Cyber defense 2

**ENSIBS Vannes** 

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# Report

SIP : Communications interception and encryption

Network basis



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# 1 Project objectives

The objectives of this project are the following :

- Set up 2 asterisks servers
- Set up 2 soft-phones on our computers or android phones.
- Intercept the SIP communication between a soft-phone and the asterisk server.
- Intercept the RTP data communication between the 2 soft-phones.
- Encrypt the SIP communication using SSL/TLS.
- Encrypt the RTP data communication using ZRTP.
- Intercept the communication while it's encrypted and prove that the encryption is efficient and reliable.
- Doing a trunk SIP between the two asterisks servers.
- Try the same things when the two clients are connected on two different asterisks servers.

The difference with the overview is that there is a new objective which is doing a voice recognition on top of the interception of the RTP data communication.

# 2 Why did we not meet all of our objectives?

Mainly because of the lack of time we had and also because when we did the overview we didn't realize it will be a problem to do the encryption. We didn't even try to bypass the SSL/TLS encryption because it takes a lot of time but we think it is possible if the the version of this is SSL v3 for example so we can use the poodle vulnerability. It took us also much time to do the voice recognition because we didn't know the tools and we are not telecoms experts, although it was really cool to learn about it. We didn't meet the VPN objective also because of the lack of time but we think it can be easily done if we use openVPN for example.

# 3 What we wanted to learn and what we actually learned

We wanted to learn more about the VoIP, especially the SIP protocol and we did learn more about that because now we know for example that the SIP is a signaling protocol and RTP is for the data communication between the two soft-phones. After that we wanted to learn more about the encryption protocols and how it works. We used SSL/TLS for the SIP encryption so we learned a lot about this type of encryption and the certificates. We also learned how to make our own self-signed certificates thanks to this project. Now we know that to encrypt RTP, we use the SRTP protocol which uses AES 128 bits and SDES to exchange the keys and ZRTP is AES 128 bits too but is using Diffie-Hellman algorithm. We also learned what we wanted to learn which was how to configure an asterisk server, intercept a SIP communication and RTP communication and encrypt these communications. We

learned how to do two asterisks server and make them communicate (which is a trunk SIP). Finally we didn't plan that but we learned how to do a voice recognition using signals analysis. What we didn't learn because we didn't have the time for it is how to settle a VPN tunnel between the 2 asterisks servers.

#### 3.1 Challenges encountered

During our project we encountered some difficulties, the first one was how to intercept a SIP communication. Indeed, at the beginning we didn't know much about this protocol so we though SIP and RTP were the same thing .... So after we studied the protocol we were able to make the difference and then set up the right filters in Wireshark. Although that wasn't enough, we connected the two soft-phones, asterisk server and the attacker on the same network and the attacker couldn't intercept all the traffic which was a real problem. Then we understood that we needed to set up the network card into monitor mode and do some arp poisoning (we will explain that in details later on) and that solved the problem.

The next challenge was to encrypt the SIP communication because at the beginning we didn't really understand how SSL/TLS works. Also, after we understood that, we had to install the right soft-phones because some of them don't handle the SSL/TLS encryption. Once everything has been done, we had to configure the asterisk server and generate all the certificates, which was hard to do but with a lot of tutorials we managed to succeed. Finally, the big challenge was to do the voice recognition.

## 4 How did we share the code

We shared the code using GitHub and e-mails sometimes, for the proof here is the GitHub link: https://github.com/KevinF49/ProjetReseau

# 5 Communications Analysis: Wireshark to the rescue!

In this section we will mainly talk about the different communications that we did and explain them to you with the help of some Wireshark captures.

#### 5.1 ARP Communications

Address Resolution Protocol (reply)

As the title says, we will show you a capture of some ARP communications in here. We are doing this because this is our way to explain how the attack works and how we tested the security flaw.

1 0.000000000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.100? Tell 192.168.1.1			
2 0.002311000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.100 is at 64:5a:04:8d:45:c4			
3 0.002340000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.100 is at 64:5a:04:8d:45:c4			
4 0.003638000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.14? Tell 192.168.1.1			
5 0.006725000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.14 is at 64:5a:04:8d:45:c4			
6 0.006745000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.14 is at 64:5a:04:8d:45:c4			
7 0.008234000	Sagemcom_78:59:7c	Broadcast	ARP	42 Gratuitous ARP for 192.168.1.1 (Request)			
8 0.010100000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 Gratuitous ARP for 192.168.1.1 (Reply) (duplic			
9 0.010122000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 Gratuitous ARP for 192.168.1.1 (Reply) (duplic			
10 0.011486000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.101? Tell 192.168.1.1			
11 0.014539000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.101 is at 64:5a:04:8d:45:c4			
12 0.014561000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.101 is at 64:5a:04:8d:45:c4			
13 0.014882000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.76? Tell 192.168.1.1			
14 0.019033000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.17? Tell 192.168.1.1			
15 0.022326000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.17 is at 64:5a:04:8d:45:c4			
16 0.022356000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.17 is at 64:5a:04:8d:45:c4			
17 0.151017000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.15? Tell 192.168.1.1			
18 0.151034000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.15 is at 64:5a:04:8d:45:c4			
19 0.151051000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.100? Tell 192.168.1.1			
20 0.151059000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.16? Tell 192.168.1.1			
21 0.151066000	Sagemcom_78:59:7c	Broadcast	ARP	42 Who has 192.168.1.14? Tell 192.168.1.1			
22 0.154606000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.100 is at 64:5a:04:8d:45:c4			
23 0.154632000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.100 is at 64:5a:04:8d:45:c4			
24 0.154675000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.16 is at 64:5a:04:8d:45:c4			
25 0.154684000	ChiconyE_8d:45:c4	Sagemcom_78:59:7c	ARP	42 192.168.1.16 is at 64:5a:04:8d:45:c4			
26 0.154719000	ChiconvF 8d:45:c4	Sagemcom 78:59:7c	ARP	42 192.168.1.14 is at 64:5a:04:8d:45:c4			
▶ Frame 2: 42 bytes	on wire (336 bits). 42	2 bytes captured (336 b	oits) on :	interface 0			
▶ Frame 2: 42 bytes on wire (336 bits), 42 bytes captured (336 bits) on interface 0 ▶ Ethernet II, Src: ChiconyE 8d:45:c4 (64:5a:04:8d:45:c4), Dst: Sagemcom 78:59:7c (d0:84:b0:78:59:7c)							
- Editinice 11, 31c. diletinyE_da.43.c4 (04.34.04.43.c4), 53c. Sageilletin_76.33.7c (40.04.50.76.33.7c)							

Figure 1: ARP Flow

First of all, a scan host is established with Ettercap. Then the malicious person sends in broadcast some arp packets and matches his mac address with the ip address of the gateway. The router sends arp requests to fill his arp table. Thus attacker's mac address answers to all the requests which means his mac address is matched with every IP address on the network. Thereby, he fills his arp table and all request above the layer 2 are intercepted by him. The router notices there are two gateways and inform all the devices on the network by sending in broadcast some gratuitous arp packets. ChiconyE\_8d:45:c4 (64:5a:04:8d:45:c4) is the attackers' mac address and it is spoofing every IP address on the network. There is a solution to counter/avoid that "arp poisoning" attack which is setting the arp table in static mode.

#### 5.2 SIP Communications

In this section we will talk about the results of our project but not how we did it. In order to do that we will show you some Wireshark captures and explain them.

```
Protocol Length Info
        Time
                     Source
                                            Destination
  18622 143.31741506 192.168.1.14
                                            192.168.1.100
                                                                               409 Request: REGISTER sip:192.168.1.100
  18623 143.3215250( 192.168.1.14
                                            192.168.1.100
                                                                  STP
                                                                               409 Request: REGISTER sip:192.168.1.100
  18624 143.3260780( 192.168.1.100
                                            192.168.1.14
                                                                   SIP
                                                                               543 Status: 401 Unauthorized
                                                                                                                (0 bindings)
  18625 143.3300910(192.168.1.100
                                            192.168.1.14
                                                                   SIP
                                                                               543 Status: 401 Unauthorized
                                                                                                                (0 bindings)
  18626 143.3500080(192.168.1.14
                                                                                569 Request: REGISTER sip:192.168.1.10
  18627 143.3535120( 192.168.1.14
                                            192.168.1.100
                                                                   SIP
                                                                               569 Request: REGISTER sip:192.168.1.100
  18631 143.3599840( 192.168.1.100
                                            192.168.1.14
                                                                   SIP
                                                                               634 Request: OPTIONS sip:202@192.168.1.14;line=921f
  18632 143.3606660( 192.168.1.100
                                            192.168.1.14
                                                                  SIP
                                                                               577 Status: 200 OK
                                                                                                    (1 bindings) |
  18635 143 36132206 192 168 1 100
                                            197 168 1 14
                                                                   STP
                                                                               634 Request: OPTIONS sin:2020192 168 1 14:line=921
   ▶ CSeq: 4 REGISTER
   ▼ Contact: <sip:202@192.168.1.14;line=921ff75f4ca5b68>
     ▶ Contact URI: sip:202@192.168.1.14;line=921ff75f4ca5b68
   ▼ Authorization: Digest username="202", realm="asterisk", nonce="4569f643", uri="sip:192.168.1.100", response="774bc42c9eabb67b
      Authentication Scheme: Digest
      Username: "202"
      Realm: "asterisk"
      Nonce Value: "4569f643"
      Authentication URI: "sip:192.168.1.100"
      Digest Authentication Response: "774bc42c9eabb67b3d02eb8fea7a254f"
      Algorithm: MD5
     Max-Forwards: 70
     User-Agent: Linphone/3.5.2 (eXosip2/3.6.0)
     Expires: 3600
     Content-Length: 0
0000 64 5a 04 8d 45 c4 e4 d5
                               3d 73 5b 3f 08 00 45 68
                                                          dZ..E... =s[?..Eh
0010 02 2b 5d 83 40 00 40 11
                               57 14 c0 a8 01 0e c0 a8
                                                          .+].@.@. W.....
     01 64 13 c4 13 c4 02 17
                               2b 75 52 45 47 49 53 54
0020
                                                          .d..... +uREGIST
0030
     45 52 20 73 69 70 3a 31 39 32 2e 31 36 38 2e 31
                                                          ER sip:1 92.168.1
0040
     2e 31 30 30 20 53 49 50 2f 32 2e 30 0d 0a 56 69
                                                          .100 SIP /2.0..Vi
                                                                                                    Profile: Default
File: "/media/kevin/E1405115UBS/COU...
                                       Packets: 25570 · Displayed: 24 (0,1%) · Load time: 0:00.325
```

Figure 2: SIP REGISTER

On this screenshot we can see a weakness of the SIP protocol. User password is hashed with MD5 algorithm therefore it's easy to decrypt it.

With this picture only, we have all the information that we need to connect to the asterisk server:

- Username
- URI
- MD5 hash

### 5.3 TLS Encryption

Now that we saw that SIP is a vulnerable protocol, we will show you that it can be secured if we use SSL/TLS Encryption. For that we used the Blink soft-phone and we created certificates and private/public keys with the openSSL command.

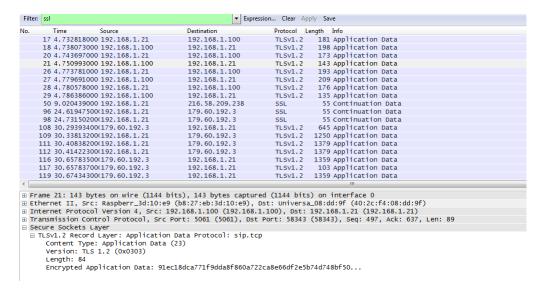


Figure 3: TLS REGISTER SUCCEED

In this capture you can see that all the packets are encrypted with the TLS Encryption. Just for the record, this is a capture when the user authenticates, which means you could see the right password with SIP but now you just can't see anything because it is encrypted.

Although, this is only the signalization in this conversation, which means we can still intercept the data packets between the two users, carried by the RTP protocol.

#### 5.4 RTP Communications

Like we said in the last subsection, the RTP protocol is used to transmit data packets in a VoIP communication. Concretely, it means that voice is carried by this protocol. Which is interesting because we wanted to see if we can intercept these packets and then extract the voice to replay the conversation, and we did.

627 4.074439000	192.168.1.17	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=0, Ti	
628 4.078532000	192.168.1.17	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=0, Ti	
629 4.080616000	192.168.1.100	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=0, Ti	
630 4.082854000	192.168.1.100	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=0, Ti	
631 4.085506000	192.168.1.17	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=1, Ti	
632 4.086109000	192.168.1.17	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=1, Ti	
634 4.096043000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=2, Ti	
639 4.098096000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=2, Ti	
648 4.107406000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=3, Ti	
650 4.108443000	192.168.1.100	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=1, Ti	
652 4.110561000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=3, Ti	
654 4.110670000	192.168.1.100	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=1, Ti	
665 4.144761000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=4, Ti	
666 4.146383000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=4, Ti	
667 4.164805000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=5, Ti	
668 4.166730000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=5, Ti	
669 4.177705000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,		
670 4.178534000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=6, Ti	
671 4.196593000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=7, Ti	
672 4.198813000	192.168.1.17	192.168.1.16	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x1A066B8,	Seq=7, Ti	
673 4.210562000	192.168.1.16	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x47A4CE48,	Seq=0, T	
674 4.210582000	192.168.1.16	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x47A4CE48,		
675 4.214847000	192.168.1.16	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x47A4CE48,	Seq=0, T	
676 4.214935000	192.168.1.16	192.168.1.100	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x47A4CE48,	Seq=1, T	
677 4.217616000	192.168.1.100	192.168.1.17	RTP	214 PT=ITU-T G.711 PCMU, SSRC=0x47A4CE48,		
678 4.217628000	192.168.1.100	192.168.1.17	RTP	214 PT=TTU-T G.711 PCMU. SSRC=0x47A4CF48.	Sea=1. T	
Frame 629: 214 bytes on wire (1712 bits), 214 bytes captured (1712 bits) on interface 0						
> Ethernet II, Src: Raspberr 3d:10:e9 (b8:27:eb:3d:10:e9), Dst: ChiconyE 8d:45:c4 (64:5a:04:8d:45:c4)						
Internet Protocol Version 4, Src: 192.168.1.100 (192.168.1.100), Dst: 192.168.1.16 (192.168.1.16)						
▶ User Datagram Protocol, Src Port: 13678 (13678), Dst Port: 7078 (7078)						
▶ Real-Time Transport	: Protocol					

Figure 4: RTP

This capture shows the conversation between two soft-phones with the RTP protocol. We can see that the host 192.168.1.17 wants to speak with 192.168.1.16. But when the host believes that he is sending the packets at the asterisk server, he actually sends the packets to the pirate. That's the same when the real host answers, the packets are for the attacker. To sum up, communications are intercepted by the malicious person. This is a hyperlink to a conversation that we intercepted: click to play the sound!

#### 5.5 SRTP Communications

We saw before that RTP is a weak protocol because it is not encrypted, so now we will encrypt it. For that we used SRTP or ZRTP protocols, both are extensions to RTP that incorporates enhanced security features. SRTP for example encrypts RTP using AES 128 bits.

14233 274.016610000	192.168.1.100	192.168.1.71	SRTP	224 PT=ITU-T G.711 PCMU, SSRC=0x5F58D097, Seq=2
14234 274.016749000	192.168.1.100	192.168.1.71	SRTP	224 PT=ITU-T G.711 PCMU, SSRC=0x5F58D097, Seq=2
14235 274.018675000	192.168.1.100	192.168.1.71	SRTP	224 PT=ITU-T G.711 PCMU, SSRC=0x5F58D097, Seq=2
14236 274.018748000	192.168.1.100	192.168.1.71	SRTP	224 PT=ITU-T G.711 PCMU, SSRC=0x5F58D097, Seq=2
14237 274.027551000	192.168.1.100	192.168.1.71	SRTP	224 PT=ITU-T G.711 PCMU, SSRC=0x5F58D097, Seq=2
14238 274.030624000	192.168.1.100	192.168.1.71	SRTP	224 PT=ITU-T G.711 PCMU, SSRC=0x5F58D097, Seq=2
14239 274.050116000	192.168.1.100	192.168.1.71	SRTP	224 PT=ITU-T G.711 PCMU, SSRC=0x5F58D097, Seq=2

Figure 5: SRTP Flow

In this capture the principle is the same that the one above but in this one the conversation is encrypted with SRTP protocol. We won't detail how it works here because this isn't the purpose of this report but basically, it counters our attack because even if we intercept all the SRTP packets, and then we replay the conversation, we will only hear noise because of the encryption, so it's impossible here to hear the voices of the people in a conversation as you can see in this audio file

### Conclusion

As you saw when you read this report, it is not about how we configured the servers and soft-phones to get them communicate using VoIP but more about how they do it. For that we had to analyze some Wireshark captures and that is what we explained in here. To end this report, we would say that SIP along with RTP are really vulnerable therefore if someone wants to set up a ToIP infrastructure they have to secure these protocols using at least SSL/TLS and ZRTP/STRP. Indeed, not only they are vulnerable but the attack is quite easy to do considering it's a simple man in the middle. We didn't do the best secure network because on top of what we did a VPN can be settled between the two asterisk servers and after that we think it is pretty secure.