263 (ffdshow)
Up
Down
Video formats
176 x 144
352 x 288
128 x 96
704 x 576
Up
Down
OK
Cancel



Etat	Extension	Statuts utilisateurs	Files d'attente	Nom
Enregistré (inoccupé)	100	Disponible	Appel	BOB
Non enregistré	101	Disponible	Appel	ALICE
Non enregistré	102	Disponible	Appel	PAUL
Non enregistré	103	Disponible	Appel	

Using Wireshark, answers the following questions :

- **Capture, display and insert in this document the SIP sequence when the terminal is registrering at the server (user authentification and registration procedures).**



No. Time	Source	Destination	Protocol	Length	Info
[illegible]5134	192.168.137.101	192.168.137.101	SIP	596	Request: REGISTER sip:192.168.137.101:5060 (1 binding) |
4 3.646924	192.168.137.101	192.168.137.101	SIP	565	Status: 407 Proxy Authentication Required |
5 3.755234	192.168.137.101	192.168.137.101	SIP	812	Request: REGISTER sip:192.168.137.101:5060 (1 binding) |
146 3.856873	192.168.137.101	192.168.137.101	SIP	486	Status: 200 OK (1 binding) |
147 3.965212	192.168.137.101	192.168.137.101	SIP	640	Request: SUBSCRIBE sip:100@192.168.137.101:5060;transport=UDP |
148 4.066862	192.168.137.101	192.168.137.101	SIP	566	Status: 407 Proxy Authentication Required |
149 4.175455	192.168.137.101	192.168.137.101	SIP	874	Request: SUBSCRIBE sip:100@192.168.137.101:5060;transport=UDP |
150 4.276848	192.168.137.101	192.168.137.101	SIP	439	Status: 489 Event Package Not Supported |

- **Are the UserID and password transmitted in cleartext between your terminal and the server (yes/no) ? If not, explain how it does ?**

```
Message Header
  Via: SIP/2.0/UDP 0.0.0.0:62779;branch=z9hG4bK-d8754z-e513df58bd745346-1---d8754z-;rport
  Max-Forwards: 70
  Contact: <sip:100@0.0.0.0:62779;rinstance=e70f827219e8e9e8>
  To: "100"<sip:100@192.168.137.101:5060>
  From: "100"<sip:100@192.168.137.101:5060>;tag=48324d09
  Call-ID: OGUxMDQxZDlhZTllZDVhZWI5MjM2ZjhlNmJmOGIyYWE.
  [Generated Call-ID: OGUxMDQxZDlhZTllZDVhZWI5MjM2ZjhlNmJmOGIyYWE.]
  CSeq: 1 REGISTER
  Expires: 120
  Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REGISTER, SUBSCRIBE, NOTIFY, REFER, INFO, MESS
  Supported: replaces
  User-Agent: 3CXPhone 6.0.20943.0
```

No, we can only see the UserID which is same as extension number, but password is encrypted. We can't find it in cleartext.

– **What is the server port number ?**

Port 5060 is used

```
Internet Protocol Version 4, Src: 192.168.137.101, Dst: 37.171.169.38
User Datagram Protocol, Src Port: 64815, Dst Port: 5060
Session Initiation Protocol (REGISTER)
  Request-Line: REGISTER sip:37.171.169.38:5060 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 192.168.137.101:64815:branch=z9hG4bK-d8754z-3d6
```

– **What is the SIP method used to register to the server ?**

Client send first a "REGISTER" request, with important data like server address, its own IP, extension number, connection time, protocol (UDP)… Then the server returns a 407 Authentication required so the client send his credentials.

```
Session Initiation Protocol (REGISTER)
  Request-Line: REGISTER sip:37.171.169.38:5060 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 192.168.137.101:64815;branch=z9hG4bK-d8754z-3d62424bd0437302-1---d8754z-;rport
    Max-Forwards: 70
    Contact: <sip:101@37.173.92.71:50557;transport=UDP;rinstance=428366dbe9b286a7>
      Contact URI: sip:101@37.173.92.71:50557;transport=UDP;rinstance=428366dbe9b286a7
    To: "101"<sip:101@37.171.169.38:5060>
    From: "101"<sip:101@37.171.169.38:5060>;tag=1972f017
    Call-ID: NjNlYTQ2MTQyZTBiMGE5ZjU3ZWYzYTk2NTBhOWEyZDc.
    [Generated Call-ID: NjNlYTQ2MTQyZTBiMGE5ZjU3ZWYzYTk2NTBhOWEyZDc.]
    CSeq: 1 REGISTER
      Sequence Number: 1
      Method: REGISTER
    Expires: 120
    Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REGISTER, SUBSCRIBE, NOTIFY, REFER, INFO, MESSAGE
    Supported: replaces
    User-Agent: 3CXPhone 6.0.20943.0
    Content-Length: 0
```

– **What is the SIP code number used to register ?**

We can't find code number in the REGISTER request.

– **What is the SIP code number used by the server to confirm your registration ?**

Code "200" is returned by server to tell everything has been successful, like for HTTP protocol.

11 **Initiate** a call between 2 SIP Softphones (example BOB -> ALICE) through your SIP server using an extension number (example “100”). Using Wireshark, **capture** the sequence of SIP messages between the 2 terminals and the server from the beginning to the end of your voip session.

	Etat	Extension	Statuts utilisateurs	Files d'attente	Nom
	Enregistré (inoccupé)	100	Disponible	Appel	BOB
	Enregistré (inoccupé)	101	Disponible	Appel	ALICE

- **Using Wireshark options, display and insert in this document, the flow chart of the SIP signalling messages exchanged between the two terminals from Registration to End call. Use advanced wireshark options. Look at the**

sip

o.	Time	Source	Destination	Protocol	Length	Info
7	3.370624	192.168.137.101	192.168.137.101	SIP/SDP	1252	Request: INVITE sip:101@192.168.137.101:5060
267	3.479680	192.168.137.101	192.168.137.101	SIP	339	Status: 100 Trying |
373	3.836975	192.168.137.101	192.168.137.101	SIP	435	Status: 180 Ringing |
585	8.245267	192.168.137.101	192.168.137.101	SIP/SDP	809	Status: 200 OK |
586	8.354027	192.168.137.101	192.168.137.101	SIP	698	Request: ACK sip:101@192.168.137.101:5060 |
1414	13.007213	192.168.137.101	192.168.137.101	SIP	698	Request: BYE sip:101@192.168.137.101:5060 |
1535	13.115828	192.168.137.101	192.168.137.101	SIP	427	Status: 200 OK |

- **Are the source and destination port numbers used by the clients identical to the ones used with the direct voice call procedure (Partie 1)?**

  No, they are not the same.

  Port used in Part 1:

```
1414 13.007213    192.168.137.101    192.168.137.101    SIP    698 Request: BYE s

Frame 1414: 698 bytes on wire (5584 bits), 698 bytes captured (5584 bits) on interface \De
Null/Loopback
Internet Protocol Version 4, Src: 192.168.137.101, Dst: 192.168.137.101
User Datagram Protocol, Src Port: 62779, Dst Port: 5060
Session Initiation Protocol (BYE)
 > Request-Line: BYE sip:101@192.168.137.101:5060 SIP/2.0
 v Message Header
    > Via: SIP/2.0/UDP 192.168.137.101:62779;branch=z9hG4bK-d8754z-9a073d4bd54ca55f-1---d8
```

  Ports used during call:

```
758 11.704673    127.0.0.1    127.0.0.1    TCP    44 53
759 11.704776    127.0.0.1    127.0.0.1    TCP    52 53

Null/Loopback
Internet Protocol Version 4, Src: 127.0.0.1, Dst: 127.0.0.1
Transmission Control Protocol, Src Port: 53373, Dst Port: 5482, Seq: 1, Ack: 9,
   Source Port: 53373
   Destination Port: 5482
```

- **Are the voice IP packets passing through the SIP server or directly between the 2 clients** ?

  Voice IP packets are communicated directly between both clients.

- **What is the role of the SIP server during this test** ?

  The SIP server is used to manage sessions and allow connexion between two phones.

Les parties 2, 3, 4, et 5 ci-dessous vous permettront d'implémenter et tester des fonctionnalités CTI (couplage téléphonie et informatique) :

Partie 2 : Call transfert and Call redirection

12. BOB (100), ALICE (101) and PAUL (102) are logged on to your SIP server. Establish a voice call between BOB and ALICE, and after ALICE hangs-up, she **Transfers** the active call to PAUL.

<table>
<tr><th>Source</th><th>Destination</th><th>Protocol</th><th>Length</th><th>Info</th></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP/SDP</td><td>1060</td><td>Request: INVITE sip:101@192.168.137.174:59083;rinstance=5ed004e40b</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP</td><td>464</td><td>Status: 180 Ringing |</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP/SDP</td><td>996</td><td>Status: 200 OK |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP</td><td>502</td><td>Request: ACK sip:101@192.168.137.174:59083;rinstance=5ed004e40b2f4</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP/SDP</td><td>1063</td><td>Request: INVITE sip:100@192.168.137.101:5060, in-dialog |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP</td><td>362</td><td>Status: 100 Trying |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP/SDP</td><td>1014</td><td>Status: 200 OK |</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP</td><td>496</td><td>Request: ACK sip:100@192.168.137.101:5060 |</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP</td><td>587</td><td>Request: REFER sip:100@192.168.137.101:5060, in-dialog |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP</td><td>458</td><td>Status: 202 Accepted |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP/si…</td><td>613</td><td>Request: NOTIFY sip:101@192.168.137.174:59083;rinstance=5ed004e40b</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP</td><td>653</td><td>Request: INVITE sip:102@192.168.137.78:59654;rinstance=2b163319c4b</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP</td><td>653</td><td>Request: INVITE sip:102@192.168.137.78:43974;rinstance=9048adf5f50</td></tr>
<tr><td>192.168.137.78</td><td>192.168.137.101</td><td>SIP</td><td>347</td><td>Status: 100 Trying |</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP</td><td>459</td><td>Status: 200 OK |</td></tr>
<tr><td>192.168.137.78</td><td>192.168.137.101</td><td>SIP</td><td>575</td><td>Status: 180 Ringing |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP</td><td>653</td><td>Request: INVITE sip:102@192.168.137.78:43974;rinstance=9048adf5f50</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP</td><td>653</td><td>Request: INVITE sip:102@192.168.137.78:43974;rinstance=9048adf5f50</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP</td><td>653</td><td>Request: INVITE sip:102@192.168.137.78:43974;rinstance=9048adf5f50</td></tr>
<tr><td>192.168.137.78</td><td>192.168.137.101</td><td>SIP/SDP</td><td>1038</td><td>Status: 200 OK |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP/si…</td><td>622</td><td>Request: NOTIFY sip:101@192.168.137.174:59083;rinstance=5ed004e40b</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP</td><td>459</td><td>Status: 200 OK |</td></tr>
<tr><td>192.168.137.174</td><td>192.168.137.101</td><td>SIP</td><td>536</td><td>Request: BYE sip:100@192.168.137.101:5060 |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP/SDP</td><td>798</td><td>Request: ACK sip:102@192.168.137.78:59654;rinstance=2b163319c4b7a2</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.174</td><td>SIP</td><td>437</td><td>Status: 200 OK |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP</td><td>653</td><td>Request: INVITE sip:102@192.168.137.78:43974;rinstance=9048adf5f50</td></tr>
<tr><td>192.168.137.78</td><td>192.168.137.101</td><td>SIP</td><td>566</td><td>Request: BYE sip:100@192.168.137.101:5060 |</td></tr>
<tr><td>192.168.137.101</td><td>192.168.137.78</td><td>SIP</td><td>428</td><td>Status: 200 OK |</td></tr>
</table>

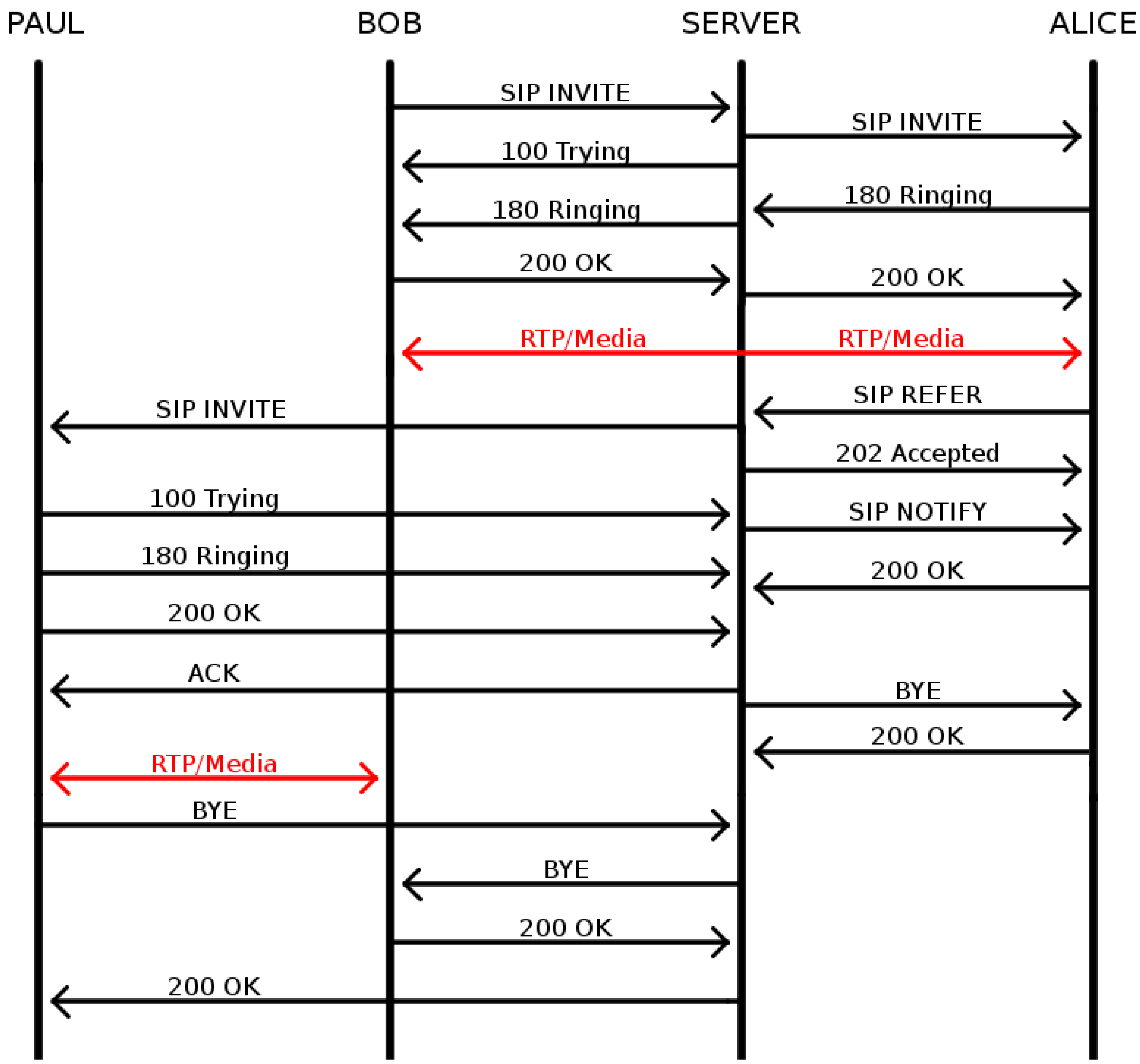
- **What is the SIP message/method used to do this call transfer ?**

  During the call, ALICE sends a request REFER to the server. Then the server requests INVITE to PAUL and tell ALICE the REFER is OK (200). Then PAUL is Trying (100) to call so ALICE receives a SIP NOTIFY. After that, PAUL and ALICE both send OK (200) to the server to tell they're ready. So the server transmits an ACK to PAUL and BYE to ALICE. ALICE answers OK (200). Finally, the call between PAUL and BOB is launched.

- **What is the SIP code used ?**

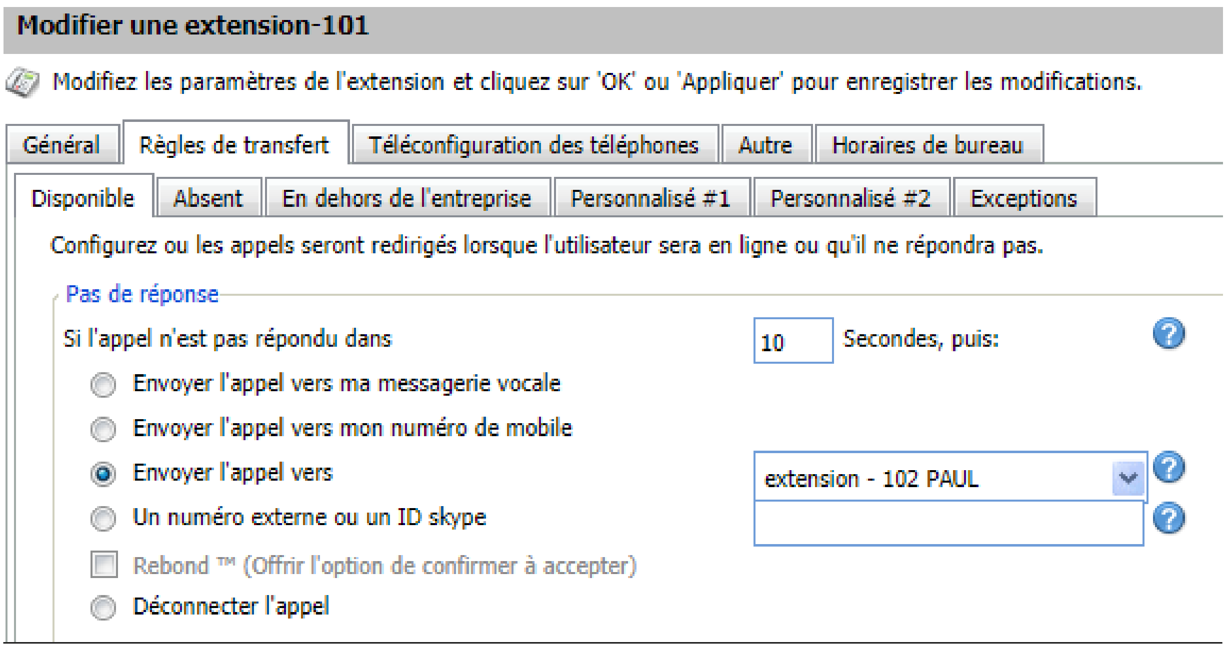
  Server sends "202 Accepted" to ALICE to notify her that the refer has been accepted, PAUL sends "100 Trying" and "180 Ringing" to the server to initiate the second call and PAUL and ALICE send "200 OK" to tell they are ready.

Below there is a diagram showing SIP packets sent from the first call to the end of the second call.



13. Modify the SIP server and the ALICE profile, to automatically redirect an incoming call to ALICE to PAUL when ALICE is not answering after 10 secondes ringing.

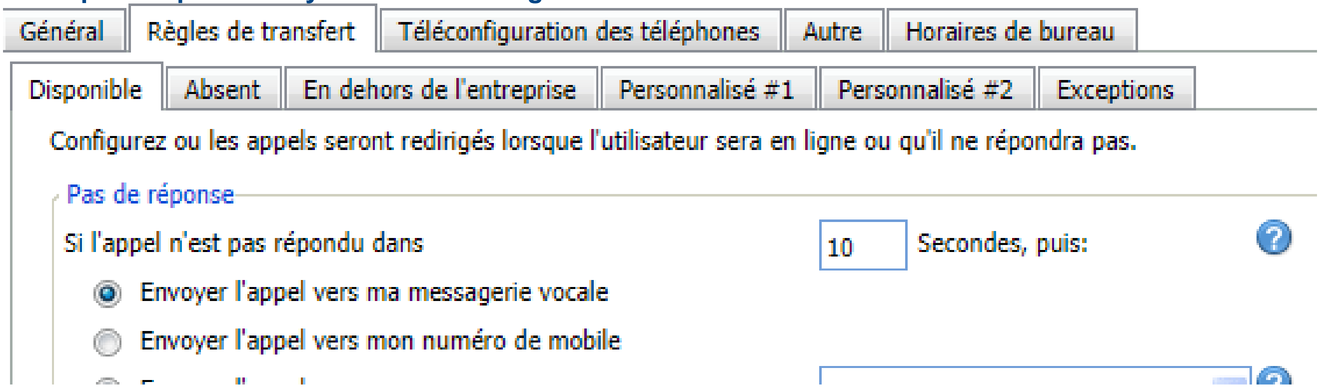
- capture the server configuration screen for setting up this functionality. Insert this picture in this document.



Partie 3 : Voice Messaging and Voice Mail

14. **modify** the SIP server and ALICE profile to redirect an inbound call to the voice messaging system if ALICE is not responding after 10 seconds.

- **capture a picture of your screen configuration and insert it in this document.**



- **Capture the SIP message/method sequence of this scenario and insert it in this document.**

Time	Source	Destination	Protocol	Length	Info
2.868468	192.168.137.101	192.168.137.174	SIP/SDP	1060	Request: INVITE sip:101@192.168.137.174:64050;rins
2.988822	192.168.137.174	192.168.137.101	SIP	464	Status: 180 Ringing |
12.893336	192.168.137.101	192.168.137.174	SIP	515	Request: CANCEL sip:101@192.168.137.174:64050;rins
13.012160	192.168.137.174	192.168.137.101	SIP	459	Status: 200 OK |
13.012160	192.168.137.174	192.168.137.101	SIP	406	Status: 487 Request Terminated |
13.013179	192.168.137.101	192.168.137.174	SIP	420	Request: ACK sip:101@192.168.137.174:64050;rinstan

- **What is the SIP message and code number used by the server to indicate that ALICE is not available ?**

  The SIP message from ALICE is a CANCEL request, where BOB answer with a 200 OK and 487 Request Terminated packets.

- **What is the phone number to dial to access to the voice messages recorded in the Server ?**

  This is 999 (and 99 if we had only 2 digits extensions). We have then to enter the PIN written in extension's profile in the server.

- **With wireshark, identify the network protocols that are used to access and listen to this voice messages ? Do you use RTSP/RTP/RTCP (yes/no) ?**

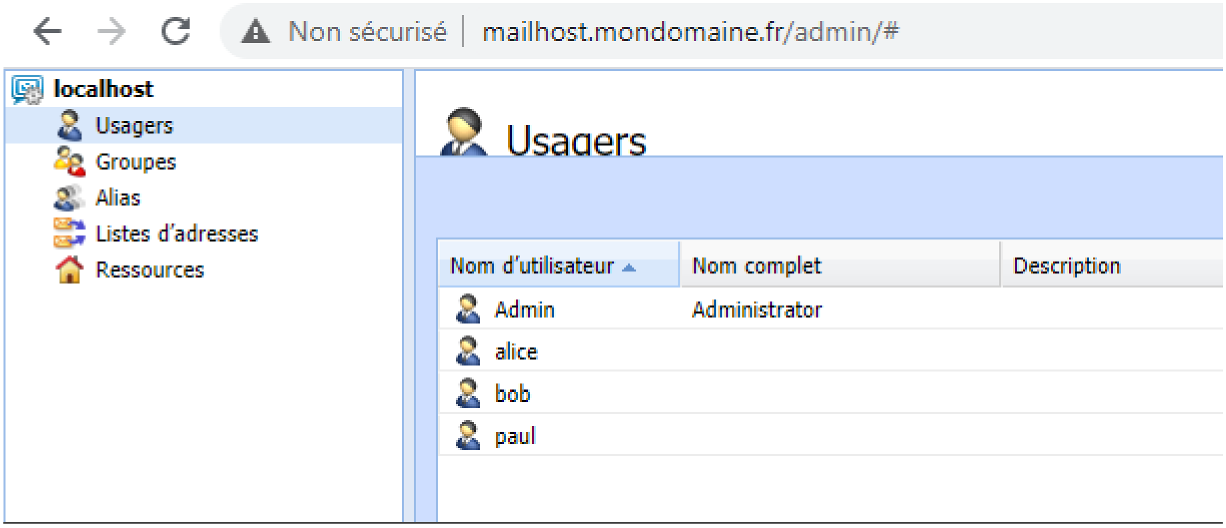
  Based on Wireshark, only RTP is used. We couldn't find RTSP or RTCP packet.

	Time	Source	Destination	Protocol	Length	Info
2197	53.223780	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13873, Time=
2198	53.231681	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20253, Time=
2199	53.238613	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13874, Time=
2200	53.251740	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20254, Time=
2201	53.266687	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20255, Time=
2202	53.272917	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13875, Time=
2203	53.279460	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13876, Time=
2204	53.286679	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20256, Time=
2205	53.306200	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13877, Time=
2206	53.306554	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20257, Time=
2207	53.318307	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13878, Time=
2208	53.331623	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20258, Time=
2209	53.337477	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13879, Time=
2210	53.346562	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20259, Time=
2211	53.363453	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13880, Time=
2212	53.371582	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20260, Time=
2213	53.380570	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13881, Time=
2214	53.386578	192.168.137.174	192.168.137.101	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x3D6C, Seq=20261, Time=
2215	53.403029	192.168.137.101	192.168.137.174	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x2213, Seq=13882, Time=

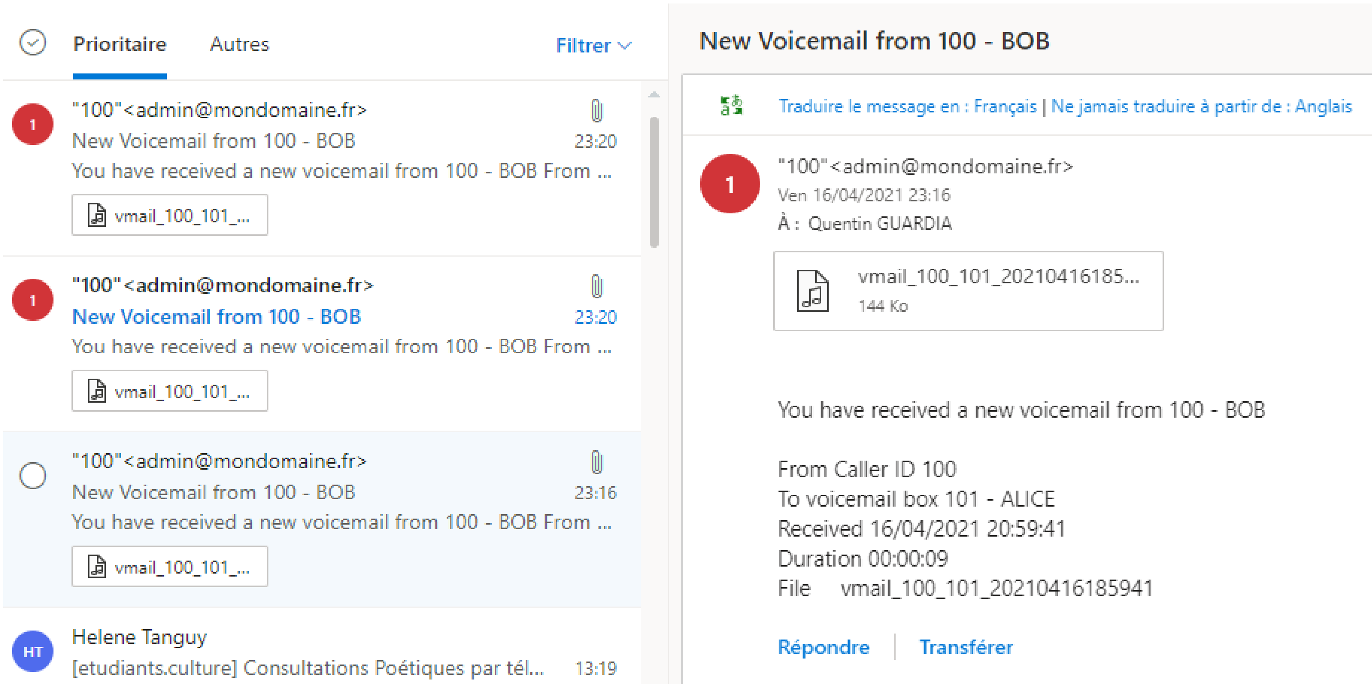
15. Install and configure an Email Server and configure it with 4 accounts (ADMIN, BOB, ALICE & PAUL)

In the VOIP Server, **Configure** the ALICE profile to automatically receive an email notification with the attached voice message using an external SMTP server (your webmail or your university email account)

We used kerio mailserver:



- **capture a screen picture of your email message with the attached voice mail. Insert this picture in this document.**

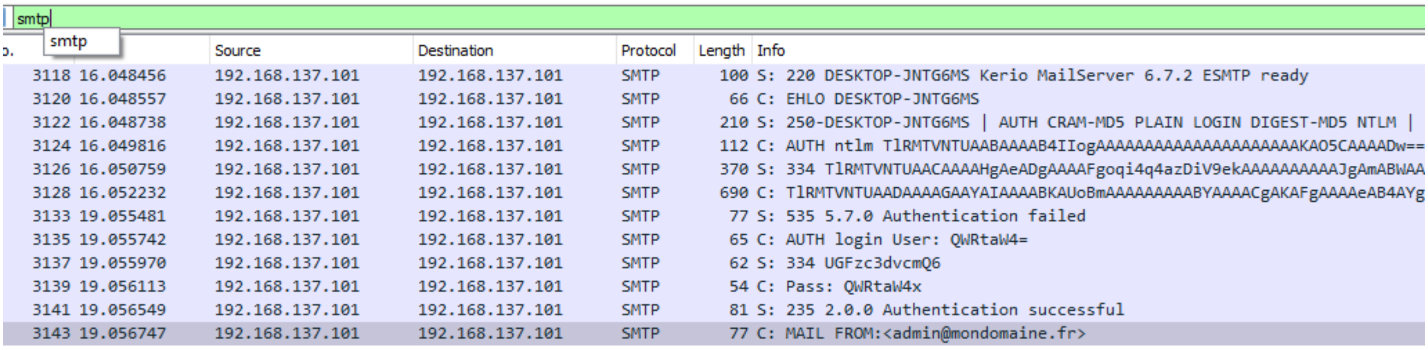


- **What is the audio format of this voice message ?**

  The audio file is a .wav file.

- **What is the email protocol to send this message to the client (POP3, IMAP4, SMTP) ?**

  The protocol SMTP is used a shown below:



No.	Time	Source	Destination	Protocol	Length	Info
3118	16.048456	192.168.137.101	192.168.137.101	SMTP	100	S: 220 DESKTOP-JNTG6MS Kerio MailServer 6.7.2 ESMTP ready
3120	16.048557	192.168.137.101	192.168.137.101	SMTP	66	C: EHLO DESKTOP-JNTG6MS
3122	16.048738	192.168.137.101	192.168.137.101	SMTP	210	S: 250-DESKTOP-JNTG6MS | AUTH CRAM-MD5 PLAIN LOGIN DIGEST-MD5 NTLM |
3124	16.049816	192.168.137.101	192.168.137.101	SMTP	112	C: AUTH ntlm TlRMTVNTUAABAAAAB4IIogAAAAAAAAAAAAAAAAAAAAAAKAO5CAAAADw==
3126	16.050759	192.168.137.101	192.168.137.101	SMTP	370	S: 334 TlRMTVNTUAACAAAAHgAeADgAAAAFgoqi4q4azDiV9ekAAAAAAAAAAJgAmABWAA
3128	16.052232	192.168.137.101	192.168.137.101	SMTP	690	C: TlRMTVNTUAADAAAAGAAYAIAAAABKAUoBmAAAAAAAAABYAAAACgAKAFgAAAAeAB4AYg
3133	19.055481	192.168.137.101	192.168.137.101	SMTP	77	S: 535 5.7.0 Authentication failed
3135	19.055742	192.168.137.101	192.168.137.101	SMTP	65	C: AUTH login User: QWRtaW4=
3137	19.055970	192.168.137.101	192.168.137.101	SMTP	62	S: 334 UGFzc3dvcmQ6
3139	19.056113	192.168.137.101	192.168.137.101	SMTP	54	C: Pass: QWRtaW4x
3141	19.056549	192.168.137.101	192.168.137.101	SMTP	81	S: 235 2.0.0 Authentication successful
3143	19.056747	192.168.137.101	192.168.137.101	SMTP	77	C: MAIL FROM:<admin@mondomaine.fr>

Partie 4 : Interactive Voice Assistant

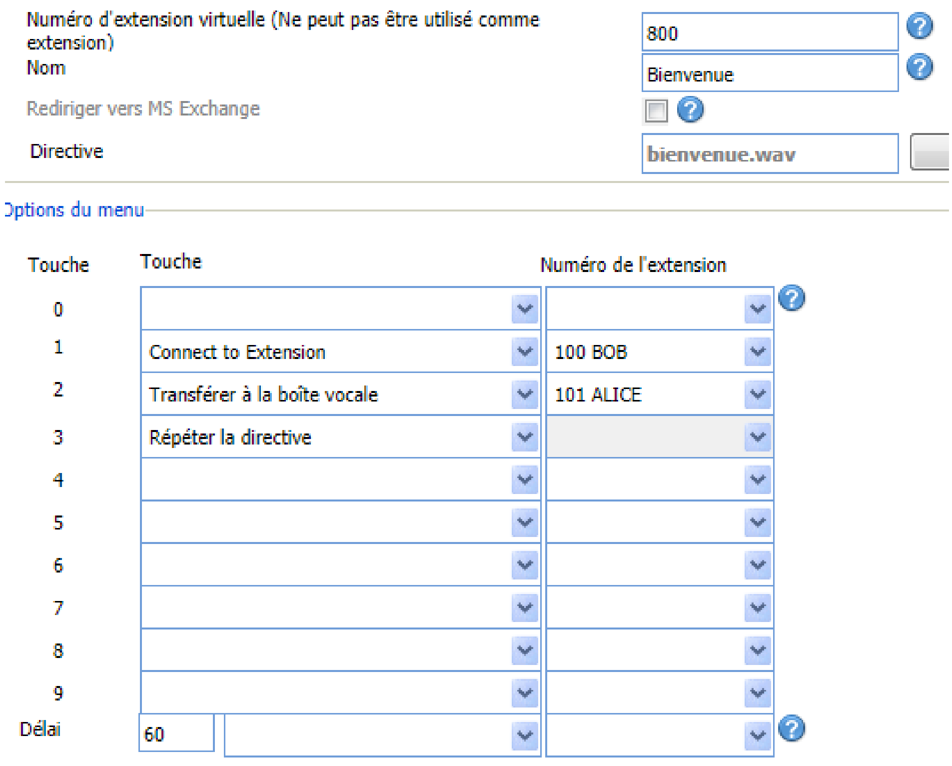
16. Modify ALICE profile to activate an Interactive Voice Assistant when an incoming call is not answered after 10 sec. Record a "Welcome voice message" in audio WAVE format that propose the following interactive scenario to the caller:

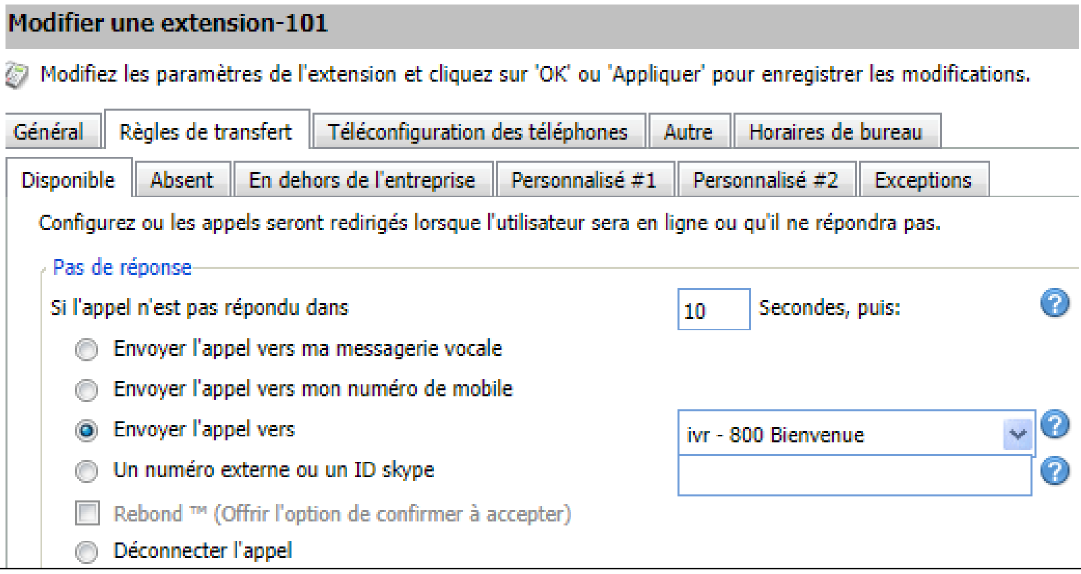
Please dial #1 : to transfer your call to BOB

Please dial #2 : to transfer your call to the voice messaging

Please dial #3 : to repeat listen this message again

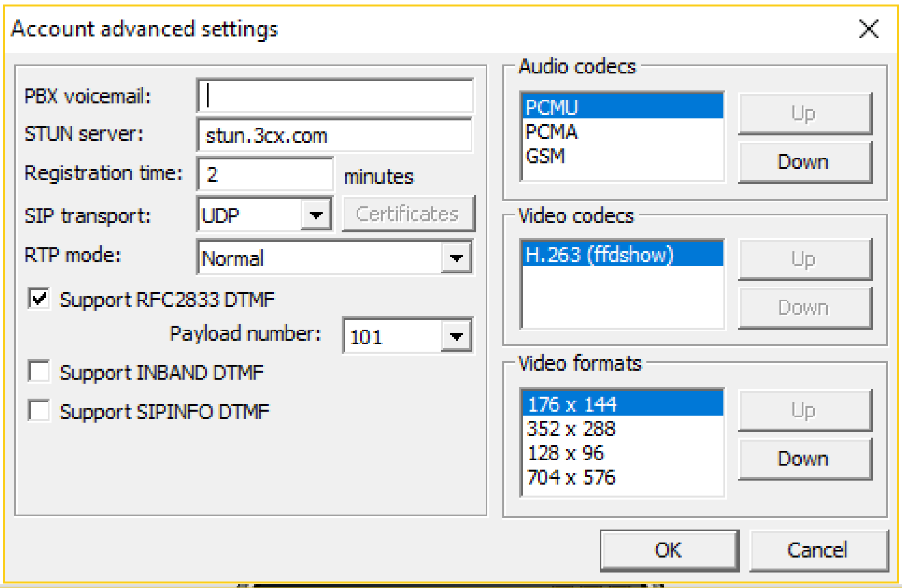
- **capture a screen picture of your email message with the attached voice mail. Insert this picture in this document.**

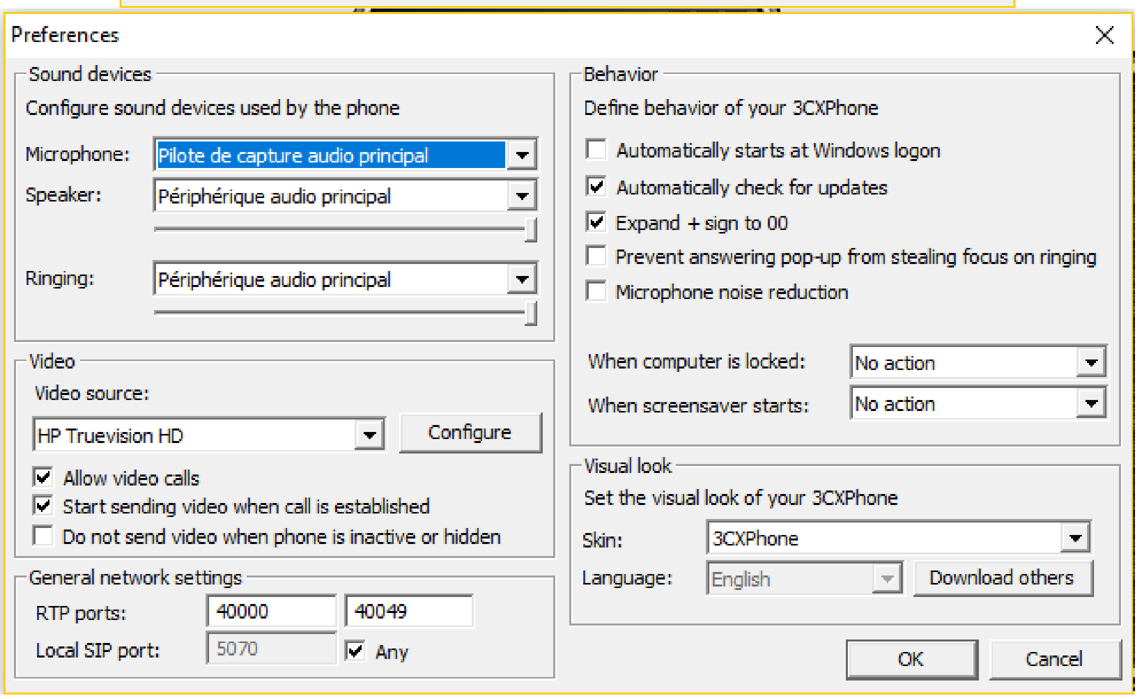




Partie 5 : VOIP and VideoConferencing with 3CX clients and server

17 Configure the **3CX VOIP clients** to use Video-Conferencing without any security options.





- **What is the video codec type used ? What is the video resolution (pixels) ?**

  The H.263 ffdshow codec is used, with a resolution of 176*144 pixels.

- **What is the audio codec type used ?**

  PCMA audio codec is used.

- **How many sockets (TCP and UDP ports) at the client side are used for this videoconference ?**

  Ports from 40 000 to 40 049 are used during videoconference, looking at the preferences window and Wireshark

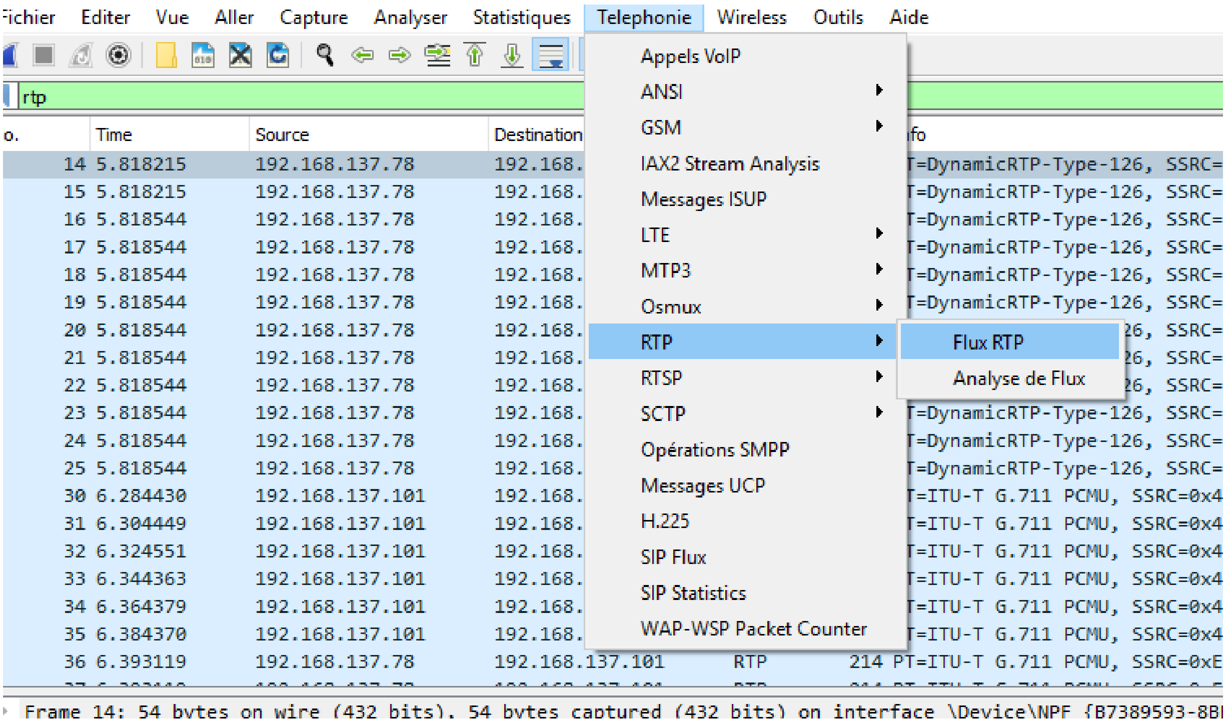
Partie 6 : VOIP and Security with 3CX

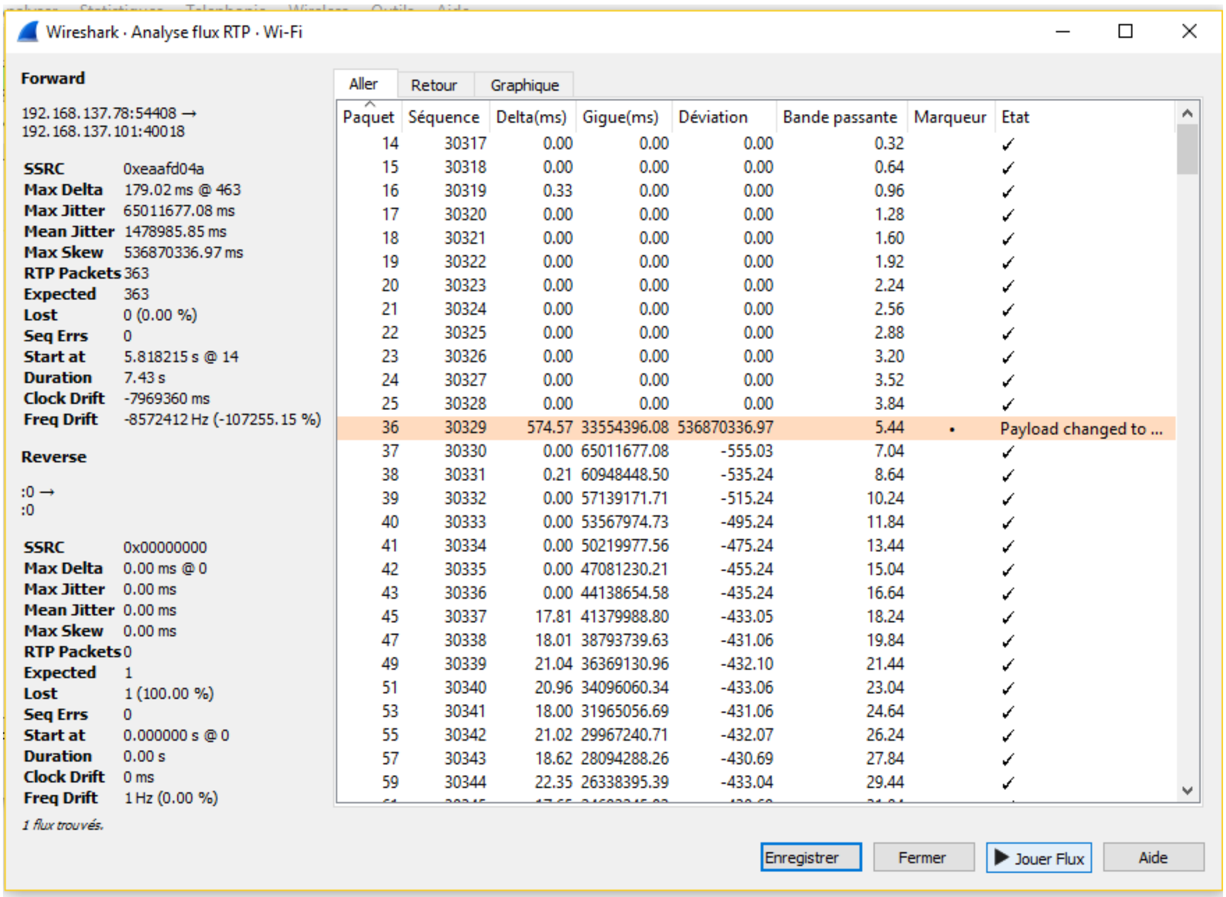
18. VOIP vulnerabilities :

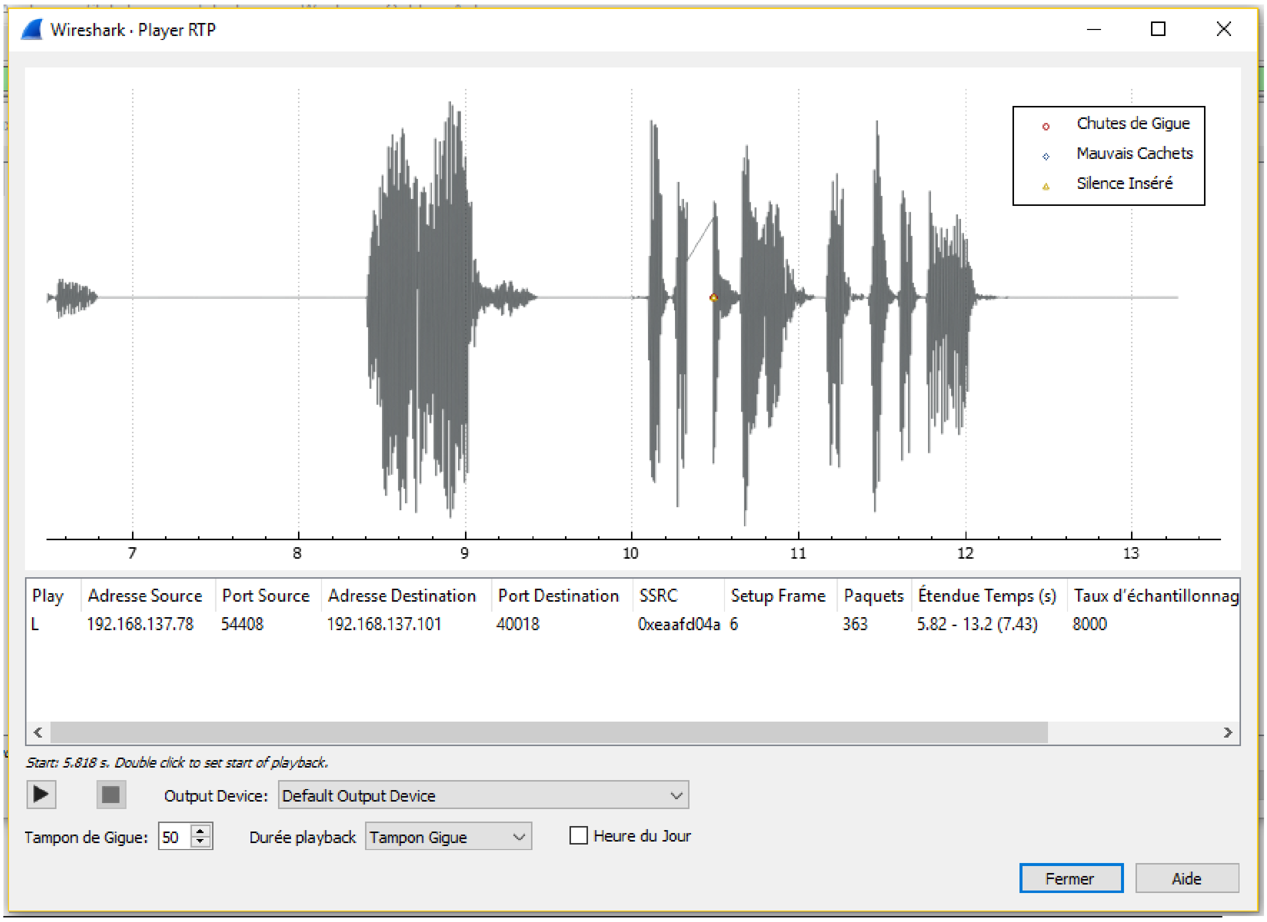
- search on the internet about the different vulnerabilities and threats on TOIP services. List these on a table below.

Man-In-The-Middle and replay
Denial of Service
Hijacking and theft data which can cost lots of fees when hacker use them to do expansive acts
Intrusion on the organism, to put viruses and trojan
Internet Bound Traffic: packets can be sniffed
Spoofing: a hacker can act with society’s properties to ask credentials to customers
Call tampering: the hacker send lots of datas to affect clarity of call
Eavesdropping (ex:VOMIT,...)
Spams (ex:SPIT,...)
Gateway, Firewall, Application, Phones can be also victims of vulnerabilities and so on...

- initiate a voice call (without security using PCMA or PCMU codec) and capture the voice packets with wireshark. Record the voice packet into a file with .wav file extension. Play this file with your media player to listen to the conversation.

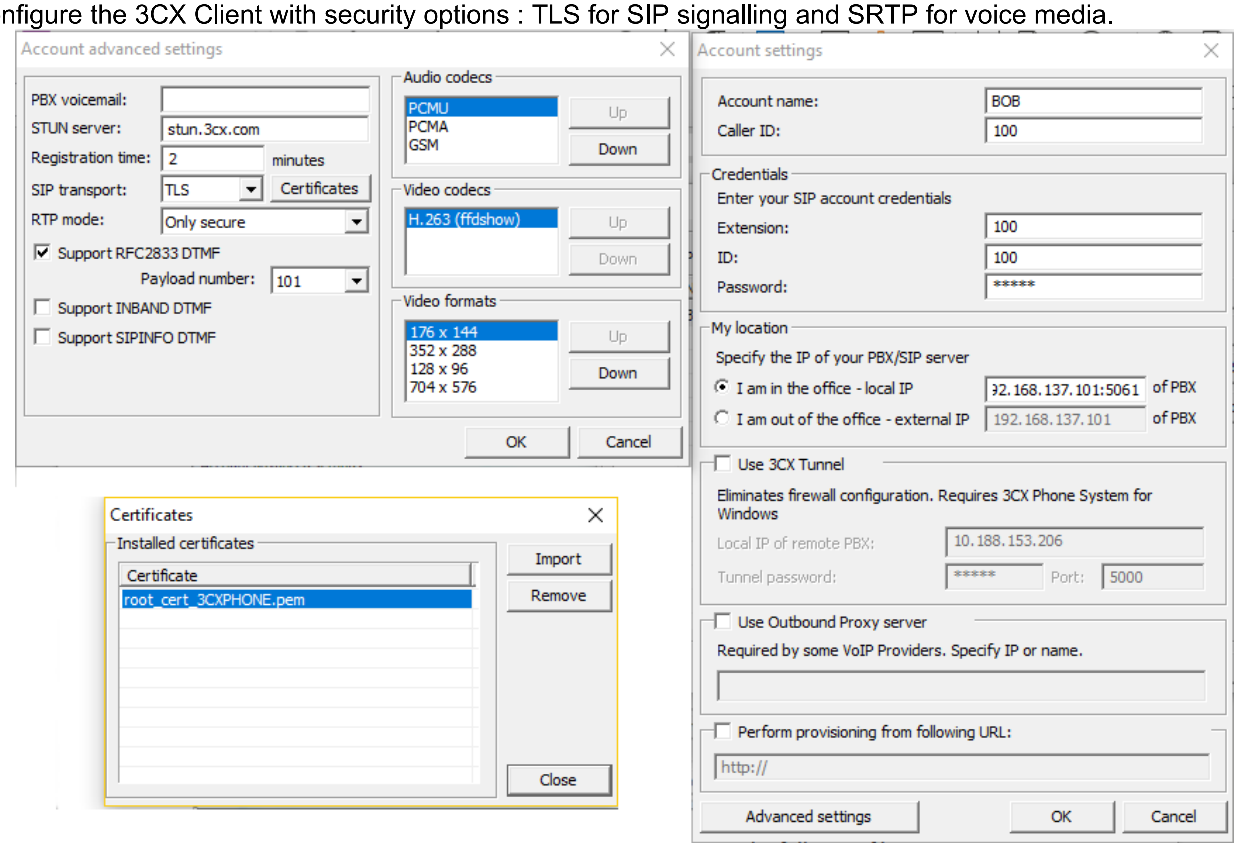

Fichier Editer Vue Aller Capture Analyser Statistiques Telephonie Wireless Outils Aide
rtp
Time Source Destination
Appels VoIP
ANSI
GSM
IAX2 Stream Analysis
Messages ISUP
LTE
MTP3
Osmux
RTP
Flux RTP
Analyse de Flux
RTSP
SCTP
Opérations SMPP
Messages UCP
H.225
SIP Flux
SIP Statistics
WAP-WSP Packet Counter
14 5.818215 192.168.137.78 192.168.
15 5.818215 192.168.137.78 192.168.
16 5.818544 192.168.137.78 192.168.
17 5.818544 192.168.137.78 192.168.
18 5.818544 192.168.137.78 192.168.
19 5.818544 192.168.137.78 192.168.
20 5.818544 192.168.137.78 192.168.
21 5.818544 192.168.137.78 192.168.
22 5.818544 192.168.137.78 192.168.
23 5.818544 192.168.137.78 192.168.
24 5.818544 192.168.137.78 192.168.
25 5.818544 192.168.137.78 192.168.
30 6.284430 192.168.137.101 192.168.
31 6.304449 192.168.137.101 192.168.
32 6.324551 192.168.137.101 192.168.
33 6.344363 192.168.137.101 192.168.
34 6.364379 192.168.137.101 192.168.
35 6.384370 192.168.137.101 192.168.
36 6.393119 192.168.137.78 192.168.137.101 RTP 214 PT=ITU-T G.711 PCMU, SSRC=0xE
T=DynamicRTP-Type-126, SSRC=
T=ITU-T G.711 PCMU, SSRC=0x4
Frame 14: 54 bytes on wire (432 bits), 54 bytes captured (432 bits) on interface \Device\NPF_{B7389593-8B


Wireshark · Analyse flux RTP · Wi-Fi
Forward
192.168.137.78:54408 →
192.168.137.101:40018
SSRC 0xeaafd04a
Max Delta 179.02 ms @ 463
Max Jitter 65011677.08 ms
Mean Jitter 1478985.85 ms
Max Skew 536870336.97 ms
RTP Packets 363
Expected 363
Lost 0 (0.00 %)
Seq Errs 0
Start at 5.818215 s @ 14
Duration 7.43 s
Clock Drift -7969360 ms
Freq Drift -8572412 Hz (-107255.15 %)
Reverse
:0 →
:0
SSRC 0x00000000
Max Delta 0.00 ms @ 0
Max Jitter 0.00 ms
Mean Jitter 0.00 ms
Max Skew 0.00 ms
RTP Packets 0
Expected 1
Lost 1 (100.00 %)
Seq Errs 0
Start at 0.000000 s @ 0
Duration 0.00 s
Clock Drift 0 ms
Freq Drift 1 Hz (0.00 %)
1 flux trouvés.
Aller Retour Graphique
Paquet Séquence Delta(ms) Gigue(ms) Déviation Bande passante Marqueur Etat
14 30317 0.00 0.00 0.00 0.32 ✓
15 30318 0.00 0.00 0.00 0.64 ✓
16 30319 0.33 0.00 0.00 0.96 ✓
17 30320 0.00 0.00 0.00 1.28 ✓
18 30321 0.00 0.00 0.00 1.60 ✓
19 30322 0.00 0.00 0.00 1.92 ✓
20 30323 0.00 0.00 0.00 2.24 ✓
21 30324 0.00 0.00 0.00 2.56 ✓
22 30325 0.00 0.00 0.00 2.88 ✓
23 30326 0.00 0.00 0.00 3.20 ✓
24 30327 0.00 0.00 0.00 3.52 ✓
25 30328 0.00 0.00 0.00 3.84 ✓
36 30329 574.57 33554396.08 536870336.97 5.44 • Payload changed to ...
37 30330 0.00 65011677.08 -555.03 7.04 ✓
38 30331 0.21 60948448.50 -535.24 8.64 ✓
39 30332 0.00 57139171.71 -515.24 10.24 ✓
40 30333 0.00 53567974.73 -495.24 11.84 ✓
41 30334 0.00 50219977.56 -475.24 13.44 ✓
42 30335 0.00 47081230.21 -455.24 15.04 ✓
43 30336 0.00 44138654.58 -435.24 16.64 ✓
45 30337 17.81 41379988.80 -433.05 18.24 ✓
47 30338 18.01 38793739.63 -431.06 19.84 ✓
49 30339 21.04 36369130.96 -432.10 21.44 ✓
51 30340 20.96 34096060.34 -433.06 23.04 ✓
53 30341 18.00 31965056.69 -431.06 24.64 ✓
55 30342 21.02 29967240.71 -432.07 26.24 ✓
57 30343 18.62 28094288.26 -430.69 27.84 ✓
59 30344 22.35 26338395.39 -433.04 29.44 ✓
Enregistrer Fermer Jouer Flux Aide

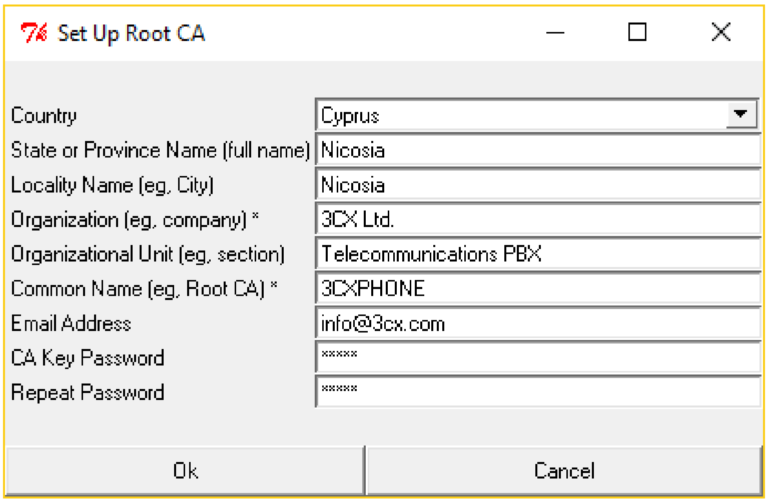


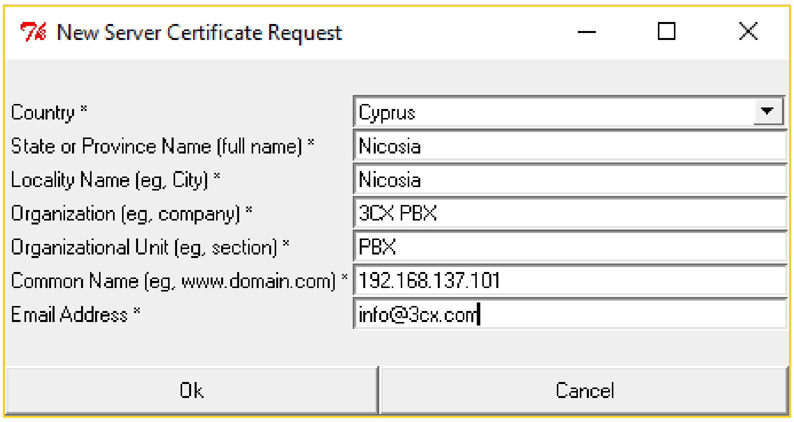
We want now to secure the TOIP communications by configuring both the VOIP clients and the server to establish a secured (encrypted) authentication between the clients and the server AND a secured (encrypted) voice call between the two VOIP client.

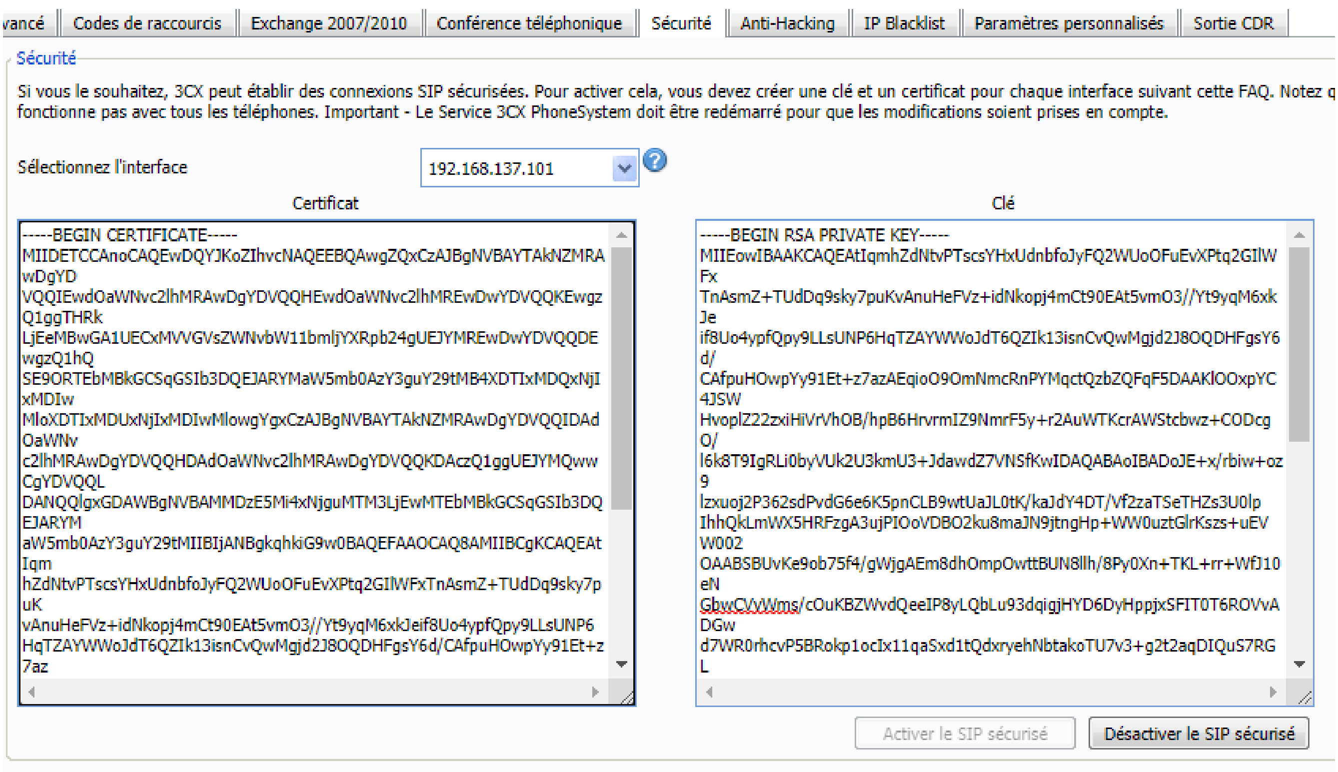
Configure the 3CX Client with security options : TLS for SIP signalling and SRTP for voice media.

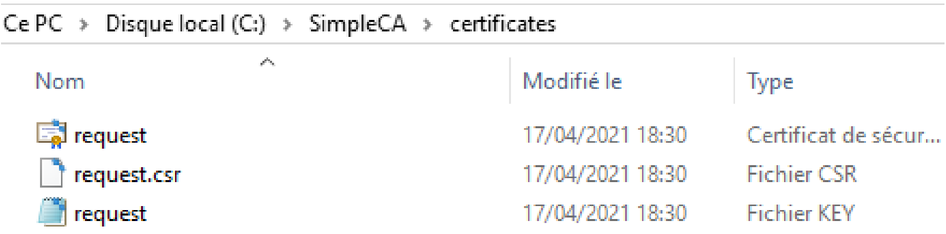


Configure the 3CX Server with secure SIP (SIPS) using “SimpleCA” to generate an X.509 SSL certificate for this server. Use the instructions presented in the “security user guide”.



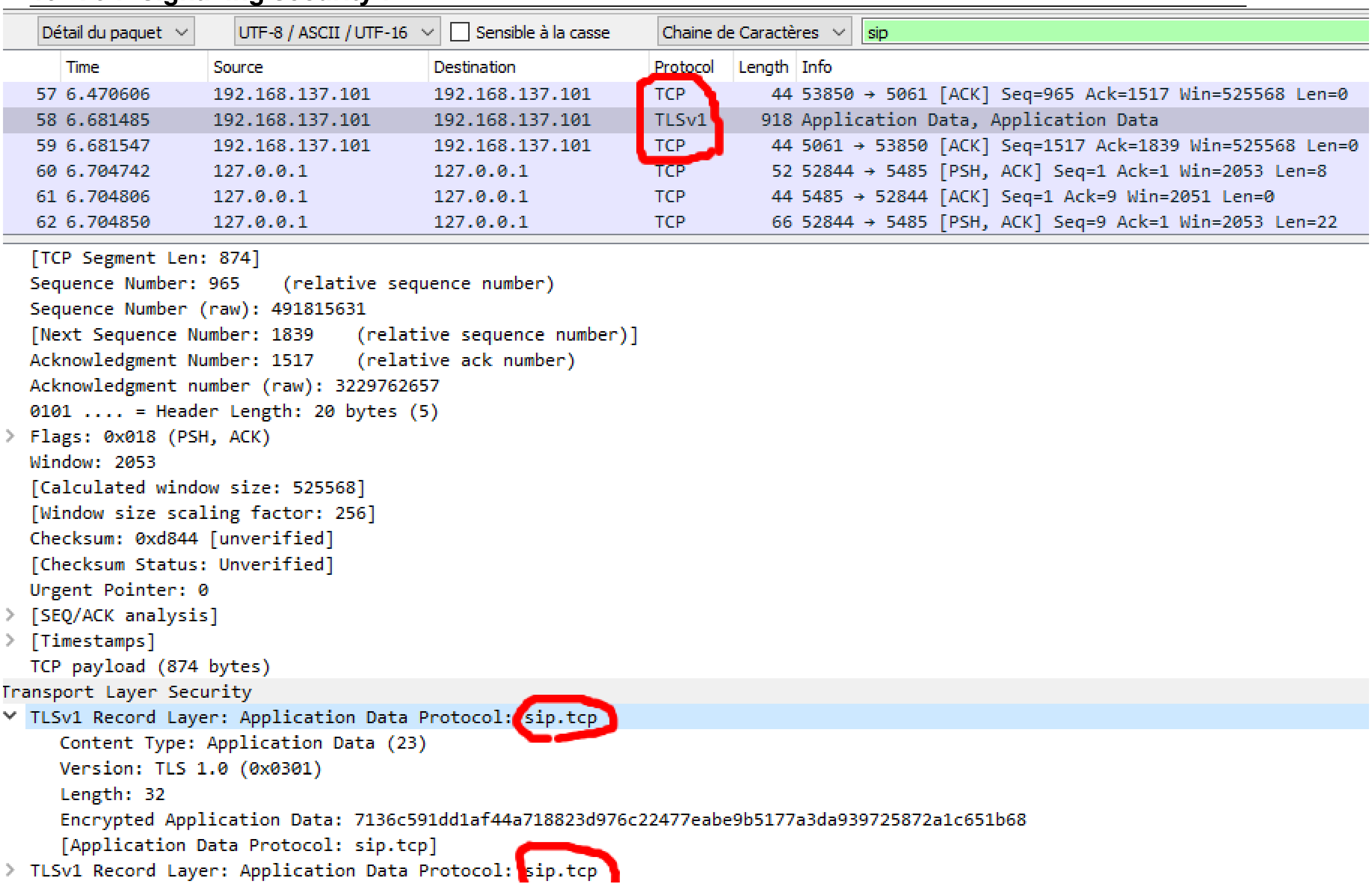






Using Wireshark, answer the following questions.

19. VoIP signalling security :



- **Explain How the SIP messages are secured between clients and Server ? and between Client and Client ?**

  Between clients and server, SIPS is used to secure SIP. It's a SIP version using TSL/SSL. Between two clients, SRTP is used to ensure media exchange. On the same way, it's RTP with data encryption. We explain in the next answers how it works.

- **What is the transport protocol used (TCP/UDP) with SIP security ? Explain Why, you don't have the same protocole than the non-secured SIP call ?**

  TLSv1 protocol is used. This is a protocol only used for secured exchange, a special handshake is done at the beginning. It's not the case for a non-secured communication, where this protocol is not asked. We could also have used a SSL tunnel.

- **Explain what is SIPS (SIP over SSL)**

  SIPS gives more security during a client/server exchange than stndard SIP. First, client and server check each other certificate, then they exchange key, and to finish they choose the cipher mode. After that, the SIP/RTP exchanges are fully encrypted, using the keys and the cipher mode.

- **What is the port number of SIPS ?**

  SIPS requires port 5061 instead of 5060

20. VoIP data security :

- **Are the default voice/RTP packets encrypted (yes/no) ?**

  No, as we saw before, RTP packets are not encrypted by default and we can listen to the call using Wireshark

- **what is the encryption algorithm used to secure voice data ?**

  AES in CBC mode has been used (cf screen below)

- **Explain what is Secure RTP (SRTP). Find the IETF RFC number for SRTP.**

  RFC 3711 explains what Secure Real-Time Transfer Protocol is. This is a protocol based on RTP, with more security strength, like message authentication, confidentiality or protection against replay.

- **What is the transport protocol used (TCP/UDP) with SIP security ?**

  We can see TCP packets on Wireshark. It must be the SRTP data

- **How the SRTP secret key is exchanged between the two systems ?**

  Key management protocol allows secret key to be known by both systems. Actually, one single master key is exchanged, then keys are derived from this master key on both sides.

- **Insert the pictures of the captured packets with Wireshark to justify your answers.**

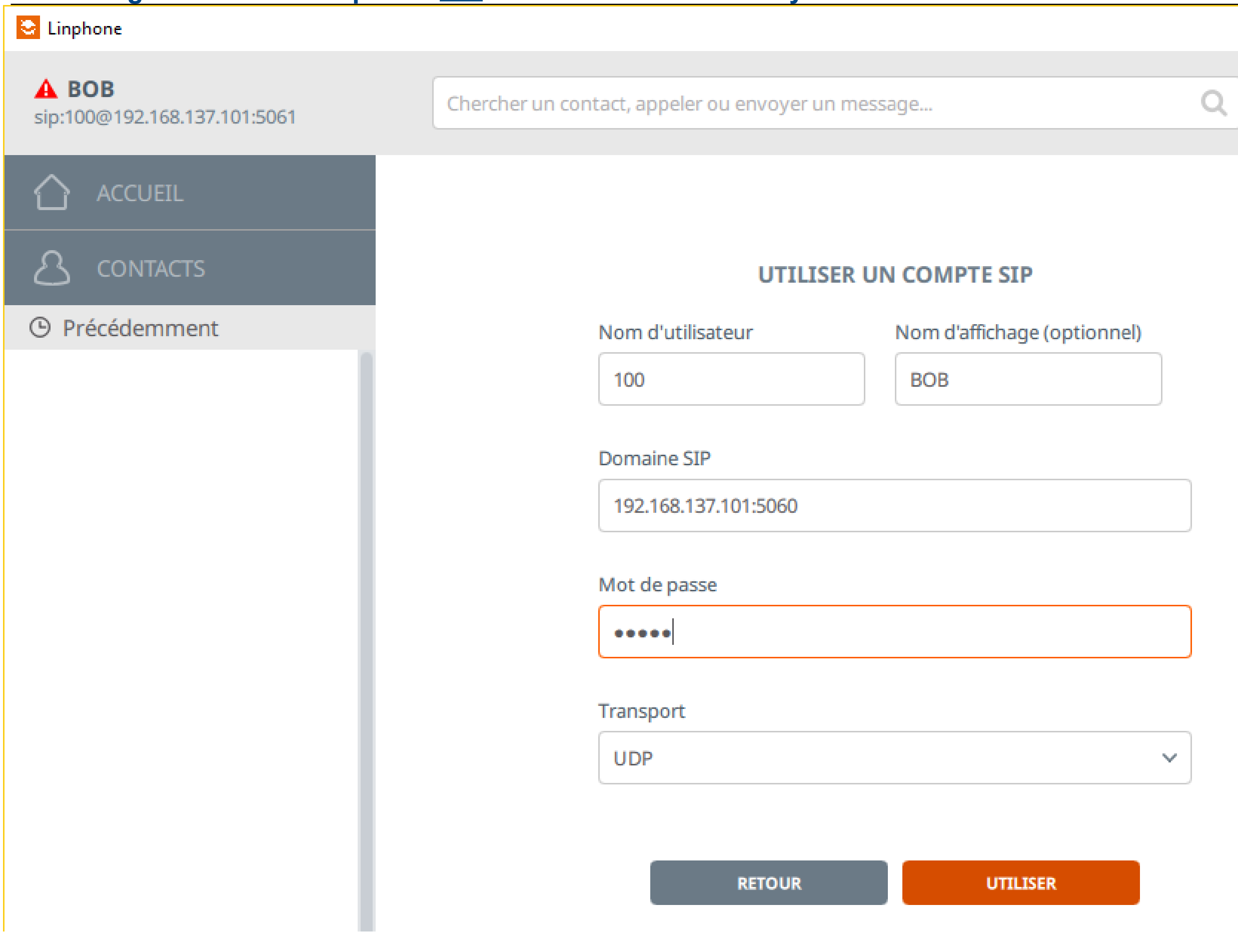
	Time	Source	Destination	Protocol	Length	Info
43	6.259049	192.168.137.101	192.168.137.101	TLSv1	144	Client Hello
44	6.259102	192.168.137.101	192.168.137.101	TCP	44	5061 → 53850 [ACK] Seq=1 Ack=101 Win=525568 Len=0
45	6.260432	192.168.137.101	192.168.137.101	TLSv1	915	Server Hello, Certificate, Server Hello Done
46	6.260506	192.168.137.101	192.168.137.101	TCP	44	53850 → 5061 [ACK] Seq=101 Ack=872 Win=524544 Len=0
47	6.263255	192.168.137.101	192.168.137.101	TLSv1	242	Client Key Exchange, Change Cipher Spec, Encrypted Handshake Message
48	6.263319	192.168.137.101	192.168.137.101	TCP	44	5061 → 53850 [ACK] Seq=872 Ack=299 Win=525312 Len=0
49	6.267948	192.168.137.101	192.168.137.101	TLSv1	103	Change Cipher Spec, Encrypted Handshake Message
50	6.268013	192.168.137.101	192.168.137.101	TCP	44	53850 → 5061 [ACK] Seq=299 Ack=931 Win=524544 Len=0
51	6.268433	192.168.137.101	192.168.137.101	TLSv1	710	Application Data, Application Data
52	6.268472	192.168.137.101	192.168.137.101	TCP	44	5061 → 53850 [ACK] Seq=931 Ack=965 Win=524544 Len=0

```
    Version: TLS 1.0 (0x0301)
    Length: 74
  v Handshake Protocol: Server Hello
      Handshake Type: Server Hello (2)
      Length: 70
      Version: TLS 1.0 (0x0301)
    > Random: 607b1849c9627f1541d83831550b66c85f691023383e37dcf0153d36c6f0eb90
      Session ID Length: 32
      Session ID: dcec888c40d1a493f43444b55d6f9fbdfe9e7498557986fa915528f26595842f
      Cipher Suite: TLS_RSA_WITH_AES_256_CBC_SHA (0x0035)
      Compression Method: null (0)
v TLSv1 Record Layer: Handshake Protocol: Certificate
    Content Type: Handshake (22)
    Version: TLS 1.0 (0x0301)
    Length: 778
  v Handshake Protocol: Certificate
      Handshake Type: Certificate (11)
```

	Time	Source	Destination	Protocol	Length	Info
46	6.260506	192.168.137.101	192.168.137.101	TCP	44	53850 → 5061 [ACK] Seq=101 Ack=872 Win=524544 Len=0
47	6.263255	192.168.137.101	192.168.137.101	TLSv1	242	Client Key Exchange, Change Cipher Spec, Encrypted Handshake Message
48	6.263319	192.168.137.101	192.168.137.101	TCP	44	5061 → 53850 [ACK] Seq=872 Ack=299 Win=525312 Len=0
49	6.267948	192.168.137.101	192.168.137.101	TLSv1	103	Change Cipher Spec, Encrypted Handshake Message
50	6.268013	192.168.137.101	192.168.137.101	TCP	44	53850 → 5061 [ACK] Seq=299 Ack=931 Win=524544 Len=0
51	6.268433	192.168.137.101	192.168.137.101	TLSv1	710	Application Data, Application Data
52	6.268472	192.168.137.101	192.168.137.101	TCP	44	5061 → 53850 [ACK] Seq=931 Ack=965 Win=524544 Len=0
53	6.317205	127.0.0.1	127.0.0.1	TCP	56	53851 → 5000 [SYN] Seq=0 Win=64240 Len=0 MSS=65495 WS=256 SACK_PERM=1
54	6.317292	127.0.0.1	127.0.0.1	TCP	56	5000 → 53851 [SYN, ACK] Seq=0 Ack=1 Win=65535 Len=0 MSS=65495 WS=256
55	6.317400	127.0.0.1	127.0.0.1	TCP	44	53851 → 5000 [ACK] Seq=1 Ack=1 Win=525568 Len=0
56	6.470538	192.168.137.101	192.168.137.101	TLSv1	630	Application Data, Application Data
57	6.470606	192.168.137.101	192.168.137.101	TCP	44	53850 → 5061 [ACK] Seq=965 Ack=1517 Win=525568 Len=0
58	6.681485	192.168.137.101	192.168.137.101	TLSv1	918	Application Data, Application Data
59	6.681547	192.168.137.101	192.168.137.101	TCP	44	5061 → 53850 [ACK] Seq=1517 Ack=1839 Win=525568 Len=0
60	6.704742	127.0.0.1	127.0.0.1	TCP	52	52844 → 5485 [PSH, ACK] Seq=1 Ack=1 Win=2053 Len=8
61	6.704806	127.0.0.1	127.0.0.1	TCP	44	5485 → 52844 [ACK] Seq=1 Ack=9 Win=2051 Len=0
62	6.704850	127.0.0.1	127.0.0.1	TCP	66	52844 → 5485 [PSH, ACK] Seq=9 Ack=1 Win=2053 Len=22
63	6.704882	127.0.0.1	127.0.0.1	TCP	44	5485 → 52844 [ACK] Seq=1 Ack=31 Win=2051 Len=0
64	6.704921	127.0.0.1	127.0.0.1	TCP	48	52844 → 5485 [PSH, ACK] Seq=31 Ack=1 Win=2053 Len=4
65	6.704953	127.0.0.1	127.0.0.1	TCP	44	5485 → 52844 [ACK] Seq=1 Ack=35 Win=2051 Len=0
66	6.704986	127.0.0.1	127.0.0.1	TCP	517	52844 → 5485 [PSH, ACK] Seq=35 Ack=1 Win=2053 Len=473
67	6.705018	127.0.0.1	127.0.0.1	TCP	44	5485 → 52844 [ACK] Seq=1 Ack=508 Win=2049 Len=0
68	6.705249	127.0.0.1	127.0.0.1	TCP	52	5485 → 52844 [PSH, ACK] Seq=1 Ack=508 Win=2049 Len=8
69	6.705302	127.0.0.1	127.0.0.1	TCP	44	52844 → 5485 [ACK] Seq=508 Ack=9 Win=2053 Len=0
70	6.705470	127.0.0.1	127.0.0.1	TCP	60	5485 → 52844 [PSH, ACK] Seq=9 Ack=508 Win=2049 Len=16
71	6.705534	127.0.0.1	127.0.0.1	TCP	44	52844 → 5485 [ACK] Seq=508 Ack=25 Win=2053 Len=0
72	6.705582	127.0.0.1	127.0.0.1	TCP	48	5485 → 52844 [PSH, ACK] Seq=25 Ack=508 Win=2049 Len=4

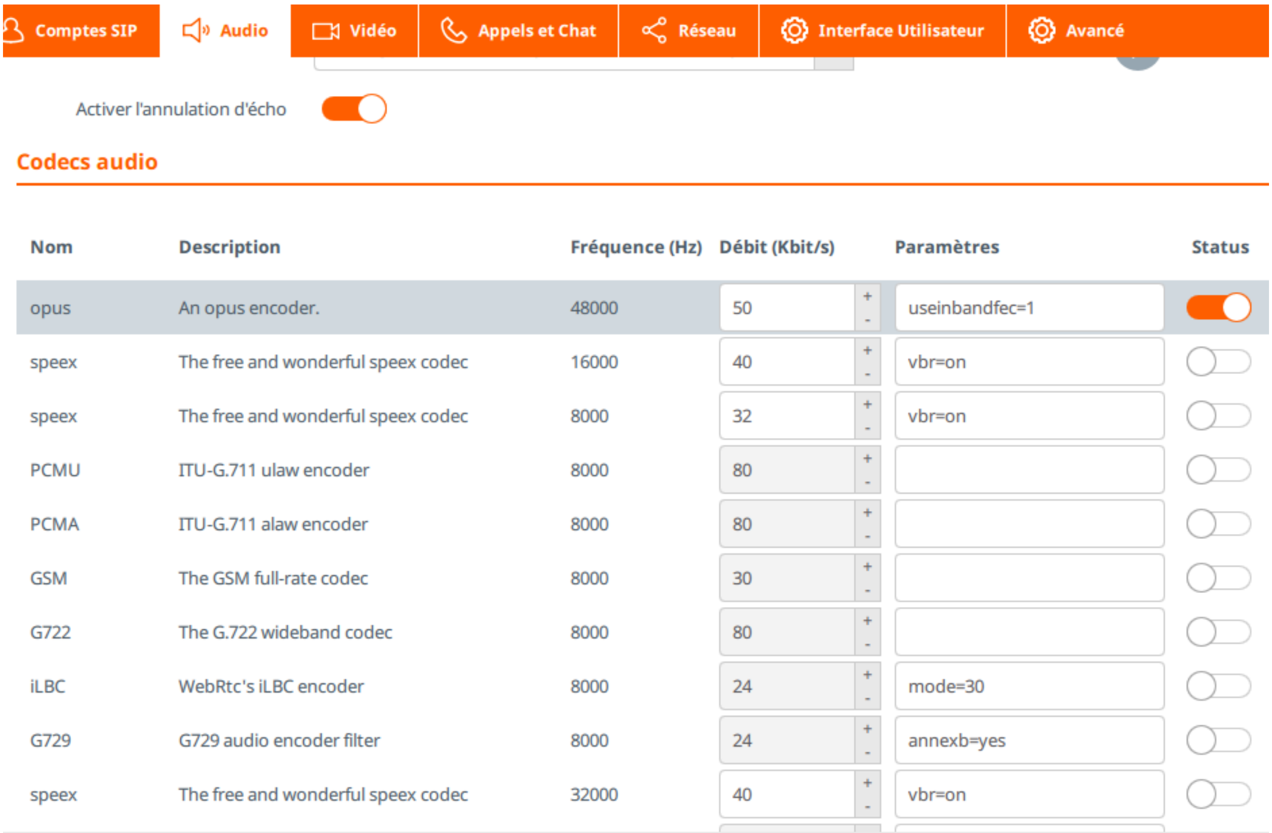
– Partie 7 : VOIP and Security with Linephone

- **Install the TOIP Linephone client in your terminals. Beware to unistall 3CX TOIP clients first. https://www.linphone.org**
- **Configure first the Linephone SIP accounts to connect to your 3CX server**



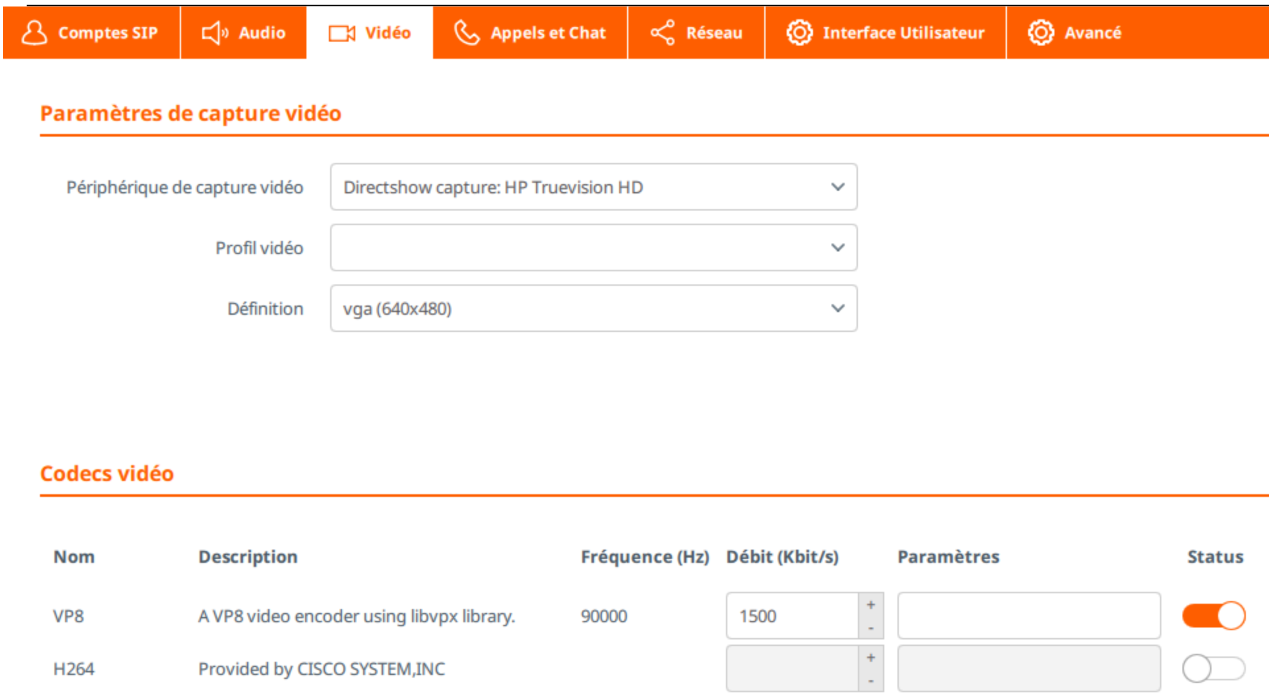
- **Configure then the Linephone audio codec. Explain what audio codec you selected. What is your audio bit rate ?**

  I selected the opus audio codec with a bitrate of 50 Kbit/s. Screenshot is available on the next page.

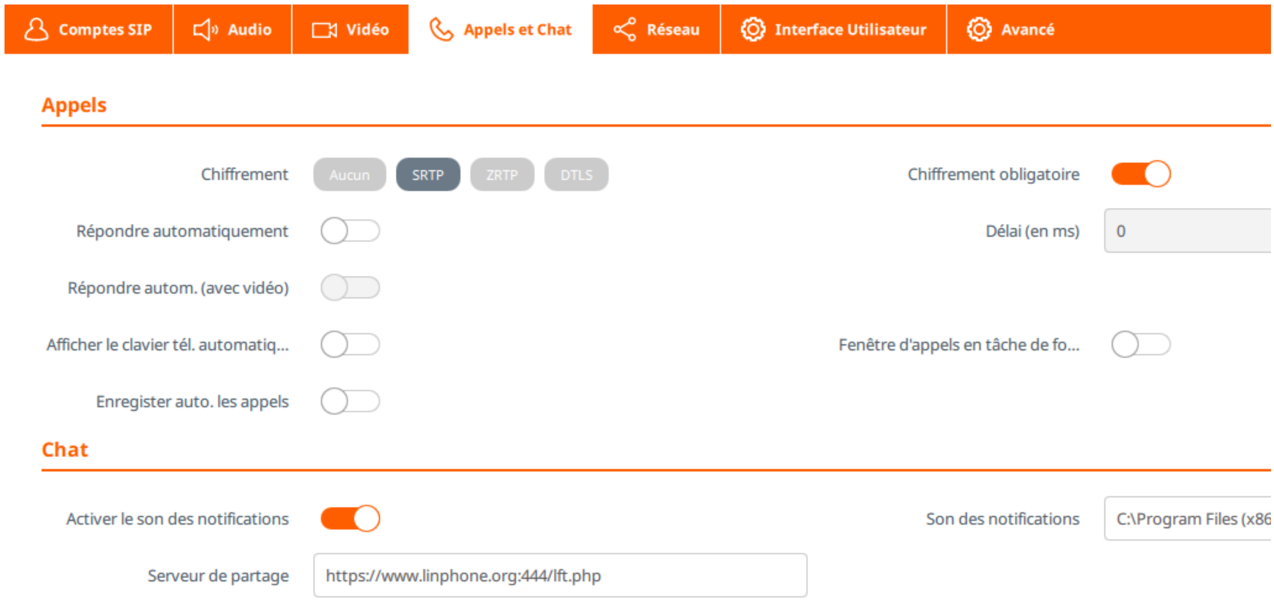


- **Configure then the Linephone <u>video</u> codec. Explain what video codec you selected (Cisco H.264 or google VP8). What is your video resolution (pixels) ? What is your video bitrate ?**

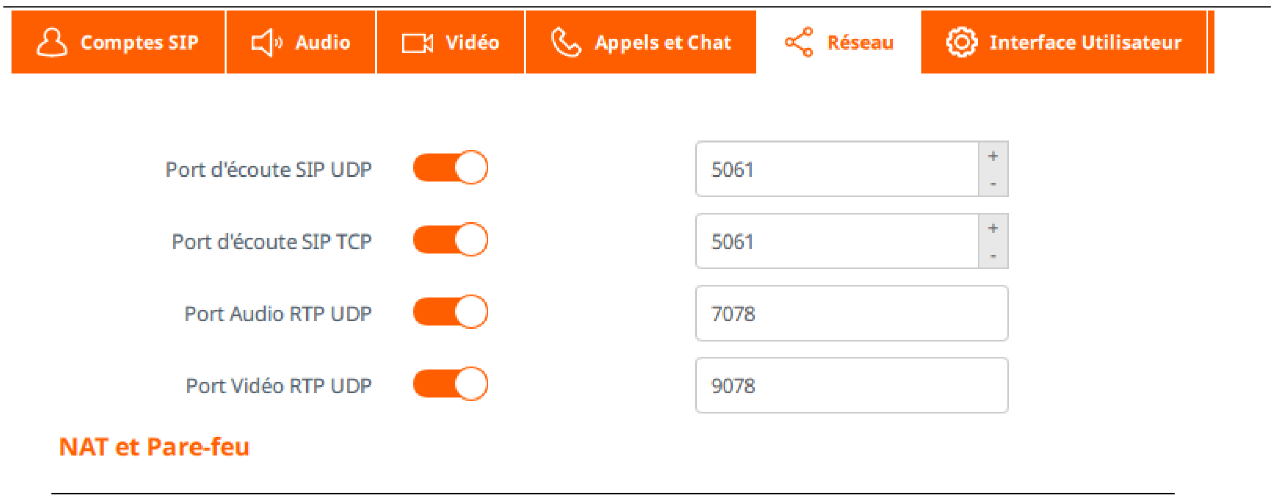
  I selected Google VP8 video codec, which can be used with a 640*480 pixels resolution. The bitrate is 1500 Kbit/s.



- **Configure then the Linephone <u>Security</u> options to select SRTP (data) and SIPS (signaling).**


Comptes SIP
Audio
Vidéo
Appels et Chat
Réseau
Interface Utilisateur
Avancé
Appels
Chiffrement
Aucun
SRTP
ZRTP
DTLS
Chiffrement obligatoire
Répondre automatiquement
Délai (en ms)
0
Répondre autom. (avec vidéo)
Afficher le clavier tél. automatiq...
Fenêtre d'appels en tâche de fo...
Enregister auto. les appels
Chat
Activer le son des notifications
Son des notifications
C:\Program Files (x86
Serveur de partage
https://www.linphone.org:444/lft.php


Paramètres principaux du compte SIP
Adresse SIP*
"BOB" <sip:100@192.168.137.101>
Adresse du serveur SIP*
<sip:192.168.137.101;transport=tls>
Durée d'enregistrement
3600
Transport
TLS
Route
Paramètres de contact
message-expires=604800
Intervalle standard RTCP AVPF (...
1
S'enregistrer
ANNULER
CONFIRMER


Comptes SIP
Audio
Vidéo
Appels et Chat
Réseau
Interface Utilisateur
Port d'écoute SIP UDP
5061
Port d'écoute SIP TCP
5061
Port Audio RTP UDP
7078
Port Vidéo RTP UDP
9078
NAT et Pare-feu

- **What is ZRTP ? what are the differences with SRTP ?**

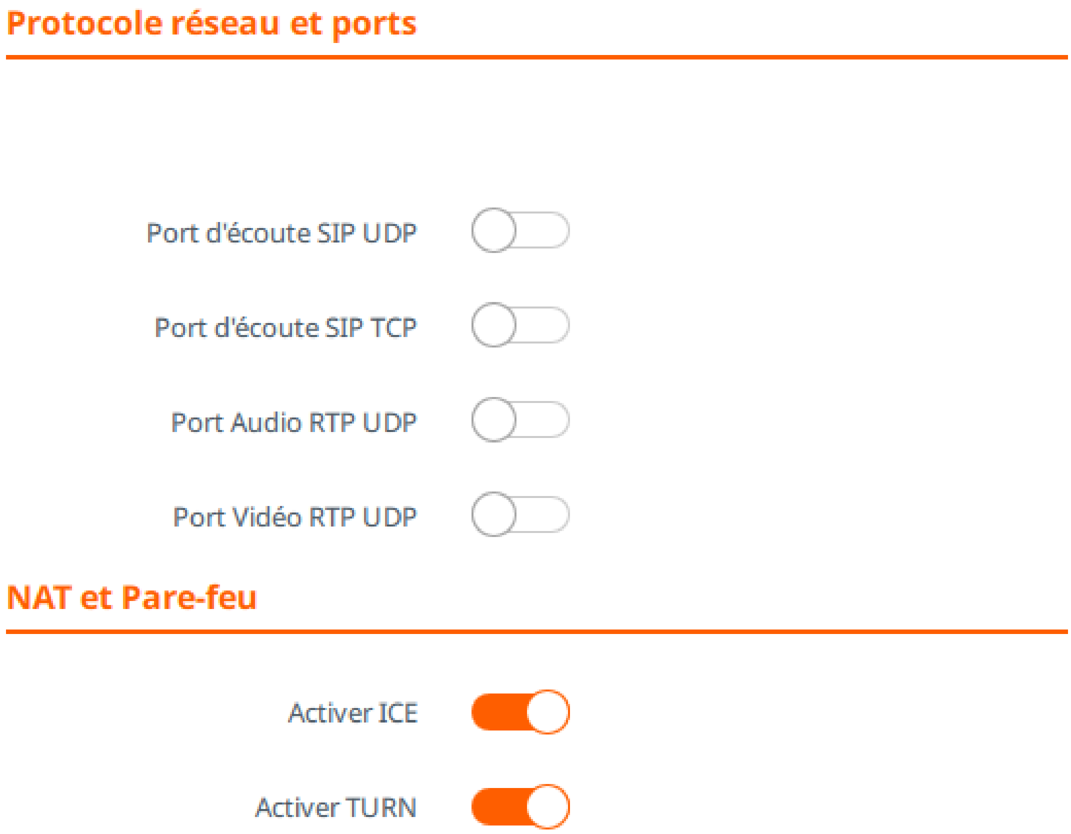
  ZRTP is a protocol which provides security between two end points in a VoIP network. To do so, Secure RTP (SRTP) is used, with the help of a Diffie-Hellman key exchange before any data communication.

- **What is Datagram TLS (DTLS). Find the RFC number for DTLS. What are the differences with SRTP ?**

  DTLS is defined in the RFC 4347 and in RFC 6347 for the version 1.2. It is a communication protocol, preventing from attacks like message forgery, tampering or eavesdropping. This is based on the TLS protocol and it’s similar to it. Actually, SRTP uses DTLS for the handshake and so the key exchange. Then, RTP data are encrypted using SRTP method during the communication between both end points.

- **Configure then the Linephone <u>network</u> options to setup your UDP/TCP port numbers for facilitating your firewall configurations**

  We didn’t understand what was expecting here, but we disabled fixed ports and we have activated ICE and TURN in the NAT and firewall options.



- **What is STUN/TURN protocols ? Why we need these protocols with TOIP ? Provide the name and IP address of two public STUN/TURN servers available online.**

  STUN protocol allows NAT clients in a local network to pass through firewall to communicate with an external VoIP operator. To do so, information like public IP, NAT router type or port used must be known and exchanged by the clients. So two VoIP end points can connect with the help of a STUN server. TURN is an extension of STUN, which aims to facilitate NAT traversal for media traffic. Media traffic is more heavy than network discovery packets, so it is relayed by a dedicated server: the STUN server.

  These protocols are important for TOIP because clients sometimes need to join another client in an external network. So handshake and then media traffic need to traverse firewall, with the help of STUN and TURN servers.

  Here are two public STUN/TURN servers available online:

- stun.3cx.com:3478 with IP 54.39.182.217

- stun.12voip.com:3478 with IP 77.72.169.212

We can register to them online.

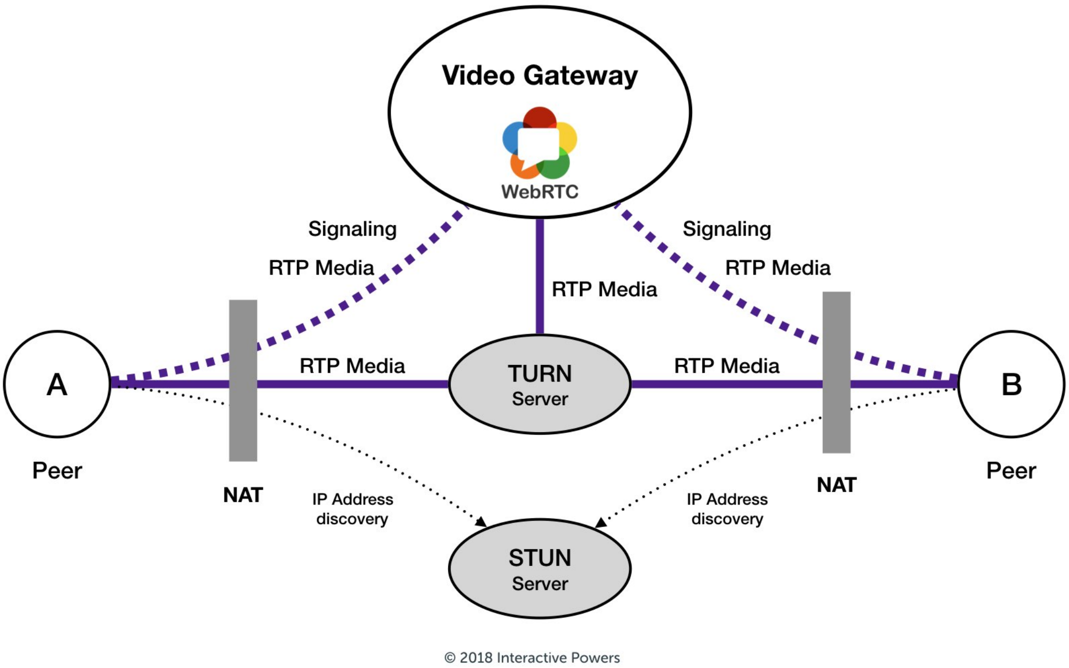


Diagram showing the roles of TURN and STUN servers from
https://blog.ivrpowers.com/post/technologies/what-is-stun-turn-server/