Problems on Voice over IP (VoIP)
Problem 1

The following trace refers to the registration phase of a SIP user. Answer the following questions:

1. What is the IP address of the SIP client?
2. What is the IP address of the SIP proxy?
3. Why the first registration attempt does fail?
4. Provide a brief explanation of the whole registration process

<table>
<thead>
<tr>
<th>N.</th>
<th>Network</th>
<th>Transport</th>
<th>Application</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>IP: 130.192.225.36 =&gt; 130.192.225.79 (Len 60)</td>
<td>UDP: port 4136 =&gt; 53</td>
<td>DNS Query</td>
</tr>
<tr>
<td>2</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.36 (Len 328)</td>
<td>UDP: port 53 =&gt; 4136</td>
<td>DNS Response</td>
</tr>
<tr>
<td>3</td>
<td>IP: 130.192.225.36 =&gt; 130.192.225.79 (Len 601)</td>
<td>UDP: port 63772 =&gt; 5000</td>
<td>SIP: REGISTER sip:ipv6полито.it SIP/2.0</td>
</tr>
<tr>
<td>4</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.36 (Len 728)</td>
<td>UDP: port 5060 =&gt; 63772</td>
<td>SIP: SIP/2.0 401 Unauthorized</td>
</tr>
<tr>
<td>5</td>
<td>IP: 130.192.225.36 =&gt; 130.192.225.79 (Len 800)</td>
<td>UDP: port 63772 =&gt; 6060</td>
<td>SIP: REGISTER sip:ipv6полито.it SIP/2.0</td>
</tr>
<tr>
<td>6</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.36 (Len 706)</td>
<td>UDP: port 5060 =&gt; 63772</td>
<td>SIP: SIP/2.0 200 OK</td>
</tr>
</tbody>
</table>
Problem 1 - answers

1. The IP address of the UA SIP is 130.192.225.36
2. The IP address of the SIP proxy is 130.192.225.79
3. The first registration attempt fails because SIP UA did not include credentials for authentication in the REGISTER message
4. The registration procedure is required for
   1. Authenticating a user that tries to access a SIP domain
   2. Associating the SIP URI with the SIP UA (host) where the user is connected
      1. In this way, the user can be reached using his SIP URI
      2. If the user moves to a different IP address, he should re-register with his domain SIP server

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<td>130.192.225.36 ⇒ 130.192.225.36 (Len: 328)</td>
<td>UDP: port 53 ⇒ 4136</td>
<td>DNS Response</td>
</tr>
<tr>
<td>4</td>
<td>130.192.225.79 ⇒ 130.192.225.36 (Len: 726)</td>
<td>UDP: port 5060 ⇒ 63772</td>
<td>SIP: SP/2.0 401 Unauthorized</td>
</tr>
<tr>
<td>5</td>
<td>130.192.225.36 ⇒ 130.192.225.79 (Len: 800)</td>
<td>UDP: port 63772 ⇒ 5060</td>
<td>SIP: REGISTER sip:ip:6.polflo.it SIP/2.0</td>
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<tr>
<td>6</td>
<td>130.192.225.79 ⇒ 130.192.225.36 (Len: 706)</td>
<td>UDP: port 5060 ⇒ 63772</td>
<td>SIP: SP/2.0 200 OK</td>
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Problem 2

Explain the meaning and the role of the main SIP headers of the following message:

```
----- Packet Details -----
General information
   Ethernet 802.3
   IPv4 (Internet Protocol version 4)
   UDP (User Datagram protocol)
   Session Initiation Protocol
Command = REGISTER sip:ipv6.polito.it SIP/2.0
Header Field = Via SIP/2.0/UDP 130.152.225.36:87772;branch=z9hG4bK_d87543-7a65e658044a0865-1-1587543-xport
Header Field = Max-Forwards: 70
Header Field = Contact: <sip:test_user@130.152.225.36:87772;instance=025c7a3c7a9da0e2>
Header Field = To: "test_user4"<sip:test_user@ipv6.polito.it>
Header Field = From: "test_user4"<sip:test_user@ipv6.polito.it>;tag=8e77c213
Header Field = Call-ID: HjksMDQzrVZWxM2RmMyMwMyYzczM0ZmZTU5OGE3YTIQ.
Header Field = CSeq 2 REGISTER
Header Field = Expires: 3600
Header Field = Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Header Field = User-Agent: X-Lite release 1011s stamp 41150
Header Field = Authorization: Digest username="test_user", realm="ipv6.polito.it", nonce="1814e521cbac3d655c1105f17b"
Header Field = Content-Length: 0
```
Problem 2 - solution

The "To" header stores the URI identifying the SIP user.

The "contact" header stores the information about the current position of the user (IP+port number).

Period of validity of the registration

Authentication credentials
Problem 3

Given the following SIP REGISTER message, assuming a correct configuration for the SIP client, what could be the answer to this request?

--- Packet Details ---
- General Information
- Ethernet 802.3
- IPv4 (Internet Protocol version 4)
- UDP (User Datagram protocol)
- Session Initiation Protocol
  - Command: REGISTER sip:ipv6.pollito.it SIP/2.0
  - Header Field: Via SIP/2.0/UDP 130.192.225.36;branch=z9hG4bK-d675f5c-3f770c2ab62b675d-1-d67543, rport
  - Header Field: Max-Forwards: 70
  - Header Field: Contact: <sip:test_user@130.192.225.36;branch=z9hG4bK-d675f5c-3f770c2ab62b675d-1-d67543, rport>
  - Header Field: To: "test_user4"<sip:test_user@ipv6.pollito.it>
  - Header Field: From: "test_user4"<sip:test_user@ipv6.pollito.it>;tag=8e7f213
  - Header Field: Call-ID: NkZmDDQzYWZkM2RIMDIyMAVhMyZcgzMQzMTU5OGExYTQz
  - Header Field: CSeq: 1 REGISTER
  - Header Field: Expires: 3600
  - Header Field: Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
  - Header Field: User-Agent: X-Lite release 1011s stamp 41:59
  - Header Field: Content-Length: 0
Problem 3 - answer

Since most of the SIP server require a user to authenticate itself, the most likely answer is “401 Unauthorized”, because the message does not include credentials for the authentication. No username – IP association will be created and the server response will include a challenge for the authentication.
Problem 4

The following is a trace of an INVITE session, assuming that the caller is in the SIP domain “ipv6.polito.it”, answer the following questions:

1. What is the username of both clients?
2. What is the IP address and port number for both clients?
3. What is the meaning of the “100 Trying” message?
4. What is the meaning of the “180 Ringing” message?
5. Record-routing is enabled in the SIP proxy(ies)?
6. What is the minimum number of SIP proxies that can be traversed by an INVITE message?

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<td>DNS Query</td>
<td>IP: 130.192.225.135 =&gt; 130.192.225.79 (Len 60)</td>
</tr>
<tr>
<td>2</td>
<td>DNS Response</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.135 (Len 328)</td>
</tr>
<tr>
<td>3</td>
<td>SIP: INVITE sip:example@130.192.225.135 SIP/2.0</td>
<td>IP: 130.192.225.135 =&gt; 130.192.225.79 (Len 759)</td>
</tr>
<tr>
<td>4</td>
<td>SIP: SIP/2.0 100 trying -- your call is important to us</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.135 (Len 604)</td>
</tr>
<tr>
<td>5</td>
<td>SIP: SIP/2.0 180 Ringing</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.135 (Len 491)</td>
</tr>
<tr>
<td>6</td>
<td>SIP: SIP/2.0 200 OK</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.135 (Len 811)</td>
</tr>
<tr>
<td>7</td>
<td>SIP: ACK sip:example@130.192.225.79 SIP/2.0</td>
<td>IP: 130.192.225.135 =&gt; 130.192.225.79 (Len 509)</td>
</tr>
<tr>
<td>8</td>
<td>SIP: BYE sip:livio@130.192.225.135:226 SIP/2.0</td>
<td>IP: 130.192.225.79 =&gt; 130.192.225.135 (Len 655)</td>
</tr>
<tr>
<td>9</td>
<td>SIP: SIP/2.0 200 OK</td>
<td>IP: 130.192.225.135 =&gt; 130.192.225.79 (Len 485)</td>
</tr>
</tbody>
</table>
Problem 4: answers (1/3)

1. Caller username: "test_user"; called username: "livio"

2. Caller address: 130.192.225.135:7226
   Called address: 130.192.225.36:63772

   - IP addresses and port numbers can be read from the messages ACK and BYE (respectively)
   - Once the dialog is established, the two UAs can communicate directly, using the addresses and port numbers discovered during the INVITE process
Problem 4: answers (2/3)

3. The message “100 Trying” is sent from the proxy to the caller UA to signal that the request has been received, and it is under process.

4. The message “180 Ringing” is sent from the called UA to the caller one (through the proxies) to confirm the reception of the request and for notifying the caller that the called phone is now ringing.
   - Only when the human user will take the phone "off-hook" the response 200 OK is generated.
5. It is possible to see that the proxy is always involved in all the SIP messages. Hence record routing is enabled.

6. Since both users belong to the same SIP domain, the minimum number of proxies traversed is 1 (it would be 2 if they were in 2 different domains)
Problem 5 (without solution)

The following is a trace of an INVITE session, assuming that the caller is in the SIP domain “ipv6.polito.it”, answer the following questions:

- What is the IP address and port number of both SIP UA?
- Is record routing enabled?

<table>
<thead>
<tr>
<th>No.</th>
<th>Time</th>
<th>Source</th>
<th>Destination</th>
<th>Protocol</th>
<th>Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0.000060</td>
<td>130.192.225.79</td>
<td>130.192.225.133</td>
<td>SIP/SD</td>
<td>Request: INVITE sip:tel:+39010234567893456</td>
</tr>
<tr>
<td>2</td>
<td>0.021484</td>
<td>130.192.225.133</td>
<td>130.192.225.79</td>
<td>SIP</td>
<td>Status: 180 Ringing</td>
</tr>
<tr>
<td>3</td>
<td>2.740769</td>
<td>130.192.225.133</td>
<td>130.192.225.79</td>
<td>SIP/SD</td>
<td>Status: 200 OK, with session description</td>
</tr>
<tr>
<td>4</td>
<td>2.855290</td>
<td>130.192.225.36</td>
<td>130.192.225.133</td>
<td>SIP</td>
<td>Request: ACK sip:+39010234567893456</td>
</tr>
<tr>
<td>5</td>
<td>8.651164</td>
<td>130.192.225.133</td>
<td>130.192.225.36</td>
<td>SIP</td>
<td>Request: BYE sip:test_user@130.192.225.36:78884</td>
</tr>
<tr>
<td>6</td>
<td>8.754173</td>
<td>130.192.225.36</td>
<td>130.192.225.133</td>
<td>SIP</td>
<td>Status: 200 OK</td>
</tr>
</tbody>
</table>
Problem 6

If two SIP users have the following URI correctly registered in their domain:

- alice@ipv6.polito.it
- bob@ipv6.polito.it

1. **Draw a diagram with all the messages exchanged between Alice and Bob, in order to set up a call, including:**
   1. Possible auxiliary messages
   2. Messages sent by the proxy
Problem 6 - answer

The requested diagram is shown below.

- Since the registration has already taken place, both UAs can access the proxy without additional DNS queries, but taking advantage of their DNS caches.
Problem 7

- If two SIP users have the following URI correctly registered in their domain
  - alice@ipv6.polito.it
  - bob@iptel.org

1. **Draw a diagram with all the messages exchanged between Alice and Bob, in order to set up a call, including:**
   1. Possible auxiliary messages
   2. Messages sent by the proxies
Esercizio 7 - soluzione

Domain ipv6.polito.it
alice@ipv6.polito.it

proxy SIP
ipv6.polito.it

DNS server

proxy SIP
Iptel.org

Domain iptel.org
bob@ipv6.polito.it

INVITE
100 Trying

DNS 'NAPTR' query
DNS 'NAPTR' resp
DNS 'SRV' query
DNS 'SRV' resp
DNS 'A' query
DNS 'A' resp

INVITE
180 Ringing

180 Ringing
200 OK

ACK
200 OK

INVITE
180 Ringing

180 Ringing
200 OK