

# Challenge 4: VoIP (intermediate)

## Submission Template

Submit your solution at <http://www.honeynet.org/challenge2010/> by 17:00 EST, Wednesday, June 30th 2010. Results will be released on Wednesday, July 21st 2010.

Name (required): Franck Guénichot	Email (required): franck.guenichot@orange.fr
Country (optional): France	Profession (optional): _ Student- _ Security Professional- _ Other

**Nota:** In this document, I've use some picviz graphs and custom scripts. These elements can be found in a password-protected zip archive at <http://malphx.free.fr/dotclear/public/HPFC4-stuff.zip>. The secret password is: \$v0ipch4lL3nge!

Section 1/ Question 1. What protocol is being used? Is it TCP or UDP?	Possible Points: 1pt
Tools Used: awk, sort, uniq, grep, SIPlogparser.rb (custom tool)	
Awarded Points:	
Answer	
<p>Session Initiation Protocol (SIP) message as specified in RFC 3261, must have a "Via" header field. This field must indicate the protocol and protocol version: SIP 2.0, but it specifies also the transport protocol selected (TCP or UDP).</p> <p>So, to know which transport protocol was used, we just have to look at all the "Via" parameter of all the logged SIP messages.</p> <p>We can quickly scan the log file with this one-liner:</p> <pre>grep "Via:" logs_v3.txt  awk '{match(\$0,"Via: SIP\/2.0\/(TCP UDP)");print substr(\$0,1,RLENGTH);}'  sort  uniq</pre> <pre>franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4\$ grep "Via:" logs_v3.txt  awk '{match(\$0,"Via: SIP\/2.0\/(TCP UDP)");print substr(\$0,1,RLENGTH);}'  sort  uniq</pre> <pre>awk: AVERTISSEMENT: séquence d'échappement « \/ » traitée simplement comme « / »</pre> <pre>Via: SIP/2.0/UDP</pre>	
<p>We know that the Via header field is mandatory and is constructed like below: Via: SIP/2.0/(transport) where transport can be TCP or UDP. So, all the messages stored in this log file have used UDP as transport protocol.</p> <p>I've written a small ruby script to parse the given log file. It is far from being perfect but it makes the job ! Its name: SIPlogparser.rb</p>	

With “-z” option switch, SIPlogparser.rb is able to display general statistics about the SIP messages that have been logged in the given file:

```

franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ ruby SIPlogparser.rb -r
logs_v3.txt -z

...::: General Statistics :::...

0/4266 messages filtered
.....
4266 UDP messages / 0 TCP messages
.....
4 SIP INVITE messages
7 SIP SUBSCRIBE messages
4254 SIP REGISTER messages
1 SIP OPTIONS messages
.....

```

The result above also indicates that all the messages within this log files have traveled using UDP.

Section 1/ Question 2. Could this log be the result a simple nmap scan being run against the honeynet? Explain

Possible Points:  
1pt

Tools Used:

Answer

If “simple nmap scan” refers to a basic usage of this powerful tool (like a “simple” UDP scan for example), well, no, this log could not be the result of an nmap recon.

Because, an UDP scan could not have sent SIP methods (OPTIONS, REGISTER...) like those in the log.

Here the explanation from the nmap book by Fyodor:

“ UDP scan works by sending an empty (no data) UDP header to every targeted port”

But, I think it could be possible to have similar results using the Nmap Scripting Engine (NSE) and a well-written script.

Section 1/ Question 3a. Name the scanning tools that may have been used to by the attacker.	Possible Points: 1pt
---	-------------------------

Tools Used: vi

Answer

There is at least two informations which can help us guessing the tool that was used by the attacker. The first one is the name "sipvicious" and the second one is the user-agent string: "friendly-scanner"

```
Source: 210.184.X.Y:1083
Datetime: 2010-05-02 01:43:05.606584

Message:

OPTIONS sip:100@honey.pot.IP.removed SIP/2.0
Via: SIP/2.0/UDP 127.0.0.1:5061;branch=z9hG4bK-2159139916;rport
Content-Length: 0
From: "sipvicious"<sip:100@1.1.1.1>; tag=X_removed
Accept: application/sdp
User-Agent: friendly-scanner
To: "sipvicious"<sip:100@1.1.1.1>
Contact: sip:100@127.0.0.1:5061
CSeq: 1 OPTIONS
Call-ID: 845752980453913316694142
Max-Forwards: 70
```

Googling for these two words sends you to this site: <http://blog.sipvicious.org/>  
Here we learn that SIPvicious is a set of tools written by Sandro Gauci

From the README file of the SIPvicious package:

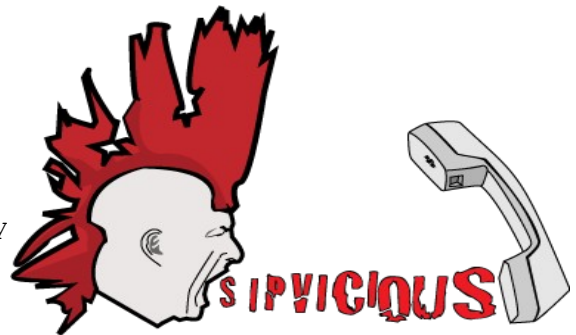
The tools:

**svmap** - this is a sip scanner. When launched against ranges of ip address space, it will identify any SIP servers which it finds on the way. Also has the option to scan hosts on ranges of ports.

**svwar** - identifies working extension lines on a PBX. A working extension is one that can be registered. Also tells you if the extension line requires authentication or not.

**svcrack** - a password cracker making use of digest authentication. It is able to crack passwords on both registrar servers and proxy servers. Current cracking modes are either numeric ranges or words from dictionary files.

**svreport** - able to manage sessions created by the rest of the tools and export to pdf, xml, csv and plain text.



**svlearnfp** - allows you to generate new fingerprints by simply running the tool against a host. It will attempt to guess most values and allow you to save the information to the local fingerprint db. Then you can choose to upload it to the author so that it can be added to the database.

For usage help make use of -h or --help switch.

Also check out the wiki:

<http://code.google.com/p/sipvicious/w/list>

And if you're stuck you're welcome to contact the author.

Sandro Gauci

sandrogauc at gmail dot com

(I like the reference to the well-known bass player of the punk band Sex Pistols named Sid Vicious)

Section 1/ Question 3b. What was the tool suite author's intended use of this tool suite ? Who was it designed to be used by?

Possible Points:  
1pt

Tools Used: SIPvicious Project site

Answer

Well, originally, SIPvicious Tools suite had been developed to help VoIP Administrators and Security professionals to assess SIP systems security.

Quote from the author (FAQ):

<http://code.google.com/p/sipvicious/wiki/FrequentlyAskedQuestions>

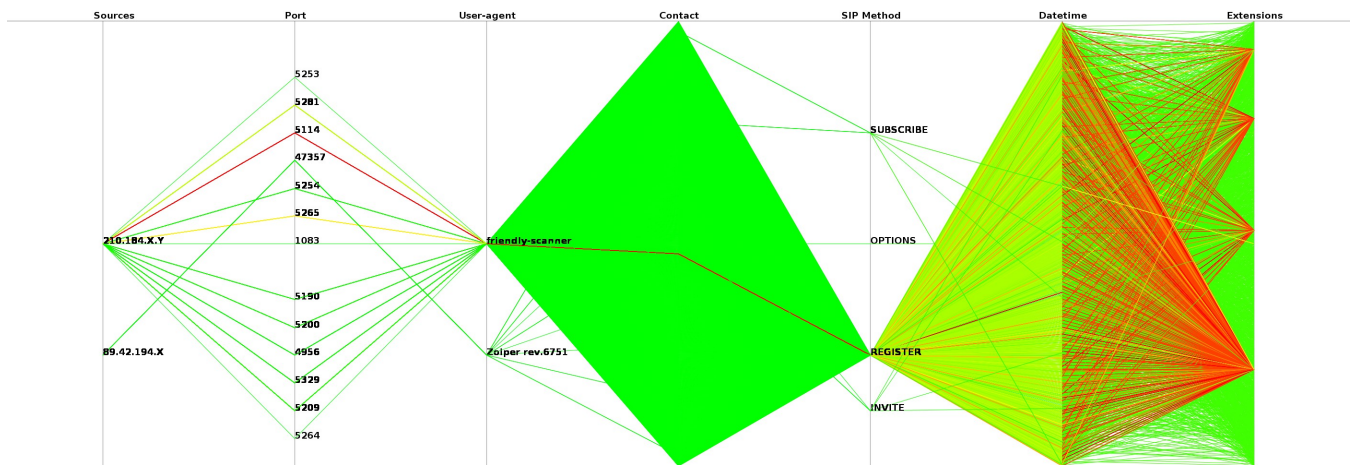
*The idea behind the tools is to aid administrators and security folks make informed decisions when evaluating the security of their SIP-based servers and devices. The tools are intended to be used for educational and demonstrational purposes. We advise people to request permission before making use of the tool suite against any network. Just like a knife, it can be used for good and bad. We hope that SIPVicious tool suite proves to be a very sharp one.*

Section 1/ Question 3c. One of these tools was only used against a small subset of extensions. Which were these extensions and why were only they targeted with this tool ?	Possible Points: 2pts
Tools Used: svmap.py,svwar.py,svcrack.py source files, SIPlogparser.rb, picviz	
<p>Answer</p> <p>Before answering this question, it is important to review the usage of the different SIPvicious tools, their default behavior and signatures (if any):</p> <ul style="list-style-type: none"> <li>• <b>svmap.py</b> =&gt; scans networks to find SIP PBX. <ul style="list-style-type: none"> <li>◦ Default behavior: sends an OPTIONS message to the 100 extension (exten) on the targeted IP.</li> <li>◦ Signature: <ul style="list-style-type: none"> <li>▪ From and To header field set to value: "sipvicious" &lt;sip:<a href="mailto:100@1.1.1.1">100@1.1.1.1</a>&gt;</li> <li>▪ User-agent header field set to value: friendly-scanner</li> </ul> </li> </ul> </li> <li>• <b>svwar.py</b> =&gt; scans a given SIP PBX to find valid extensions. Displays also, extension that need authentication. <ul style="list-style-type: none"> <li>◦ Default behavior: Sends REGISTER messages to a given range of exten or name store in a dictionary file.</li> <li>◦ Evaluates the PBX response code to define if the exten exists and needs authentication. The rules below are used to verify extensions validity: <ul style="list-style-type: none"> <li>▪ "200 OK" response code indicates a valid exten that doesn't need authentication</li> <li>▪ "404 Not found" response code means the exten is invalid.</li> <li>▪ "401 Unauthorized" response code means the exten is valid but needs authentication.</li> </ul> </li> <li>◦ Well, these are basic rules to understand how it works, other SIP response codes are take into account by svwar.py to evaluates extension validity.</li> <li>◦ Signature <ul style="list-style-type: none"> <li>▪ User-Agent set to: friendly-scanner</li> </ul> </li> </ul> </li> <li>• <b>svcrack.py</b> = Tries to find secret password of a given extension (doing brute-force and/or dictionary attack) <ul style="list-style-type: none"> <li>◦ Default behavior: Sends REGISTER message to sip:TARGET_IP_ADDR</li> <li>◦ uses brute-force or a dictionary to generate MD5 hashes in authorization header field.</li> <li>◦ SIP message sent by svcrack.py don't always use an authorization message header field, the reason explain by the author himself is: to force the refresh of the nonce sent by the SIP PBX.</li> <li>◦ Signature: <ul style="list-style-type: none"> <li>▪ Request-line set to REGISTER sip:TARGET_IP_ADDR SIP/2.0</li> <li>▪ Contact message header field set to: sip:<a href="mailto:123@1.1.1.1">123@1.1.1.1</a></li> <li>▪ User-Agent message header field set to: friendly-scanner</li> </ul> </li> </ul> </li> </ul> <p>With all those informations in hands, it is now possible to trace the usage of a specific tool of the SIPvicious suite against extensions on the victim's PBX. But, it is important to remember that those behaviors and signatures are only default values that will only appear if the tools are used without any specific options. For example, svmap.py and svwar.py allow the attacker to change the default SIP method used to scan.</p>	

Now it's time to answer this question, a quick analysis of the log file reveals a 4 phases attack.

1. **Phase 1:** An svmap.py scan for SIP PBX land on the fake SIP server (SIP OPTIONS To [100@1.1.1.1](http://100@1.1.1.1))
2. **Phase 2:** A scan is launched with swar.py against a large amount of extensions.
3. **Phase 3:** svcrack.py is used against extensions that were found valid, but that need authentication.
4. **Phase 4:** unprotected extensions and password-cracked extensions are used by an attacker to call international phone numbers.

I've used picviz [<http://www.wallinfire.net/picviz/index.html>] to visualize the dataset. Picviz can ease visualization of what has happened. Below is the big picture:



Well this large picture doesn't fit well in this document,so you can find it here:[http://malphx.free.fr/dotclear/public/HPFC4-stuff.zip/Big\\_picture.png](http://malphx.free.fr/dotclear/public/HPFC4-stuff.zip/Big_picture.png)

I've chosen to display the dataset based on 7 axis.

- Axe 1 => Sources (IP addresses)
- Axe 2 => Port (Source port)
- Axe 3 => SIP User-agent
- Axe 4 => SIP Contact header fiel
- Axe 5 => SIP method of the message
- Axe 6 => Datetime
- Axe 7 => Extensions (Targeted by the message)

SIPlogparser.rb has an option switch (-p) to generate a PGDL file name "graph.pcv".

This file can further be rendered with the Picviz CLI tool: pcv.

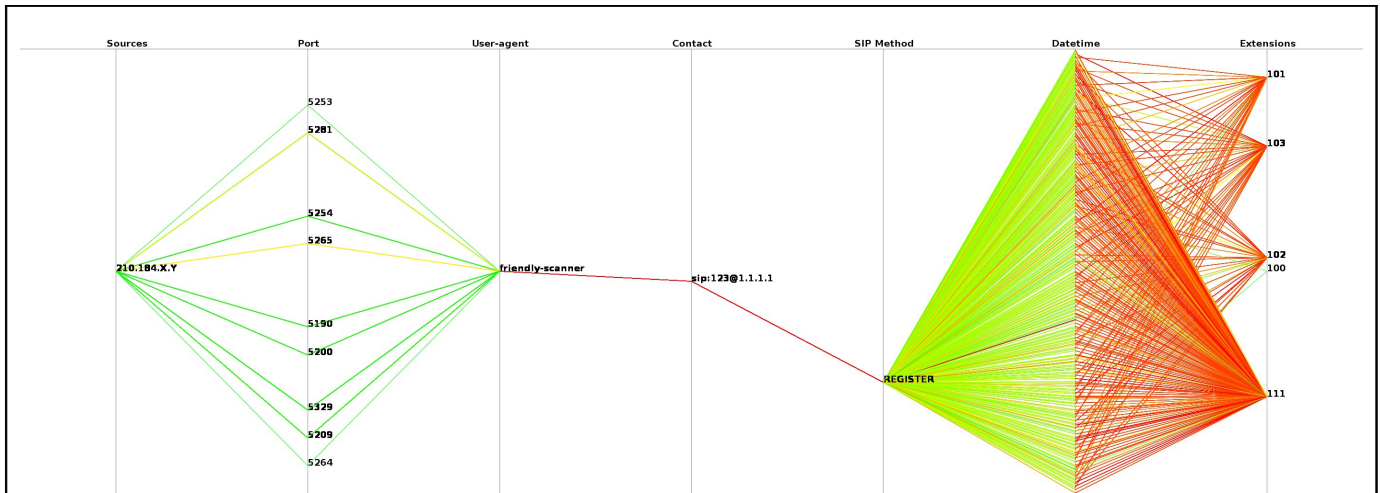
To generate the picture above, I've used the following command lines:

- `ruby SIPlogparser.rb -r logs_v3.txt -p`
- `pcv -Tpngcairo graph.pcv -rrra -o big_picture.png -Rheatmap`

We can easily notice the "red lines" against a small subset of extension, in fact 4 extensions only. And a red line beginning from source 210.184.X.Y and passing through the REGISTER SIP method. Using the tool's "signatures" explained previously, let's try to filter this graph to reveal what tool was used against these 4 extensions. After some tries, here the result:

You can generate the picture below from the "graph.pcv" file and this command line:

```
pcv -Tpngcairo graph.pcv -rrra -o svcrack.png -Rheatmap 'value="sip:123@1.1.1.1" on axis 4 and value="REGISTER" on axis 5'
```



You can find this picture here: <http://malphx.free.fr/dotclear/public/HPFC4-stuff.zip/svcrack.png>  
 This picture depicts the use of svcrack.py on a small subset of extensions.

Those extensions were:

- 101
- 102
- 103
- 111

All from the domain: honey.pot.IP.removed

Even if the SIP server responses are not logged in the given log file, we can guess what has happened and why those extensions were targeted with svcrack.py.

From phase 2 of the attack scenario, our attacker has a list of valid extensions (protected and unprotected) on this honeypot. So, with these informations, he then launched a brute-force attack against extensions that were found protected by a password. (ie: Extensions for which svmap.py has received a 401 Unauthorized Status Code from the server when it has tried to register them).

Section 1/ Question 4a. How many extensions were scanned? Are they all numbered extensions, or named as well?. List them

Possible Points:  
2pts

Tools Used: SIPlogparser.rb

Answer

From the previously explained analysis and tools use. We can consider that the scan was made with swar.py. Looking at the statistics given by SIPlogparser.rb, we conclude that the scan was done with the default proposed method: SIP REGISTER.

```
franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ ruby SIPlogparser.rb -r
logs_v3.txt -z
Parsing logs_v3.txt
```

```
.....: General Statistics :::....
```

```
0/4266 messages filtered
```

```
.....
4266 UDP messages / 0 TCP messages
.....
```

```
.....: Sources Statistics :::....
```

```
210.184.X.Y:5114 : 2607 messages sent
```

```
210.184.X.Y:5281 : 965 messages sent
```

```
89.42.194.X:47357 : 18 messages sent
```

```
210.184.X.Y:5329 : 94 messages sent
```

```
210.184.X.Y:5264 : 1 messages sent
```

```
210.184.X.Y:5253 : 1 messages sent
```

```
210.184.X.Y:5209 : 170 messages sent
```

```
210.184.X.Y:4956 : 45 messages sent
```

```
210.184.X.Y:5265 : 78 messages sent
```

```
210.184.X.Y:5254 : 98 messages sent
```

```
210.184.X.Y:1083 : 1 messages sent
```

```
210.184.X.Y:5200 : 94 messages sent
```

```
210.184.X.Y:5190 : 94 messages sent
```

```
.....: SIP Methods Statistics :::....
```

```
4 SIP INVITE messages
```

```
7 SIP SUBSCRIBE messages
```

```
4254 SIP REGISTER messages
```

```
1 SIP OPTIONS messages
.....
```

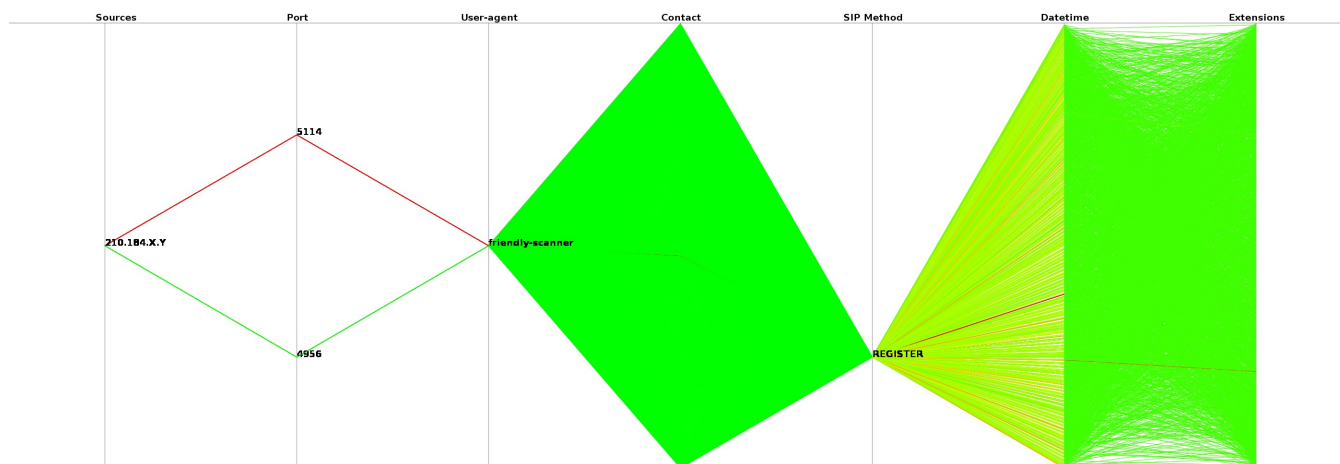
As stated previously, REGISTER message statistics don't let any doubt about the method that was used to scan. This value include extensions scanned by swar.py, but also the brute-force attempt made with svcrack.py.



With picviz, we can view this “phase 2 scan” easily:

command line:

```
pcv -Tpngcairo graph.pcv -rrra -o phase2_svar_scan.png -Rheatmap
'value="friendly-scanner" on axis 3 and value="REGISTER" on axis 5 and value !=
"sip:123@1.1.1.1" on axis 4'
```



[[http://malphx.free.fr/dotclear/public/HPFC4-stuff.zip/phase2\\_svar\\_scan.png](http://malphx.free.fr/dotclear/public/HPFC4-stuff.zip/phase2_svar_scan.png)]

This graph depicts two interesting points:

- Only two UDP source ports were used to scan all the extensions
- The yellow lines between Axis 5 (SIP method) and Axis 6 (Datetime) reveals that multiple extensions were scanned by second (The datetime axis is rounded to second)

Further analysis will reveal that:

- UDP port 5114 was used to scan numbered extensions
- UDP port 4956 was used to scan unnumbered (name) extensions

This point answer a part of the question: Scanned extensions were numbered extensions and names.

By combining -o and -m REGISTER option switches of SIPIlogparser.rb we can obtain the list of all the extensions that were scanned.

Listing extensions:

cmd: **SIPIlogparser.rb -r logs\_v3.txt -m register -o > scanned\_extensions**

This reveal that **2652 extensions** were scanned.

This number can be decomposed in:

- **2 “test” extensions**
  - "1729240413"<sip:[1729240413@honey.pot.IP.removed](#)>
    - Scanned before launching the scan against numbered extensions
  - "3428948518"<sip:[3428948518@honey.pot.IP.removed](#)>
    - Scanned before launching the scan against name extensions

These extensions doesn't appear to belong to the scan range. In fact, svar.py automatically generate these “test extens” and scan them before launching the real scans.

- **44 named extensions**

```
franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ ruby SIPlogparser.rb -r
logs_v3.txt -m register -N -o |more
Parsing logs_v3.txt
admin@honey.pot.IP.removed
info@honey.pot.IP.removed
test@honey.pot.IP.removed
postmaster@honey.pot.IP.removed
sales@honey.pot.IP.removed
service@honey.pot.IP.removed
support@honey.pot.IP.removed
marketing@honey.pot.IP.removed
manager@honey.pot.IP.removed
market@honey.pot.IP.removed
spam@honey.pot.IP.removed
user@honey.pot.IP.removed
data@honey.pot.IP.removed
cpanel@honey.pot.IP.removed
trixbox@honey.pot.IP.removed
news@honey.pot.IP.removed
fax@honey.pot.IP.removed
postfix@honey.pot.IP.removed
owner@honey.pot.IP.removed
client@honey.pot.IP.removed
operator@honey.pot.IP.removed
asterisk@honey.pot.IP.removed
oracle@honey.pot.IP.removed
temp@honey.pot.IP.removed
jobs@honey.pot.IP.removed
shop@honey.pot.IP.removed
help@honey.pot.IP.removed
orders@honey.pot.IP.removed
aaron@honey.pot.IP.removed
steve@honey.pot.IP.removed
andrew@honey.pot.IP.removed
jane@honey.pot.IP.removed
mike@honey.pot.IP.removed
joshua@honey.pot.IP.removed
christopher@honey.pot.IP.removed
abigail@honey.pot.IP.removed
richard@honey.pot.IP.removed
steven@honey.pot.IP.removed
sarah@honey.pot.IP.removed
heaven@honey.pot.IP.removed
freddy@honey.pot.IP.removed
samantha@honey.pot.IP.removed
sebastian@honey.pot.IP.removed
norman@honey.pot.IP.removed

44 extensions were scanned
```

- 2606 numbered extensions (all from domain honey.pot.IP.removed)

```

100 101 102 103 104 106 107 108
109 110 111 112 113 114 115 116 117 118
119 120 121 122 123 124 125 126 127 128
129 130 131 132 133 134 135 137 152 159
175 178 189 191 194 195 196 197 199 202
205 210 213 216 220 225 226 227 230 233
237 240 244 247 250 255 260 262 266 267
269 272 275 279 282 287 290 294 298 299
301 305 308 314 318 324 331 338 343 349
355 360 366 373 380 385 392 398 403 408
413 418 425 431 438 444 450 454 460 465
472 480 485 490 494 500 504 509 513 518
523 527 534 539 543 549 558 563 566 570
574 577 581 586 589 593 596 599 602 606
610 612 616 619 622 625 628 631 634 640
643 646 649 652 654 656 659 662 665 667
669 672 678 683 689 698 701 704 707 709
712 715 717 721 724 727 729 732 735 738
741 744 747 750 753 756 757 759 762 766
769 772 775 778 782 785 790 795 799 803
808 818 821 827 831 836 839 842 845 846
849 852 856 859 864 868 871 875 881 886
887 891 895 896 898 900 905 912 917 920
923 927 931 936 940 943 946 949 951 954
956 960 963 968 972 976 979 982 985 989
993 996 997 1000 1003 1006 1009 1013 1016 1019
1022 1023 1026 1027 1029 1032 1034 1037 1041 1045
1048 1051 1054 1059 1063 1067 1072 1077 1081 1084
1087 1091 1094 1097 1100 1105 1109 1110 1114 1118
1119 1123 1129 1135 1139 1144 1148 1154 1159 1162
1167 1172 1179 1183 1187 1192 1195 1201 1205 1209
1214 1218 1227 1230 1233 1242 1245 1250 1253 1256
1260 1264 1267 1270 1274 1277 1280 1283 1290 1293
1298 1300 1303 1305 1310 1313 1317 1321 1325 1328
1331 1335 1338 1341 1345 1348 1350 1357 1359 1362
1365 1366 1369 1372 1375 1378 1381 1384 1387 1390
1393 1396 1399 1401 1405 1409 1411 1413 1415 1418
1423 1430 1443 1448 1449 1450 1454 1458 1462 1466
1470 1474 1477 1480 1483 1487 1489 1493 1494 1496
1499 1502 1505 1508 1511 1515 1518 1523 1526 1529
1533 1537 1540 1544 1547 1551 1555 1558 1559 1561
1565 1569 1574 1577 1580 1583 1586 1592 1596 1599
1602 1605 1609 1614 1619 1623 1627 1632 1636 1641
1645 1651 1652 1656 1657 1660 1664 1668 1673 1677
1682 1687 1692 1696 1699 1703 1707 1709 1715 1719
1723 1727 1730 1733 1737 1740 1745 1751 1753 1758
1759 1763 1766 1770 1778 1784 1786 1790 1793 1796
1799 1804 1806 1810 1813 1816 1820 1824 1827 1831
1832 1834 1838 1844 1849 1852 1855 1860 1863 1866
1869 1872 1875 1878 1883 1888 1892 1896 1900 1905
1909 1913 1916 1921 1924 1928 1932 1936 1940 1943

```

1948	1950	1953	1956	1960	1963	1970	1976	1979	1987
1991	1993	1997	2000	2004	2007	2010	2014	2017	2021
2024	2028	2032	2036	2040	2042	2045	2049	2053	2056
2060	2064	2067	2068	2071	2075	2079	2083	2087	2091
2095	2098	2101	2105	2108	2109	2112	2116	2121	2125
2129	2133	2140	2149	2153	2160	2164	2167	2171	2178
2182	2186	2190	2192	2195	2202	2203	2205	2210	2213
2223	2228	2232	2235	2238	2240	2243	2246	2254	2257
2261	2265	2269	2273	2277	2281	2284	2288	2292	2296
2299	2303	2307	2311	2314	2318	2328	2331	2335	2339
2350	2355	2359	2363	2367	2370	2371	2373	2374	2376
2380	2381	2384	2389	2393	2396	2397	2409	2413	2414
2417	2421	2422	2423	2427	2430	2433	2437	2438	2440
2444	2448	2450	2453	2457	2460	2463	2464	2465	2468
2471	2474	2478	2482	2486	2489	2493	2496	2499	2501
2505	2509	2513	2516	2520	2521	2522	2523	2526	2529
2533	2536	2539	2542	2545	2548	2551	2554	2558	2563
2566	2568	2571	2575	2578	2582	2584	2587	2588	2589
2590	2594	2597	2600	2603	2606	2610	2614	2617	2621
2624	2628	2631	2639	2645	2649	2652	2659	2663	2668
2672	2679	2683	2687	2692	2693	2695	2699	2704	2707
2710	2715	2719	2725	2729	2733	2737	2741	2743	2750
2763	2768	2773	2779	2782	2786	2790	2794	2797	2802
2805	2809	2813	2817	2821	2825	2828	2832	2836	2839
2843	2847	2849	2853	2856	2860	2862	2865	2870	2874
2878	2882	2885	2890	2893	2898	2902	2905	2909	2913
2917	2920	2924	2928	2931	2933	2937	2941	2943	2946
2949	2952	2955	2958	2959	2962	2964	2966	2968	2971
2975	2981	2982	2983	2997	3002	3005	3008	3010	3014
3019	3021	3025	3028	3032	3035	3038	3042	3045	3048
3051	3054	3057	3060	3065	3067	3070	3074	3077	3080
3083	3086	3090	3093	3097	3100	3103	3110	3117	3119
3123	3126	3130	3133	3138	3142	3145	3150	3153	3159
3163	3167	3172	3176	3179	3181	3183	3186	3190	3193
3195	3198	3201	3202	3203	3205	3206	3207	3212	3214
3218	3222	3225	3228	3231	3234	3237	3239	3245	3247
3250	3253	3255	3259	3262	3266	3270	3273	3276	3280
3284	3289	3292	3296	3298	3301	3303	3304	3306	3307
3309	3313	3317	3320	3323	3327	3329	3332	3334	3340
3343	3346	3349	3352	3355	3358	3362	3364	3366	3369
3372	3375	3379	3382	3384	3387	3390	3399	3402	3405
3408	3410	3413	3416	3419	3422	3424	3427	3428	3430
3432	3434	3436	3440	3441	3447	3449	3454	3457	3459
3462	3463	3465	3468	3470	3472	3474	3476	3479	3481
3483	3488	3490	3492	3494	3496	3498	3501	3503	3504
3505	3506	3509	3512	3514	3516	3519	3522	3524	3527
3529	3532	3535	3538	3540	3543	3546	3549	3551	3552
3553	3557	3559	3561	3563	3566	3568	3570	3572	3574
3577	3580	3583	3586	3588	3591	3594	3597	3601	3605
3610	3625	3629	3632	3635	3638	3641	3646	3651	3654
3658	3660	3663	3667	3669	3673	3676	3680	3683	3686
3689	3693	3695	3699	3702	3704	3706	3709	3712	3715
3718	3722	3724	3727	3729	3732	3735	3739	3742	3746
3750	3751	3754	3758	3760	3764	3767	3771	3774	3778

3781	3783	3786	3790	3794	3798	3802	3805	3808	3810
3814	3818	3819	3820	3823	3826	3829	3830	3834	3838
3841	3845	3850	3852	3853	3857	3861	3864	3867	3871
3875	3878	3881	3884	3888	3891	3894	3897	3901	3905
3910	3914	3918	3921	3925	3930	3934	3938	3943	3948
3951	3955	3958	3962	3965	3969	3975	3977	3982	3986
3990	3993	3997	4002	4005	4009	4013	4016	4020	4023
4027	4030	4033	4036	4039	4043	4049	4055	4059	4062
4066	4068	4071	4073	4076	4080	4082	4085	4086	4088
4091	4095	4097	4101	4104	4108	4115	4117	4121	4123
4126	4128	4131	4134	4136	4138	4140	4142	4144	4146
4149	4151	4154	4156	4158	4161	4164	4167	4168	4171
4174	4176	4180	4186	4189	4192	4195	4197	4199	4201
4203	4205	4209	4212	4214	4220	4222	4224	4227	4230
4232	4235	4237	4239	4241	4244	4247	4249	4251	4253
4255	4257	4259	4262	4265	4267	4270	4271	4275	4282
4289	4293	4296	4300	4303	4307	4309	4312	4315	4318
4321	4323	4326	4330	4332	4335	4337	4340	4342	4345
4346	4348	4350	4352	4354	4356	4358	4360	4363	4367
4369	4372	4374	4377	4379	4381	4383	4384	4387	4389
4392	4397	4399	4401	4403	4405	4410	4412	4414	4416
4418	4421	4424	4426	4429	4432	4434	4437	4439	4440
4441	4443	4446	4449	4452	4455	4458	4461	4464	4467
4469	4472	4475	4478	4479	4482	4485	4487	4490	4493
4496	4499	4502	4504	4507	4510	4513	4517	4520	4523
4526	4530	4535	4539	4543	4547	4551	4554	4555	4558
4561	4564	4568	4569	4573	4576	4579	4584	4588	4589
4591	4596	4601	4606	4612	4618	4625	4630	4635	4641
4645	4650	4654	4670	4678	4684	4689	4695	4707	4715
4722	4727	4733	4737	4743	4747	4752	4758	4761	4765
4769	4773	4777	4780	4784	4786	4791	4794	4798	4803
4812	4817	4822	4827	4833	4836	4840	4846	4850	4853
4858	4861	4864	4868	4871	4876	4879	4883	4887	4891
4894	4899	4903	4906	4911	4915	4919	4922	4927	4931
4935	4940	4953	4957	4962	4968	4972	4977	4983	4986
4990	4994	4999	5005	5010	5015	5020	5029	5035	5041
5046	5053	5058	5062	5068	5091	5101	5115	5119	5123
5124	5129	5133	5138	5143	5148	5152	5157	5161	5165
5170	5176	5180	5183	5188	5193	5197	5200	5205	5209
5213	5218	5224	5231	5235	5239	5243	5248	5253	5258
5264	5268	5271	5272	5275	5279	5284	5288	5291	5295
5299	5303	5307	5310	5315	5318	5322	5326	5330	5334
5338	5341	5345	5348	5353	5356	5360	5361	5364	5367
5368	5371	5374	5377	5380	5384	5386	5390	5393	5395
5399	5400	5403	5406	5409	5412	5415	5417	5420	5423
5426	5429	5433	5436	5440	5444	5448	5452	5455	5457
5460	5463	5466	5469	5472	5478	5482	5484	5487	5489
5494	5496	5498	5499	5502	5506	5508	5511	5514	5517
5519	5522	5524	5528	5530	5532	5535	5538	5541	5552
5558	5561	5568	5571	5574	5577	5580	5583	5587	5590
5594	5598	5601	5605	5609	5612	5615	5617	5625	5628
5632	5634	5640	5643	5646	5648	5651	5656	5659	5662
5666	5669	5673	5678	5682	5683	5685	5689	5693	5697
5703	5706	5710	5714	5719	5723	5729	5733	5739	5745

5748	5752	5756	5760	5764	5767	5772	5776	5779	5783
5788	5792	5796	5797	5799	5803	5807	5811	5816	5821
5824	5829	5832	5836	5842	5847	5853	5858	5862	5867
5868	5871	5878	5888	5897	5903	5916	5923	5929	5936
5941	5946	5950	5956	5961	5965	5970	5976	5981	5986
5992	5999	6003	6004	6009	6015	6021	6027	6032	6038
6041	6045	6051	6056	6062	6067	6072	6078	6083	6088
6092	6096	6101	6105	6110	6115	6120	6124	6129	6133
6138	6144	6149	6154	6159	6163	6168	6173	6179	6184
6188	6193	6198	6204	6208	6212	6216	6220	6224	6225
6229	6233	6237	6242	6248	6251	6255	6258	6262	6267
6270	6274	6279	6284	6288	6294	6300	6304	6308	6310
6316	6320	6325	6330	6335	6340	6345	6351	6356	6360
6364	6368	6371	6377	6381	6385	6389	6394	6398	6402
6405	6409	6414	6417	6421	6425	6429	6433	6436	6440
6445	6449	6455	6458	6462	6467	6472	6476	6480	6485
6498	6506	6520	6525	6529	6533	6538	6542	6545	6548
6552	6557	6564	6569	6572	6578	6589	6611	6617	6626
6632	6638	6644	6648	6654	6658	6664	6668	6673	6675
6681	6687	6693	6697	6703	6707	6712	6717	6724	6729
6734	6738	6744	6749	6756	6762	6767	6773	6779	6785
6791	6797	6802	6808	6813	6819	6824	6828	6832	6833
6835	6840	6843	6850	6854	6858	6861	6869	6870	6873
6876	6879	6882	6883	6886	6887	6888	6890	6891	6892
6898	6903	6907	6917	6920	6923	6927	6930	6935	6939
6943	6947	6950	6953	6956	6960	6964	6967	6970	6974
6978	6981	6984	6989	6992	6995	6998	7001	7002	7006
7009	7011	7013	7016	7018	7021	7024	7029	7036	7038
7041	7044	7046	7047	7049	7052	7055	7060	7061	7063
7066	7069	7073	7076	7080	7084	7088	7091	7095	7096
7098	7101	7104	7108	7112	7115	7116	7119	7123	7126
7129	7133	7135	7139	7141	7144	7148	7151	7154	7157
7160	7164	7168	7172	7175	7180	7183	7188	7192	7199
7203	7208	7212	7216	7220	7225	7230	7233	7235	7240
7243	7247	7251	7255	7260	7265	7269	7272	7276	7280
7285	7288	7293	7297	7301	7305	7309	7312	7316	7321
7325	7329	7333	7336	7340	7344	7348	7351	7356	7367
7372	7374	7386	7391	7395	7399	7403	7408	7412	7416
7424	7431	7446	7449	7454	7458	7464	7468	7472	7478
7481	7484	7487	7491	7494	7497	7500	7503	7506	7510
7514	7517	7521	7524	7527	7530	7534	7539	7541	7544
7547	7551	7554	7558	7560	7563	7566	7571	7575	7578
7582	7586	7591	7597	7601	7602	7606	7610	7614	7618
7623	7627	7632	7635	7638	7643	7646	7647	7650	7651
7655	7656	7658	7661	7662	7665	7666	7670	7674	7680
7681	7694	7697	7701	7705	7709	7711	7715	7719	7722
7726	7728	7733	7736	7739	7744	7748	7752	7755	7760
7763	7768	7772	7777	7781	7788	7792	7793	7795	7799
7803	7808	7813	7816	7820	7827	7831	7834	7837	7842
7847	7852	7857	7861	7866	7872	7877	7883	7887	7892
7896	7897	7902	7907	7912	7917	7922	7931	7936	7942
7946	7952	7958	7964	7976	7986	7991	7997	8004	8010
8017	8024	8031	8038	8044	8052	8061	8069	8078	8086
8093	8101	8110	8120	8121	8128	8137	8143	8149	8156

```
8161 8168 8169 8175 8180 8188 8189 8192 8198 8204
8210 8216 8222 8238 8245 8249 8255 8260 8264 8268
8276 8280 8284 8292 8296 8301 8306 8309 8310 8312
8316 8320 8324 8327 8332 8336 8338 8341 8344 8347
8350 8353 8357 8359 8363 8366 8370 8374 8377 8380
8384 8386 8390 8397 8399 8402 8404 8408 8412 8414
8417 8420 8422 8425 8429 8431 8434 8438 8442 8445
8448 8451 8453 8456 8459 8463 8467 8470 8474 8478
8481 8484 8487 8488 8491 8495 8499 8504 8508 8513
8517 8522 8527 8528 8532 8536 8541 8546 8550 8554
8559 8564 8568 8572 8577 8582 8586 8592 8596 8597
8601 8607 8612 8618 8624 8629 8630 8634 8639 8643
8647 8651 8654 8659 8663 8667 8671 8674 8677 8681
8684 8690 8693 8698 8700 8703 8706 8709 8713 8716
8719 8722 8726 8728 8731 8734 8738 8742 8745 8748
8751 8755 8758 8762 8766 8769 8772 8775 8777 8779
8782 8784 8787 8790 8793 8795 8797 8801 8804 8807
8812 8814 8817 8819 8822 8825 8831 8834 8837 8840
8843 8847 8850 8853 8856 8859 8864 8866 8870 8873
8876 8881 8886 8889 8893 8897 8902 8905 8909 8912
8916 8921 8925 8929 8934 8938 8943 8946 8950 8955
8960 8964 8970 8975 8979 8983 8988 8992 8995 8999
9003 9010 9013 9015 9018 9021 9022 9025 9028 9032
9033 9036 9040 9044 9048 9051 9056 9057 9059 9065
9069 9074 9077 9081 9085 9091 9095 9098 9102 9105
9108 9112 9113 9116 9120 9123 9128 9132 9136 9141
9146 9150 9154 9159 9163 9171 9179 9182 9191 9196
9201 9207 9211 9215 9219 9224 9229 9233 9240 9245
9250 9251 9254 9258 9262 9265 9268 9272 9275 9279
9282 9288 9294 9296 9299 9302 9306 9310 9314 9318
9321 9325 9328 9332 9335 9339 9343 9347 9350 9353
9356 9360 9363 9367 9371 9376 9380 9385 9390 9395
9400 9406 9410 9412 9416 9419 9420 9424 9427 9428
9430 9435 9438 9442 9446 9449 9452 9457 9460 9463
9467 9470 9473 9476 9479 9483 9488 9490 9493 9500
9503 9505 9518 9523 9526 9530 9532 9535 9539 9543
9547 9548 9551 9554 9558 9562 9565 9569 9573 9577
9580 9582 9586 9590 9593 9598 9602 9605 9609 9612
9616 9620 9624 9625 9629 9632 9636 9639 9642 9646
9649 9652 9654 9658 9663 9666 9670 9672 9676 9679
9683 9686 9689 9693 9697 9702 9706 9710 9715 9718
9722 9725 9726 9731 9735 9738 9741 9745 9748 9751
9756 9760 9765 9769 9772 9775 9779 9784 9788 9791
9795 9799 9803 9807 9811 9815 9818 9823 9827 9830
9835 9837 9841 9845 9849 9851 9855 9858 9861 9865
9867 9871 9874 9890 9896 9901 9906 9911 9915 9922
9928 9930 9933 9934 9939 9943 9948 9950 9954 9957
9961 9965 9970 9975 9976 9983 9991 9994
```

For a total of 2652 extensions scanned.

<p>Section 1/ Question 4b. Categorize these extensions into the following groups, and explain to method you used:</p> <ul style="list-style-type: none"> <li>• Those that exist on the honeypot, AND require authentication</li> <li>• Those that exist on the honeypot, and do NOT require authentication</li> <li>• Those that do not exist on the honeypot</li> </ul>	Possible Points: 6pts
Tools Used: SIPlogparser.rb, grep, awk, uniq, sort	
<p>Answer</p> <p>Because the given log file doesn't reveal the SIP PBX responses, we can only base our analysis on the client (attacker) behavior. The previous analysis has taught us some of the behavior of the supposed used tools.</p> <p>Here are the assumptions I've made:</p> <p>In the second phase of the attack (svwar.py) REGISTER requests were sent, trying to register 2652 extensions. For each extension scanned, the attacker has received a response from the honeypot containing a status-code:</p> <ul style="list-style-type: none"> <li>• For the valid extensions that do not require any authentication, the responses may have been: <ul style="list-style-type: none"> <li>◦ Status-code: 100 Trying and 200 Ok</li> </ul> </li> <li>• For the valid extensions that require authentication, the responses may have been: <ul style="list-style-type: none"> <li>◦ Status-code: 100 Trying and 401 Unauthorized</li> </ul> </li> <li>• And finally for the extensions that do not exist on the honeypot, the responses may have been: <ul style="list-style-type: none"> <li>◦ Status-Code: 404 Not found.</li> </ul> </li> </ul> <p>Based on these response status-codes, extensions were categorized automatically for the attacker, and only extensions which required authentication needed to be scanned again, but this time with svcrack.py, trying to brute-force passwords. (phase 3). So, to find those extensions that require authentication, one can filter REGISTER messages containing an "Authorization" header field set. For example with this command line:</p> <pre>franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4\$ cat logs_v3.txt  grep "Authorization: "  awk '{split(\$3,fields,",");print fields[1]}'  sort  uniq username="101" username="102" username="103" username="111"</pre> <p>Then, we can analyze phase 4 of the attack (starting on 2010-05-05 10:00:08.170954 and from 89.42.194.X), the "exploitation" par. In this phase, the attacker has used 2 extensions to call real telephone numbers, these two extensions were: 100 and 101. We already know the 101 was a valid extension but require authentication, but what about 100 ? 89.42.194.X has been able to use register and then use extension number 100 without any authentication, so we can consider that this extension was a valid one that do not required authentication.</p> <p>All the other extensions that were scanned were invalid on this honeypot.</p> <p>Summary:</p> <ul style="list-style-type: none"> <li>• Valid extensions without need for authentication: 100</li> <li>• Valid with authentication required: 101, 102, 103, 111</li> <li>• Invalid: all the other.</li> </ul>	

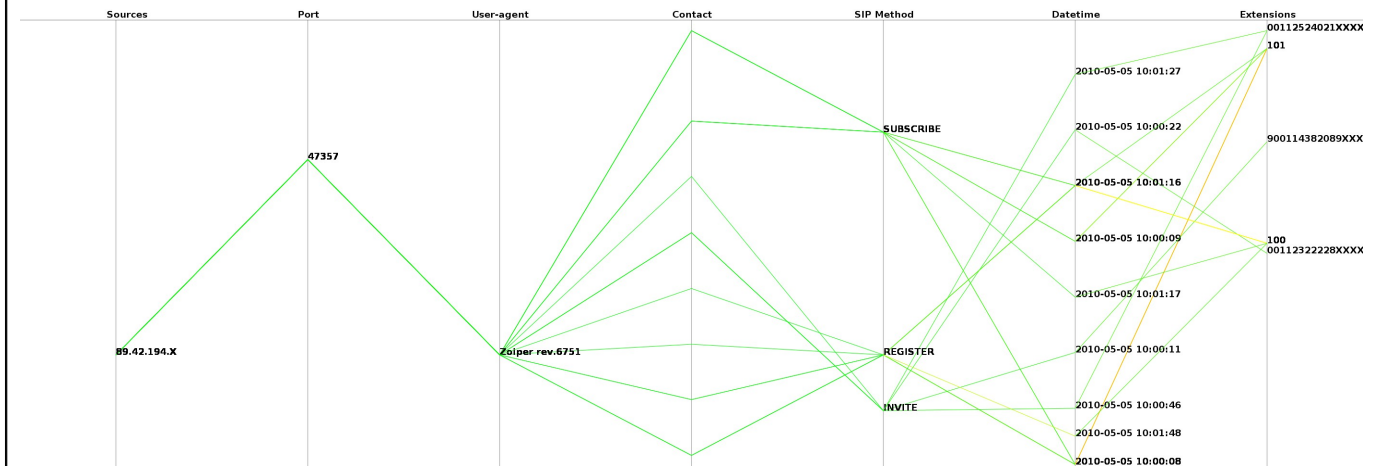


Section 1/ Question 5. Was a real SIP client used at any point ? If it was, what time was it used, and why ? Possible Points: 1pt

Tools Used: SIPlogparser.rb, picviz

Answer

On the big picture.png graph we have notice that 2 different User-agents were used. The first UA: "friendly-scanner" is known to be used by the SIPvicious tools. The second UA is **Zoiper rev.6751**, meaning that a ZoIPer SIP client was used. Let's try to depict this with a picviz graph:



[<http://malphx.free.fr/dotclear/public/HPFC4-stuff.zip/zoiper.png>]  
 This SIP client was used by 89.42.194.X, between 2010-05-05 10:00:08.170954 UTC and 2010-05-05 10:01:48.058434 UTC, to REGISTER and then use two extensions (100 and 101) from the domain honey.pot.IP.removed.  
 These two extensions were used to call 3 international numbers:

- at 10:00:11.493635 UTC => 900114382089XXXX
- at 10:00:22.019093 UTC => 00112322228XXXX
- at 10:00:46.147670 UTC => 00112524021XXXX
- at 10:01:27.633156 UTC => 00112524021XXXX

Section 1/ Question 6. List the following, include geo-location information.  
 - Source IP addresses involved Possible Points: 3pts  
 - The real world phone numbers that were attempted to be dialled

Tools Used: whois, geoipllookup

Answer

We know that only two sources appear in the log file: 210.184.X.Y and 89.42.194.0. Now, we can replace X and Y by 0 and then try to gather informations on them from whois and geoipllookup databases:

```
franc@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ whois 210.184.0.0
% [whois.apnic.net node-3]
% Whois data copyright terms    http://www.apnic.net/db/dbcopyright.html

inetnum:        210.184.0.0 - 210.184.31.255
netname:        CPCNET-HK
descr:          CPCNet Hong Kong Ltd.
descr:          20/F, Lincoln House,
descr:          Taikoo Place,
```

```

descr:      979 King's Road, Quarry Bay,
descr:      Hong Kong
country:    HK
admin-c:    NC155-AP
tech-c:     NC154-AP
mnt-by:     APNIC-HM
mnt-lower:  MAINT-HK-CPCNET
changed:    hm-changed@apnic.net 20020823
status:     ALLOCATED PORTABLE
source:     APNIC

```

```

person:     CPCNet Hong Kong Limited NOC
address:    20/F, Lincoln House,
address:    Taikoo Place,
address:    979 King's Road,
address:    Quarry Bay,
address:    Hong Kong
country:    HK
phone:      +852-2170-7101
fax-no:     +852-2372-0287
e-mail:     hostinfo@cpcnet.com
nic-hdl:    NC154-AP
mnt-by:     MAINT-HK-CPCNET
changed:    hostinfo@cpcnet.com 20100106
source:     APNIC

```

```

franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ geoiplookup 210.184.0.0
GeoIP Country Edition: HK, Hong Kong

```

The first attacker comes from Hong Kong.

```

franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ whois 89.42.194.0
% This is the RIPE Database query service.
% The objects are in RPSL format.
%
% The RIPE Database is subject to Terms and Conditions.
% See http://www.ripe.net/db/support/db-terms-conditions.pdf

% Note: This output has been filtered.
%       To receive output for a database update, use the "-B" flag.

% Information related to '89.42.192.0 - 89.42.199.255'

inetnum:    89.42.192.0 - 89.42.199.255
netname:    SC-UNIREA-EL-NINO-SRL
descr:      SC Unirea El Nino SRL
descr:      Unirii G1
descr:      Constanta Constanta
country:    ro
admin-c:    CDG40-RIPE
tech-c:     CDG40-RIPE
status:     ASSIGNED PA
remarks:    Registered through http://www.jump.ro/ip.html
mnt-by:     RO-MNT
mnt-lower:  RO-MNT

```

```
mnt-routes:    Unirea-El-Nino-MNT
source:        RIPE # Filtered

person:        Cristea Dragos George
address:       Str. Aviator Vasile Craiu Nr.
address:       30, Constanta,
address:       Romania
phone:         +40-722-462287
e-mail:        the_angelro@yahoo.com
nic-hdl:       CDG40-RIPE
mnt-by:        AS3233-MNT
source:        RIPE # Filtered

% Information related to '89.42.192.0/21AS41763'

route:         89.42.192.0/21
descr:         SC Unirea El Nino SRL
origin:        AS41763
mnt-by:        Unirea-El-Nino-MNT
source:        RIPE # Filtered
```

```
franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ geoipllookup 89.42.194.0
GeoIP Country Edition: RO, Romania
```

And the second, from Romania.

The Romanian attacker has tried to call these numbers:

- 900114382089XXXX
  - 9 + 0011 + 43 + 820 + 89XXXX
  - Location: Austria
  - Type: service number
- 00112322228XXXX
  - 0011 + 232 + 22 + 28XXXX
  - Location: Sierra Leone
- 00112524021XXXX
  - 0011 + 252 + 40 + 21XXXX
  - Location: Somalia

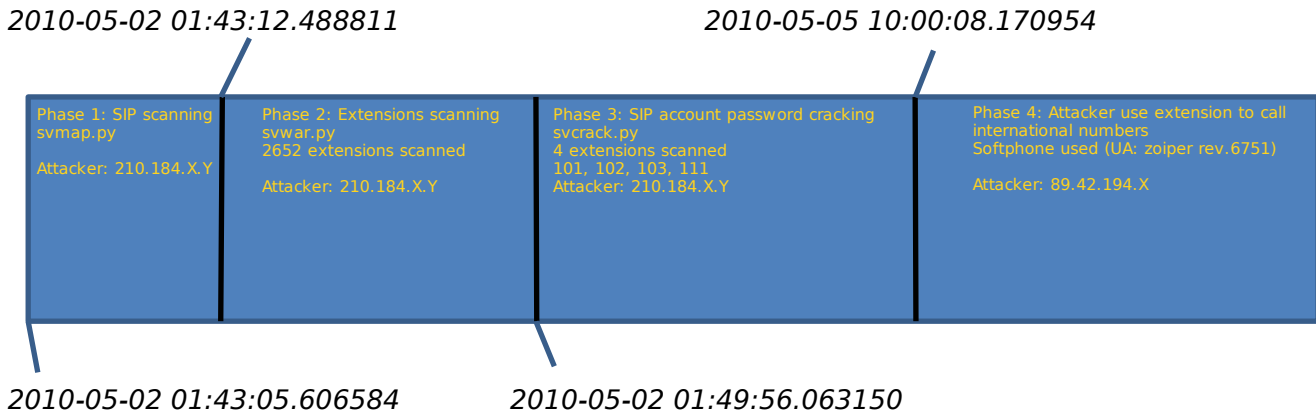
Section 1/ Question 7. Draw a simple static or animated timeline of events, describing when and *where* certain phases occurred from, and what the purpose of each phase was

Possible Points: 5pts

Tools Used:

Answer

I've decomposed this incident in 4 phases:



- **Phase 1 : Finding SIP servers**
  - Phase start: 2010-05-02 01:43:05.606584
  - Phase end: before 2010-05-02 01:43:12.488811
  - Attacker: 210.184.X.Y (Hong Kong)
  - Tool used: SIPvicious svmap.py
  - Description:
    - In this phase the attacker launch scan session on selected subnets to find valid SIP servers. In this event, the scan was done with a SIP OPTIONS message (default method used by svmap.py)
- **Phase 2: Finding valid extensions on the targeted SIP server**
  - Phase start: 2010-05-02 01:43:12.488811
  - Phase end: 2010-05-02 01:49:46.992699
  - Attacker: 210.184.X.Y (Hong Kong)
  - Tool used: SIPvicious svwar.py
  - 2652 extensions scanned (44 named extensions and 2608 numbered extensions)
  - Description:
    - The attacker tries to find valid extensions (with or without authentication required). He enumerates large range of extensions using SIP REGISTER requests. Based on the received response code from the SIP server, extensions are categorized: valid, valid with authentication, invalid.
- **Phase 3: Cracking password protected SIP accounts**
  - Phase start: 2010-05-02 01:49:56.063150
  - Phase end: 2010-05-02 01:55:11.496170
  - Attacker: 210.184.X.Y (Hong Kong)
  - Tool used: SIPvicious svcrack.py
  - 4 extensions were brute-forced: 102, 103, 101 and 111
  - Description:

- with the informations gathered in phase 2, the attacker tries to crack extension passwords, for those which need authentication. Svcrack.py has been used for doing the job.

If you look carefully at each phase starting and ending timestamps, you can notice that the delta between the end of one phase and the beginning of the next one is particularly small for the 3 first phases. This leads to think that the attacker has automated those phases, maybe in a script which gathers SIPvicious tools outputs and launch adequate actions to do next.

- **Phase 4: Owning and Using Extensions 100 & 101**

- Phase start: 2010-05-05 10:00:08.170954
- Phase end: 2010-05-05 10:01:48.058434
- Attacker: 89.42.194.X (Romania)
- Tool used: a Softphone, User-agent: Zoiper rev6751
- **3 international numbers were called**
  - **900114382089XXXX (on 2010-05-05 10:00:11.493635)**
  - **00112322228XXXX (on 2010-05-05 10:00:22.019093)**
  - **00112524021XXXX (on 2010-05-05 10:00:46.147670 and on 2010-05-05 10:01:27.633156)**
- Description:
  - This is the last phase of the incident. Another host, based in Romania registers and then use extensions 101 and 100. Although, extension 101 was protected by a password, the attacker correctly authenticates itself and can use the extension to call external numbers. (the password was known to the attacker). This could lead to think that the attacker was the same as in the other phase or maybe he has bought or received SIP usernames and passwords list gathered from the previous phase.

<p>Section 1/ Question 8a. Assuming this were a real incident, write 2 paragraphs of an Executive summary of this incident. Assume the reader does not have IT Security or VOIP experience.</p> <p>a) First Paragraph: Write, in the minimum detail necessary a description the nature and timings, and possible motives of the attack phases. (3 points)</p>	<p>Possible Points: 3pts</p>
<p>Tools Used:</p>	
<p>Answer</p> <p>Our VoIP system was targeted by a four phases VoIP attack starting on Sunday 2010-05-02 01:43:05 and was initially launched from an host located in Hong Kong. The three first phases of this attack were run on Sunday May, 2<sup>nd</sup> between 01:43:05 UTC and 01:55:11 UTC; the objectives were clearly information leakage of our internals VoIP extensions. Our analysis of this incident, based on log files, has reveal that the attacker has used publicly available tools to scan and exploit SIP based VoIP systems. The used tool suite is named SIPvicious.</p> <p>From the first three phases of the attack, the attacker has been able to find our VoIP system, to enumerate valid extensions and to crack at least one of our extensions password. This information leakage has lead to an unauthorized use of two of our extensions to mahe international telephone calls in the fourth phase of this incident.</p> <p>This last phase started on Wednesday May,5<sup>th</sup> at 10:00:08 UTC and was run by another host located in Romania. This attacker has successfully used extension number 100 and 101 of our telephony system to make four international calls. The callee were based in Austria (a service number), Sierra-Leone and Somalia. This last event ended on Wednesday May,5<sup>th</sup> at 10:01:48 UTC.</p> <p>Possible motives of the attack may be impersonation, ability to make free international calls or any other nefarious activities.</p>	

<p>Section 1/ Question 8b. Assuming this were a real incident, write 2 paragraphs of an Executive summary of this incident. Assume the reader does not have IT Security or VOIP experience.</p> <p>b) Second Paragraph: What actions would you recommend should occur following this particular incident, include any priority/urgency. Also describe any good practices that should be employed to mitigate future attacks.</p>	<p>Possible Points: 3pts</p>
<p>Tools Used:</p>	
<p>Answer</p> <p>This incident has revealed several weakness in our actual VoIP system implementation:</p> <ul style="list-style-type: none"> <li>• Some extensions doesn't require any authentication (e.g. extension number 100) <ul style="list-style-type: none"> <li>◦ <i>Anyone (internal or external) can register and then use these extensions</i></li> </ul> </li> <li>• Lack of publicly known SIP scanning tools detection <ul style="list-style-type: none"> <li>◦ <i>Our IDS/IPS systems were unable to detect/prevent this attack.</i></li> </ul> </li> <li>• Our actual VoIP protocols implementations don't use encryption and mutual authentication <ul style="list-style-type: none"> <li>◦ <i>This lead to potential eavesdrops on our communications or VoIP signaling exchanges.</i></li> <li>◦ <i>Critical informations can be gathered by an attacker.</i></li> </ul> </li> </ul> <p>Based on these facts, our recommendations to prevent or mitigate this kind of incident are listed below:</p> <ul style="list-style-type: none"> <li>• <b>Priority: Urgent</b> <ul style="list-style-type: none"> <li>◦ Define and implement IDS signatures to detect this tool suite in our IDS/IPS <ul style="list-style-type: none"> <li>▪ <i>Our analysis has revealed that the tools were used with their default values. Some of these values can be used to define IDS/IPS signatures and detect or prevent similar future attacks</i></li> </ul> </li> <li>◦ Limit the number of SIP messages (REGISTER) from external client on our perimeter devices <ul style="list-style-type: none"> <li>▪ <i>SIP scanning and extensions enumeration, like any other network scanning technique involve sending large amount of requests. This behavior can be deteted at the perimeter, as it is generally unusual in normal usage of the VoIP system.</i></li> <li>▪ <i>We must quickly define a baseline of "normal" SIP protocol requests usage (REGISTER, OPTIONS, SUBSCRIBE, INVITE...) from internal and external clients and then limit the amount of these requests adequately.</i></li> </ul> </li> <li>◦ Protect all our extensions with a strong password <ul style="list-style-type: none"> <li>▪ <i>even if tools can be used to crack SIP account password, we must protect all our extension with a password. The chosen password must be sufficiently hard to guess (not less than 8 chars length, using letters (uppercase and lowercase), numbers and special chars)</i></li> <li>▪ <i>This will, at least, slow brute-force attack against our system.</i></li> </ul> </li> <li>◦ <b>Priority: High</b> <ul style="list-style-type: none"> <li>▪ <i>We should consider using secured version of our VoIP protocols. Instead of using unencrypted SIP, we'll have to use SIP over TLS (SIPS). This will provide two-way authentication, confidentiality and messages integrity through the use of strong encryption.</i></li> <li>▪ <i>We should consider installing a Session Border Controller. This device will protect our SIP servers and devices from various VoIP attacks.</i></li> </ul> </li> </ul> </li> </ul>	

Section 2/ Question 1. Which 4 protocols are involved in the PCAP (VOIP protocols and otherwise) ? Give a brief explanation as to their purpose.	Possible Points: 4pts
--	--------------------------

Tools Used: tshark

Answer

As usual for me when I begin a pcap analysis, I use the Protocol Hierarchy Statistics of tshark to get a view of the protocols involved in the attack.

```

=====
Protocol Hierarchy Statistics
Filter: frame

frame                frames:4447 bytes:1117758
  eth                frames:4447 bytes:1117758
    ip               frames:4447 bytes:1117758
      udp            frames:3154 bytes:662385
        sip          frames:19 bytes:11971
          rtcp        frames:21 bytes:3734
            rtcp      frames:21 bytes:3734
              rtcp.length_check frames:20 bytes:3532
                rtcp  frames:1 bytes:202
                  rtcp.length_check frames:1 bytes:202
            rtp        frames:3113 bytes:646634
              rtp.event frames:125 bytes:7250
            data       frames:1 bytes:46
          tcp          frames:1291 bytes:454993
            http       frames:166 bytes:85025
              data-text-lines frames:9 bytes:7085
                xml      frames:3 bytes:1512
                  png     frames:7 bytes:6412
                    image-gif frames:1 bytes:355
              tcp.segments frames:18 bytes:12536
                http     frames:18 bytes:12536
                  data-text-lines frames:16 bytes:11450
                    png   frames:1 bytes:296
                      image-jfif frames:1 bytes:790
            icmp       frames:2 bytes:380
=====

```

**SIP (Session Initiation Protocol)**

*SIP is a signaling protocol used for controlling multimedia communication sessions, like voice or video calls over IP. The latest version of SIP is defined in RFC 3261. The protocol can be used for creating, modifying and terminating two-party or multiparty sessions consisting of one or several media streams.*

*The SIP protocol is an Application Layer protocol designed to be independent of the underlying transport layer; it can run on Transmission Control Protocol (TCP), User Datagram Protocol (UDP), or Stream Control Transmission Protocol (SCTP).[3] It is a text-based protocol, incorporating many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP), [4] allowing for direct inspection by administrators.*

Source en.wikipedia.org

Default port: 5060.

In this capture file, SIP is used to create and tear down VoIP sessions.



**RTP (Real-time Transport Protocol)**

*The Real-time Transport Protocol (RTP) defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the IETF and first published in 1996 as RFC 1889, and superseded by RFC 3550 in 2003.*

*RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications and web-based push to talk features. For these it carries media streams controlled by H.323, MGCP, Megaco, SCCP, or Session Initiation Protocol (SIP) signaling protocols, making it one of the technical foundations of the Voice over IP industry.*

*RTP is usually used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g., audio and video) or out-of-band events signaling (DTMF in separate payload type), RTCP is used to monitor transmission statistics and quality of service (QoS) information. When both protocols are used in conjunction, RTP is usually originated and received on even port numbers, whereas RTCP uses the next higher odd port number.*

Source en.wikipedia.org

In this capture file, RTP is used as the media protocol to transport voice.

**RTCP (Real-time Transport Control Protocol)**

*The Real-Time Transport Control Protocol (RTCP) is a sister protocol of the Real-time Transport Protocol (RTP). Its basic functionality and packet structure is defined in the RTP specification RFC 3550,[1] superseding its original standardization in 1996 (RFC 1889).*

*RTCP provides out-of-band statistics and control information for an RTP flow. It partners RTP in the delivery and packaging of multimedia data, but does not transport any media streams itself. Typically RTP will be sent on an even-numbered UDP port, with RTCP messages being sent over the next highest odd-numbered port[2]. The primary function of RTCP is to provide feedback on the quality of service (QoS) in media distribution by periodically sending statistics information to participants in a streaming multimedia session.*

*RTCP gathers statistics for a media connection and information such as transmitted octet and packet counts, lost packet counts, jitter, and round-trip delay time. An application may use this information to control quality of service parameters, perhaps by limiting flow, or using a different codec.*

*RTCP itself does not provide any flow encryption or authentication methods. Such mechanisms may be implemented, for example, with the Secure Real-time Transport Protocol (SRTP) defined in RFC 3711.*

Source en.wikipedia.org

**HTTP (HyperText Transport Protocol)**

*The Hypertext Transfer Protocol (HTTP) is an Application Layer protocol for distributed, collaborative, hypermedia information systems.[1]*

*HTTP is a request-response protocol standard for client-server computing. In HTTP, a web browser, for example, acts as a client, while an application running on a computer hosting the web site acts as a server. The client submits HTTP requests to the responding server by sending messages to it. The server, which stores content (or resources) such as HTML files and images, or generates such content on the fly, sends messages back to the client in response. These returned messages may contain the content requested by the client or may contain other kinds of response indications. A client is also referred to as a user agent (or 'UA' for short). A web crawler (or 'spider') is another example of a common type of client or user agent.*

*In between the client and server there may be several intermediaries, such as proxies, web caches*

or gateways. In such a case, the client communicates with the server indirectly, and only converses directly with the first intermediary in the chain. A server may be called the origin server to reflect the fact that this is where content ultimately originates from. HTTP is not constrained in principle to using TCP/IP, although this is its most popular implementation platform. Indeed HTTP can be "implemented on top of any other protocol on the Internet, or on other networks." HTTP only presumes a reliable transport; any protocol that provides such guarantees can be used.[2]

Source en.wikipedia.org

In this capture file, HTTP is used to communicate with the GUI frontend of the SIP PBX.

Section 2/ Question 2a. Which codec does the RTP stream use? Possible Points: 1pt

Tools Used: tshark

Answer

Tshark can give useful informations on RTP streams within a PCAP file:

```

franc@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ tshark -r Forensic_challenge_4.pcap -qz rtp,streams
===== RTP Streams =====
  Src IP addr  Port    Dest IP addr  Port    SSRC          Payload  Pkts      Lost  Max
Delta(ms)  Max Jitter(ms) Mean Jitter(ms) Problems?
  172.25.105.3 63184   172.25.105.40 18150  0xA254E017  ITU-T G.711 PCMU  1811    -30 (-1.7%)
1940.06      122.24      11.28 X
  172.25.105.40 18150   172.25.105.3 63184  0x42AFE59B  ITU-T G.711 PCMU  1302     0 (0.0%)
56.05        3.43         0.32 X
=====
    
```

These stats indicate that the G.711  $\mu$ -law (or u-law) codec was used for the VoIP call. Some good infos on G.711 codecs can be found here: (<http://www.en.voipforo.com/codec/codecs-g711-alaw.php>)

Section 2/ Question 2b. How long is the sampling time (in milliseconds)?	Possible Points: 1pt
Tools Used:	
<p>Answer</p> <p>G.711 family of codecs use a sampling frequency of 8kHz (8000 Hz). Meaning, the voice or audio stream is sampled 8000 times in one (1) second.</p> <p>So, the sampling time or length of one sample is <math>1/8000 \text{ s} = 0.000125 \text{ s} = \underline{0.125 \text{ ms}}</math></p>	

Section 2/ Question 3. How did the attacker gain access to the server? List ways this could have been prevented.	Possible Points: 2pts
Tools Used: tshark	
<p>Answer</p> <p>At the beginning of the attack, the attacker (172.25.105.43) sent a SIP OPTIONS request for extension 100 at 172.25.105.40. (packet #1).</p> <p>The tool used by the attacker seems to be svmap.py from the SIPvicious suite.</p> <p>Luckily, 172.25.105.40 responded to the request with a 200 OK response and a lot of information on the targeted extension. (packet #2)</p> <p>Here an extract of the informations returned:</p> <pre> franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4\$ tshark -r Forensic_challenge_4.pcap -VR "frame.number==2" Frame 2 (560 bytes on wire, 560 bytes captured)   Arrival Time: May  1, 2010 20:13:00.948226000     [Time delta from previous captured frame: 0.000353000 seconds]     [Time delta from previous displayed frame: 0.000353000 seconds]     [Time since reference or first frame: 0.000353000 seconds]   Frame Number: 2   Frame Length: 560 bytes   Capture Length: 560 bytes   [Frame is marked: False]   [Protocols in frame: eth:ip:udp:sip] Ethernet II, Src: IntelCor_87:cf:96 (00:21:6a:87:cf:96), Dst: IntelCor_9f:78:c6 (00:13:ce:9f:78:c6)   Destination: IntelCor_9f:78:c6 (00:13:ce:9f:78:c6)     Address: IntelCor_9f:78:c6 (00:13:ce:9f:78:c6)       .... 0... = IG bit: Individual address (unicast)       .... 0.  ... = LG bit: Globally unique address (factory default)   Source: IntelCor_87:cf:96 (00:21:6a:87:cf:96)     Address: IntelCor_87:cf:96 (00:21:6a:87:cf:96)       .... 0... = IG bit: Individual address (unicast)       .... 0.  ... = LG bit: Globally unique address (factory default)   Type: IP (0x0800) Internet Protocol, Src: 172.25.105.40 (172.25.105.40), Dst: 172.25.105.43 (172.25.105.43)   Version: 4   Header length: 20 bytes   Differentiated Services Field: 0x60 (DSCP 0x18: Class Selector 3; ECN: 0x00)     0110 00.. = Differentiated Services Codepoint: Class Selector 3 (0x18)     .... 0.  ... = ECN-Capable Transport (ECT): 0     .... 0... = ECN-CE: 0           </pre>	

```

Total Length: 546
Identification: 0xca73 (51827)
Flags: 0x00
  0.. = Reserved bit: Not Set
  .0. = Don't fragment: Not Set
  ..0 = More fragments: Not Set
Fragment offset: 0
Time to live: 64
Protocol: UDP (0x11)
Header checksum: 0x8371 [correct]
  [Good: True]
  [Bad : False]
Source: 172.25.105.40 (172.25.105.40)
Destination: 172.25.105.43 (172.25.105.43)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Source port: 5060 (5060)
Destination port: 5060 (5060)
Length: 526
Checksum: 0x57f4 [validation disabled]
  [Good Checksum: False]
  [Bad Checksum: False]
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Status-Code: 200
  [Resent Packet: False]
  [Request Frame: 1]
  [Response Time (ms): 0]
Message Header
  Via: SIP/2.0/UDP 127.0.1.1:5060;branch=z9hG4bK-
1453809699;received=172.25.105.43;rport=5060
    Transport: UDP
    Sent-by Address: 127.0.1.1
    Sent-by port: 5060
    Branch: z9hG4bK-1453809699
    Received: 172.25.105.43
    RPort: 5060
    From: "sipvicious"<sip:100@1.1.1.1>;
tag=6163313936393238313363340131323031353530343335
    SIP Display info: "sipvicious"
    SIP from address: sip:100@1.1.1.1
    SIP from address User Part: 100
    SIP from address Host Part: 1.1.1.1
    SIP tag: 6163313936393238313363340131323031353530343335
  To: "sipvicious"<sip:100@1.1.1.1>;tag=as18cdb0c9
    SIP Display info: "sipvicious"
    SIP to address: sip:100@1.1.1.1
    SIP to address User Part: 100
    SIP to address Host Part: 1.1.1.1
    SIP tag: as18cdb0c9
  Call-ID: 61127078793469957194131
  CSeq: 1 OPTIONS
    Sequence Number: 1
    Method: OPTIONS
  User-Agent: Asterisk PBX 1.6.0.10-FONCORE-r40
  Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
  Supported: replaces, timer

```

```

Contact: <sip:172.25.105.40>
  Contact Binding: <sip:172.25.105.40>
    URI: <sip:172.25.105.40>
      SIP contact address: sip:172.25.105.40
Accept: application/sdp
Content-Length: 0

```

One information that is particularly useful for the attacker is the User-Agent message header field of the response (highlighted in orange)

Given this information, the attacker now knows that:

- He is facing an Asterisk PBX. (<http://www.asterisk.org/>)
- the "FONCORE" mention in the UA string could indicate that this is a Tribox distribution (<http://www.tribox.org/>)

Tribox systems comes with an easy to use web-based GUI called FreePBX

(<http://www.freepbx.org/>), using it one can controls and manages an entire asterisk-based PBX.

With these clues in hands, our attacker tried to connect to the box with HTTP. (starting at packet #3)

Quickly, he tried to access the "Admin" pages via the /maint URI (packet #6). But, the access to these pages is protected by a login/password pair.

Our attacker guessed that he was facing a poorly secured PBX and tried the default administrative credentials to access /maint (packet #60).

```

franck@ODIN:~/Analysis/Sources/Honeynet/Challenge 4$ tshark -r
Forensic_challenge_4.pcap -VR "frame.number==60"

```

```
==== OUTPUT CUT====
```

```

[Number of bytes in flight: 519]
Hypertext Transfer Protocol
  GET /maint HTTP/1.1\r\n
    [Expert Info (Chat/Sequence): GET /maint HTTP/1.1\r\n]
      [Message: GET /maint HTTP/1.1\r\n]
        [Severity level: Chat]
          [Group: Sequence]
            Request Method: GET
              Request URI: /maint
                Request Version: HTTP/1.1
                  Host: 172.25.105.40\r\n
                    User-Agent: Mozilla/5.0 (X11; U; Linux i686; en-US; rv:1.9.1.9) Gecko/20100401
Ubuntu/9.10 (karmic) Firefox/3.5.9\r\n
                      Accept: text/html,application/xhtml+xml,application/xml;q=0.9,*/*;q=0.8\r\n
                        Accept-Language: en-us,en;q=0.5\r\n
                          Accept-Encoding: gzip,deflate\r\n
                            Accept-Charset: ISO-8859-1,utf-8;q=0.7,*;q=0.7\r\n
                              Keep-Alive: 300\r\n
                                Connection: keep-alive\r\n
                                  Referrer: http://172.25.105.40/user/\r\n
                                    Cookie: lng=en; PHPSESSID=gl8n9gi8r09lkelvb0hpf97na1\r\n
                                      Authorization: Basic bWFpbnQ6cGFzc3dvcmQ=\r\n
                                        Credentials: maint:password
                                          \r\n

```

Basic Authentication was used and the decoding of the base64 encoded string:

**bWFpbnQ6cGFzc3dvcmQ=**

reveals that the default login/password for a tribox system were tried by the attacker.

We can see in packet #62 that he was right and was given access to the system. At this point, the attacker had total control over the victim's PBX system.

Well, this kind of “attack” could have been prevented by:

- At least, changing the user's “maint” default password !
- Filtering HTTP access to the box.

Section 2/ Question 4. What information was gained by the attacker ?

Possible Points:  
2pts

Tools Used: tshark, httpdumper

Answer

At this point of the analysis, the attacker knows:

- The kind of SIP PBX => Asterisk
- The distribution: Tribox family.

Now, it's time to analyze the entire HTTP conversations to evaluate what informations were gained by the attacker.

To do this analysis, I've use one of my own tool called httpdumper (<http://malphx.free.fr/dotclear/public/nfpc3/httpdumper>). It's a simple ruby that gives some informations on HTTP flows. Httpdumper has a specific option to list all the requested-uri:

```
franc@ODIN:~/Analysis/Sources/HoneyNet/Challenge 4$ ruby httpdumper -r Forensic_challenge_4.pcap -s uri
Reading file Forensic_challenge_4.pcap
Parsing packets...
4447 packets read in 9.165 sec.

-----
Listing URI requested ALL clients
-----
Requested to
-----
[conv: 68] [flow: 2]
[conv: 68] [flow: 3]
[conv: 72] [flow: 1]
[conv: 72] [flow: 2]
-----
Requested to 172.25.105.40
-----
[conv: 0] [flow: 0] /maint
[conv: 1] [flow: 0] /
[conv: 2] [flow: 0] /user/
[conv: 3] [flow: 0] /maint
[conv: 4] [flow: 0] /maint
[conv: 5] [flow: 0] /maint/
[conv: 6] [flow: 0] /js/scriptaculous.js
[conv: 7] [flow: 0] /maint/js/submitSignup.js
[conv: 8] [flow: 0] /maint/js/iframeSizing.js
[conv: 9] [flow: 0] /maint/js/iframeSizing_freepbx.js
[conv: 10] [flow: 0] /maint/js/main.js
[conv: 11] [flow: 0] /maint/skin/default/css/index_tpl.css
[conv: 12] [flow: 0] /maint/skin/default/cssJavascriptWindows/default.css
[conv: 13] [flow: 0] /maint/skin/default/cssJavascriptWindows/alert.css
[conv: 14] [flow: 0] /maint/js/DHTMLAPI.js
[conv: 15] [flow: 0] /maint/js/javascriptsWindows/window.js
```

```
[conv: 16] [flow: 0] /maint/js/javascrptWindows/effects.js
[conv: 17] [flow: 0] /maint/js/javascrptWindows/window_effects.js
[conv: 18] [flow: 0] /maint/js/javascrptWindows/debug.js
[conv: 19] [flow: 0] /maint/js/javascrptWindows/popUps.js
[conv: 20] [flow: 0] /maint/js/chromejs/chrome.js
[conv: 21] [flow: 0] /maint/js/packages.js
[conv: 22] [flow: 0] /maint/js/initAnime.js
[conv: 23] [flow: 0] /maint/includes/xajax_js/xajax.js
[conv: 24] [flow: 0] /maint/skin/default/css/style.css
[conv: 25] [flow: 0] /maint/skin/default/css/chrometheme/chromestyle.css
[conv: 26] [flow: 0] /maint/skin/default/css/header.css
[conv: 27] [flow: 0] /maint/skin/default/css/footer.css
[conv: 28] [flow: 0] /maint/skin/default/cssJavascriptWindows/mac_os_x.css
[conv: 29] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube.css
[conv: 30] [flow: 0] /js/prototype.js
[conv: 31] [flow: 0] /js/builder.js
[conv: 32] [flow: 0] /js/effects.js
[conv: 33] [flow: 0] /js/slider.js
[conv: 34] [flow: 0] /js/dragdrop.js
[conv: 35] [flow: 0] /js/controls.js
[conv: 36] [flow: 0] /js/sound.js
[conv: 37] [flow: 0] /maint/skin/default/trixbox_logo.gif
[conv: 38] [flow: 0] /maint/
[conv: 39] [flow: 0] /maint/skin/default/arrow_top.gif
[conv: 40] [flow: 0] /maint/skin/default/emailIcon.gif
[conv: 41] [flow: 0] /maint/skin/default/menu_bar.gif
[conv: 42] [flow: 0] /maint/modules/home/index.php?lang=english
[conv: 43] [flow: 0] /maint/skin/default/barS.jpg
[conv: 44] [flow: 0] /maint/skin/default/loading_image.gif
[conv: 45] [flow: 0] /maint/modules/home/templates/classic/classic.css
[conv: 46] [flow: 0] /maint/modules/home/templates/classic/images/bar_right.gif
[conv: 47] [flow: 0] /maint/modules/home/templates/classic/images/bar_left.gif
[conv: 48] [flow: 0] /maint/modules/home/templates/classic/images/bar_middle.gif
[conv: 49] [flow: 0] /maint/modules/home/templates/classic/images/barS.jpg
[conv: 50] [flow: 0] /maint/modules/registrationTool/index.php
[conv: 51] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/top-middle.gif
[conv: 52] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/bottom-middle.gif
[conv: 53] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/frame-right.gif
[conv: 54] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/frame-left.gif
[conv: 55] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/button-close-focus.gif
[conv: 56] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/button-max-focus.gif
[conv: 57] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/button-min-focus.gif
[conv: 58] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/left-top.gif
[conv: 59] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/right-top.gif
[conv: 60] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/bottom-left-c.gif
[conv: 61] [flow: 0] /maint/modules/registrationTool/js/tb_reg.js
[conv: 62] [flow: 0] /maint/skin/default/cssJavascriptWindows/alphacube/bottom-right-c.gif
[conv: 63] [flow: 0] /maint/modules/registrationTool/images/registration.png
[conv: 64] [flow: 0] /maint/index.php?freepbx
[conv: 65] [flow: 0] /maint/index.php?freepbx
[conv: 66] [flow: 0] /admin
[conv: 67] [flow: 0] /admin/
[conv: 68] [flow: 0] /admin/config.php
[conv: 69] [flow: 0] /admin/common/mainstyle.css
[conv: 70] [flow: 0] /admin/config.php?handler=file&module=dashboard&file=dashboard.css
[conv: 71] [flow: 0] /admin/common/script.js.php
[conv: 72] [flow: 0] /admin/common/libfreepbx.javascripts.js
[conv: 73] [flow: 0] /admin/common/stylesheet_custom.css
[conv: 74] [flow: 0] /admin/images/
[conv: 75] [flow: 0] /admin/images/logo.png
[conv: 76] [flow: 0] /admin/images/tab.png
[conv: 77] [flow: 0] /admin/images/shadow-side-background.png
[conv: 78] [flow: 0] /admin/images/notify_update.png
[conv: 79] [flow: 0] /admin/images/notify_delete.png
[conv: 80] [flow: 0] /admin/images/notify_error.png
[conv: 81] [flow: 0] /admin/images/notify_notice.png
[conv: 82] [flow: 0] /admin/images/cancel.png
[conv: 83] [flow: 0] /admin/images/modules-hover1.png
[conv: 84] [flow: 0] /admin/config.php?
```

```

type=tool&display=index&quietmode=1&info=stats&restrictmods=core/dashboard
[conv: 85] [flow: 0] /admin/config.php?
type=tool&display=index&quietmode=1&info=stats&restrictmods=core/dashboard
[conv: 86] [flow: 0] /maint/index.php?configEdit
[conv: 87] [flow: 0] /maint/index.php?configEdit
[conv: 88] [flow: 0] /maint/modules/configedit/phpconfig.php
[conv: 89] [flow: 0] /maint/modules/configedit/images/spacer.gif
[conv: 90] [flow: 0] /maint/modules/configedit/images/bar$.jpg
[conv: 91] [flow: 0] /maint/modules/configedit/phpconfig.php?
file=sip_custom.conf&section=sip_custom.conf

```

The listing above list the request-uri in the order where they have been requested. We could analyze the progression of the attacker in the FreePBX GUI. One interesting thing is the last requested URI (conv 91).

This URI is: /maint/modules/configedit/phpconfig.php?

file=sip\_custom.conf&section=sip\_custom.conf

By requesting this URI, the attacker is trying to access the informations stored in this configuration file.

sip\_custom.conf is a configuration file that can be used on asterisk system to define SIP extensions.

Let's try to view what information the attacker has gained by dumping the PBX response in HTTP conversation #91:

```

franck@ODIN:~/Analysis/Sources/HoneyNet/Challenge 4$ ruby httpdumper -r Forensic_challenge_4.pcap -c 91
Reading file Forensic_challenge_4.pcap
Parsing packets...
4447 packets read in 9.504 sec.

FLOWS TABLE
-----
| Flow Index | Hosts | HTTP message type | HTTP Request or Content type | HTTP Content Length |
-----
| 0 | 172.25.105.43:57177 -> 172.25.105.40:80 | REQUEST | /maint/modules/configedit/phpconfig.php?file=sip_custom.conf&section=sip_custom.conf | 0 |
| 1 | 172.25.105.40:80 -> 172.25.105.43:57177 | RESPONSE | text/html | 7000 |
-----

```

A 7 KB HTML file was receive. Obviously, this file contains all the HTML stuff to render the page in the browser, but more interesting, we can also view, what information the attacker has accessed: (see next page)



## Edit: sip\_custom.conf

```
[555]
type=friend
username=555
secret=1234
host=dynamic
extension=from-trunk
context=from-trunk

[556]
type=friend
username=555
secret=1234
host=dynamic
extension=from-trunk
context=from-trunk
```

Update

The attacker has gained some valuable information on 2 SIP extensions: 555 and 556  
The most valuable informations being:

- Extensions numbers : 555 and 556
- Username and password used to register extensions : 555/1234

After having taken control over the SIP PBX by guessing the administrator credentials, our attacker has now (at least) all the informations he needs to register and use maliciously the 2 extensions listed above. But he also has total control on the Asterisk system and can potentially do anything he wants to.

Section 2/ Question 5a. The PCAP includes a (not so) hidden bonus! [hint1: You can't read it in the pcap, hint2: It's a city with an active honeynet chapter]  
 a) Describe it, and explain how you found it.

Possible Points: 10pts

Tools Used: wireshark

Answer

Quick answer: "the secret password is: MEXICO"

Following the HTTP flows between the attacker and the Asterisk PBX. A new host 172.25.105.3 successfully registered the 555 extension. (packet #1294 - #1297). (It can be the attacker again, registering the extension from a new host, an accomplice of the attacker or a legitimate user.)

No.	Time	Source	Destination	Protocol	Info
1294	2010-05-01 20:16:08.031326	172.25.105.3	172.25.105.40	SIP	Request: REGISTER sip:172.25.105.40
1295	2010-05-01 20:16:08.031642	172.25.105.40	172.25.105.3	SIP	Status: 401 Unauthorized (0 bindings)
1296	2010-05-01 20:16:08.035292	172.25.105.3	172.25.105.40	SIP	Request: REGISTER sip:172.25.105.40
1297	2010-05-01 20:16:08.039213	172.25.105.40	172.25.105.3	SIP	Status: 200 OK (1 bindings)
1298	2010-05-01 20:16:08.042676	172.25.105.3	172.25.105.40	SIP	Request: SUBSCRIBE sip:555@172.25.105.40
1299	2010-05-01 20:16:08.042985	172.25.105.40	172.25.105.3	SIP	Status: 401 Unauthorized
1300	2010-05-01 20:16:08.045939	172.25.105.3	172.25.105.40	SIP	Request: SUBSCRIBE sip:555@172.25.105.40
1301	2010-05-01 20:16:08.046208	172.25.105.40	172.25.105.3	SIP	Status: 404 Not found (no mailbox)
1302	2010-05-01 20:16:22.261857	172.25.105.3	172.25.105.40	SIP/SDP	Request: INVITE sip:1000@172.25.105.40, with session description
1303	2010-05-01 20:16:22.262702	172.25.105.40	172.25.105.3	SIP	Status: 401 Unauthorized
1304	2010-05-01 20:16:22.265247	172.25.105.3	172.25.105.40	SIP	Request: ACK sip:1000@172.25.105.40
1305	2010-05-01 20:16:22.266798	172.25.105.3	172.25.105.40	SIP/SDP	Request: INVITE sip:1000@172.25.105.40, with session description
1306	2010-05-01 20:16:22.267373	172.25.105.40	172.25.105.3	SIP	Status: 100 Trying
1307	2010-05-01 20:16:22.273915	172.25.105.40	172.25.105.3	SIP/SDP	Status: 200 OK, with session description
1309	2010-05-01 20:16:22.297365	172.25.105.3	172.25.105.40	SIP	Request: ACK sip:1000@172.25.105.40
4444	2010-05-01 20:17:01.052671	172.25.105.3	172.25.105.40	SIP	Request: BYE sip:1000@172.25.105.40
4446	2010-05-01 20:17:01.053384	172.25.105.40	172.25.105.3	SIP	Status: 200 OK

Then 172.27.105.3 tries to establish a call to [1000@172.25.105.40](tel:1000@172.25.105.40) via a SIP INVITE message (packet #1302 and #1305)

The call confirmation is done by the IPBX, first by sending a 100 Trying response code (packet #1306) and then a 200 OK response with SDP informations in packet #1307

Informations exchanged in SDP (Session Description Protocol) in packet #1305 and #1307 define all the parameters of the audio communication that will start between 555@172.25.105.40 and [1000@172.25.105.40](tel:1000@172.25.105.40).

Using the VoIP Call option of the Wireshark's Telephony menu, it is possible to list all the VoIP call contained in a pcap file.

No.	Time	Source	Destination	Protocol	Info
1294	2010-05-01 20:16:08.031326	172.25.105.3	172.25.105.4	SIP	Request: REGISTER sip:172.25.105.4
1295	2010-05-01 20:16:08.031642	172.25.105.4	172.25.105.3	SIP	Status: 401 Unauthorized (0 bindings)
1296	2010-05-01 20:16:08.035292	172.25.105.3	172.25.105.4	SIP	Request: REGISTER sip:172.25.105.4
1297	2010-05-01 20:16:08.039213	172.25.105.4	172.25.105.3	SIP	Status: 200 OK (1 bindings)
1298	2010-05-01 20:16:08.042676	172.25.105.3	172.25.105.4	SIP	Request: SUBSCRIBE sip:555@172.25.105.4
1299	2010-05-01 20:16:08.042905	172.25.105.4	172.25.105.3	SIP	Status: 401 Unauthorized
1300	2010-05-01 20:16:08.045939	172.25.105.3	172.25.105.4	SIP	Request: SUBSCRIBE sip:555@172.25.105.4
1301	2010-05-01 20:16:08.046208	172.25.105.4	172.25.105.3	SIP	Status: 404 Not found (no mailbox)
1302	2010-05-01 20:16:22.261857	172.25.105.3	172.25.105.4	SIP/SDP	Request: INVITE sip:1000@172.25.105.4
1303	2010-05-01 20:16:22.262702	172.25.105.4	172.25.105.3	SIP	Status: 401 Unauthorized
1304	2010-05-01 20:16:22.265247	172.25.105.3	172.25.105.4	SIP	Request: ACK sip:1000@172.25.105.4
1305	2010-05-01 20:16:22.266798	172.25.105.3	172.25.105.4	SIP/SDP	Request: INVITE sip:1000@172.25.105.4
1306	2010-05-01 20:16:22.267374	172.25.105.4	172.25.105.3	SIP	Status: 100 Trying
1307	2010-05-01 20:16:22.273515	172.25.105.3	172.25.105.4	SIP/SDP	Status: 200 OK, with session description
1308	2010-05-01 20:16:22.288646	172.25.105.3	172.25.105.4	RTCP	Receiver Report Source description
1309	2010-05-01 20:16:22.297365	172.25.105.3	172.25.105.4	SIP	Request: ACK sip:1000@172.25.105.4
1310	2010-05-01 20:16:22.364054	172.25.105.3	172.25.105.4	RTP	PT=ITU-T G.711 PCMU, SSRC=0xA25...

Detected 1 VoIP Call. Selected 0 Calls.

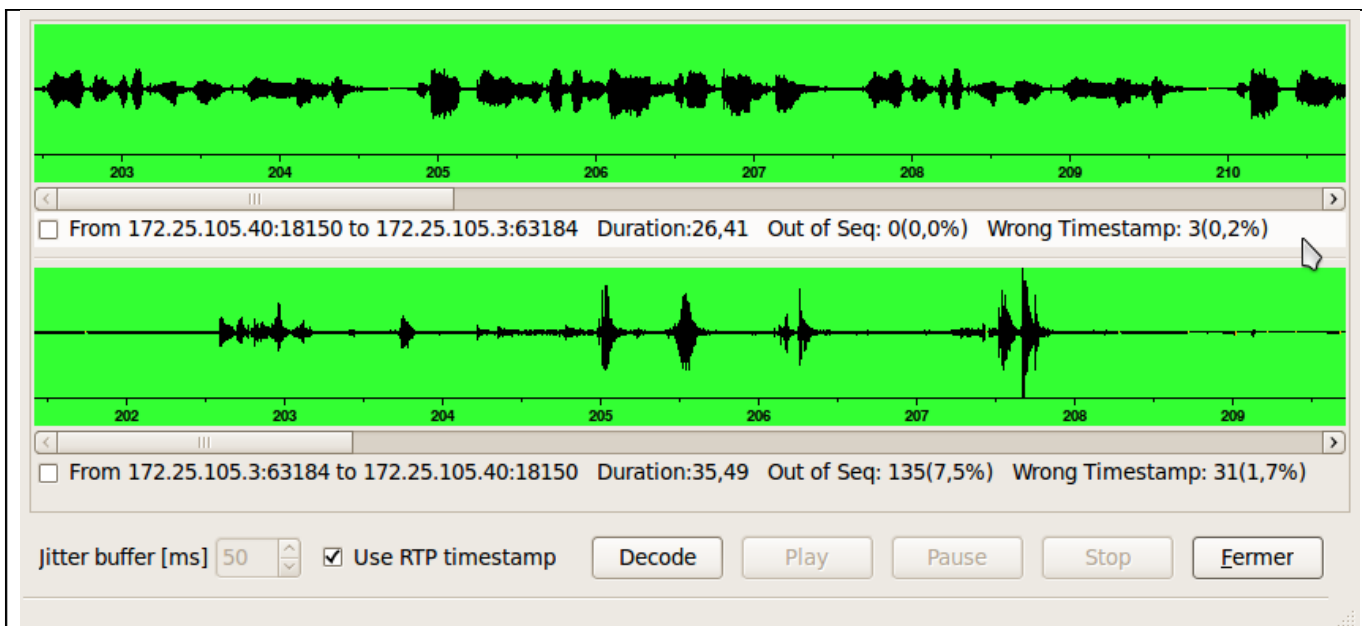
Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
201.313	240.105	172.25.105.3	sip:555@172.25.105.40	sip:1000@172.25.105.4	SIP	9	COMPLETED	

Total: Calls: 1 Start packets: 0 Completed calls: 1 Rejected calls: 1

Buttons: Prepare Filter, Graph, Player, Select All, Fermer

Wireshark offer a simple player capable of reading unencrypted RTP audio streams. This player can be launched by selecting a VoIP call and pressing the “Player” button. The next window is used to adjust the jitter buffer manually or to use RTP timestamps stored in RTP packets to decode the RTP streams. Select “Use RTP timestamp” and press “Decode”

Jitter buffer [ms]   Use RTP timestamp



The RTP player's window appears and now to listen to the audio stream, just select one or both of the audio streams and press "Play".

The audio stream reveals that the user is trying to access an audio conference room (1000) and after having typed 2 bad pin numbers he finally accessed the conference. The user was the only person in this conference.

After some times in the conference room, the user said:

*"Congratulations, you're listening to an unencrypted VoIP call. The secret password is Mexico, so write it down and submit your challenge. Good job, thank you!"*

Section 2/ Question 5b. If VOIP packets between the two calling parties traverse an untrusted network (eg the wireless/internet) and a similar PCAP was captured by a malicious party, would you think this a security problem? why?	Possible Points: 3pts
--	--------------------------

Tools Used:

Answer

Yes it is.

For obvious reason, if the audio stream is not encrypted , an attacker can eavesdrop on the conversation and some business-critical informations can be recorded or stolen. DTMF tones = passwords/pin numbers can also be stolen (heard).

The signaling protocol messages (here SIP) also travel unencrypted (in our case) and can give to an attacker some valuable informations on the calling parties, like:

- Extension numbers
- Call-ID values
- Cseq values
- Authentication digest
  - That can be brute-forced off-line with tools like SIPcrack (<http://www.darknet.org.uk/2008/08/sipcrack-sip-login-dumper-hashpassword-cracker/>)
  - That can be used in attacks which need some packets to be replayed, like the "SIP

unregister attack" (<http://www.idc.ac.il/publications/files/238.pdf>)

So, If VoIP packets need to traverse an untrusted network, security must be used for the signaling protocol (SIP over TLS / Secure SIP) and for the media stream (SRTP, ZRTP...)

Section 2/ Question 5c. Wireshark has an option "Use RTP timestamp". What is the function of this option?	Possible Points: 2pts
Tools Used:	
<p>Answer</p> <p>This is an option of the Wireshark's RTP Player. It is used to decode and play an RTP stream based on RTP timing stored in RTP packet instead of on packet arrival time. Each RTP packets includes a timestamp which define the sampling instant of the first octet of the data packet. Quote from RFC 3550 (RTP): <i>The timestamp reflects the sampling instant of the first octet in the RTP data packet.</i></p> <p>This feature is useful when the original IP/RTP packets have been encapsulated or tunneled and original timing is lost. In this case Wireshark will use RTP timestamp values to order and decode the audio stream. When using this feature Wireshark cannot simulate the jitter buffer and so, this option is grayed out.</p> <p>One drawback of this feature is that the RTP player doesn't render the audio as the calling parties have heard it.</p>	

Section 2/ Question 6. What technologies or protocols can be used to protect confidentiality of RTP traffic as it traverses untrusted networks.	Possible Points: 3pts
Tools Used:	
<p>Answer</p> <p>Multiples solutions exists to protect and secure RTP exchanges. They mostly rely on message authentication, encryption. Here are some examples of such technologies:</p> <ul style="list-style-type: none"> <li>• SRTP (RFC 3711) can be used to protect RTP traffic. It's an RTP profile which provides confidentiality (through encryption), message authentication and replay protection to the RTP and RTCP traffic.</li> <li>• ZRTP (<a href="http://tools.ietf.org/html/draft-zimmermann-avt-zrtp-21">http://tools.ietf.org/html/draft-zimmermann-avt-zrtp-21</a>) which provides a key-agreement protocol to exchange key informations (using Diffie-Hellman exchange) between calling parties in RTP packets (in-band). Then ZRTP uses SRTP to secure the data stream.</li> </ul>	

- Using a protocol like RTSP which provides a way to multiplex data and control in a single stream (RTSP + RTP data) supported by an unique TCP connection. This connection can then be secured using TLS hence offering the expected confidentiality.
- RTP can also be protected with the security offered by the network layer (by the use of IPsec for example).

A good reference on this subject can be find in this document:  
<http://tools.ietf.org/html/draft-ietf-avt-srtp-not-mandatory-05>

Section 3/ Question 1. What is "RTP injection" and describe how it functions.  
 What conditions are required to allow this?

Possible Points:  
 2pts

Tools Used:

Answer

RTP injection is a kind of attack where the attacker is able to inject or mix RTP packets in an on-going call between two parties. One objective of this attack can be to diffuse "SPIT" (SPams over Internet Telephony) by injecting a pre-recorded audio message in an established VoIP call. This attack targets only the media protocol (RTP) and hence is totally independent of the signaling protocol used to setup the call.

Nevertheless, RTP injection is only possible when some specific conditions are met:

- The targeted RTP stream must be unencrypted.
- The use of UDP protocol as transport protocol for RTP
- The attacker must be able to capture at least one valid RTP packet from the stream. This packet will be used as a template to construct the spoofed RTP packets that will be later injected in the stream.
- From this packet, the attacker has to get critical informations on the stream to successfully inject RTP packets. These informations are:
  - The payload type
    - needed to send correctly encoded audio data.
  - The RTP Sequence number
    - Will be set in the spoofed packets to a higher value than the legitimate RTP packets. This will force the receiver "thinking" they are older than the spoofed ones and hence will be dropped.
  - The RTP timestamps
    - in the same fashion as sequence numbers, timestamp will be set to an higher value than the legitimate packets.
  - Synchronization Source Identifier (SSRC)
    - This value remain the same during all the call. So the attacker has just to set SSRC with the same value as the captured packet's SSRC field.
  - IP ID
    - Again the IP ID will be set to a higher value than the legitimate RTP/IP packets.

If all this conditions are met, the attacker should be able to correctly craft RTP packets and to inject them in the on-going call.

Using a secure media protocol, like SRTP or ZRTP, prevent this kind of attack.

Section 3/ Question 2. Explain how a SIP password digest could be intercepted or stolen. Is this a security issue? why or why not.	Possible Points: 2pts
Tools Used:	
<p>Answer</p> <p>At least two ways can be used to intercept and steal SIP password digest:</p> <ul style="list-style-type: none"> <li>• By Sniffing SIP traffic <ul style="list-style-type: none"> <li>◦ Attacker can take control of a poorly secure switch (password brute-force, exploit, social engineering...) and the configure traffic mirroring</li> <li>◦ Attacker can have previously attacked a poorly secure Wireless LAN and then can sniff traffic "over the air".</li> <li>◦ Intrusive: attacker can insert a hub or a Pc with two NIC cards on the traffic path ...</li> </ul> </li> <li>• By Redirecting SIP traffic flowing between a client and a server to the attacker using Man-in-the-Middle (MitM) attack. <ul style="list-style-type: none"> <li>◦ On today's switched networks, an attacker cannot easily eavesdrops on the traffic not destined to him. So, he has to use traffic diversion by launching a MitM attack against a SIP client and a SIP proxy for example. <ul style="list-style-type: none"> <li>▪ DNS entry modification to divert traffic to the attacker host</li> <li>▪ ARP spoofing (gratuitous ARP) directed to a client and associating the IP address of the SIP server to the attacker's MAC address</li> <li>▪ Flooding switches with lot of unknown MAC address to exhaust CAM table and force the switch to broadcast all the packets over all the ports...</li> </ul> </li> <li>◦ If attack is successful, the attacker will be able to eavesdrop on SIP traffic (and surely other kind of network traffic flowing between the two parties) and steal password digest.</li> </ul> </li> </ul> <p>In the case where SIP traffic travels unencrypted on the network, these two ways can give to an attacker the possibility of stealing SIP password digest. Then, he will be able to use them in some "replay attacks" or try to brute-force them with specialized offline cracking tools (e.g sipcrack). So, this can be considered as a security issue.</p> <p>However, use of the secured version of SIP, SIP over TLS, may thwart these attacks.</p>	

Section 3/ Question 3. Is DDoS a threat to VOIP systems? Are there any general functional requirements of telephony systems that would be impaired by a DDoS?	Possible Points: 2pts
Tools Used:	
Answer	
<p>DDoS stands for Distributed Deny of Service.</p> <p>It is an attempt to make a particular service, offered by a server, unavailable to its legitimate users by denying it or bringing the server that offers the service down (Crash, reboot-loop...).</p> <p>“Distributed” involves the use of a medium to large amount of previously compromised computers (e.g Zombies and BotNets) to launch a synchronized attack against an unique target. The primary objective is to exhaust server resources, thus making it unable to process legitimate user's requests.</p> <p>VoIP systems like any other “servers” need resources to make their jobs (Calls handling for example), these resources are CPU, memory, network bandwidth...</p> <p>This is making them vulnerable to DDoS attacks.</p> <p>Some examples of such attacks are:</p> <ul style="list-style-type: none"><li>• Flooding a SIP proxy with SIP REGISTER or SIP INVITE messages, making him unable to process legitimate calls or user's requests.</li><li>• Exhausting resources by sending large amount of SIP REGISTER messages to extensions that need authentication. (Database lookup)</li></ul>	