FINAL EXAM

NAME:

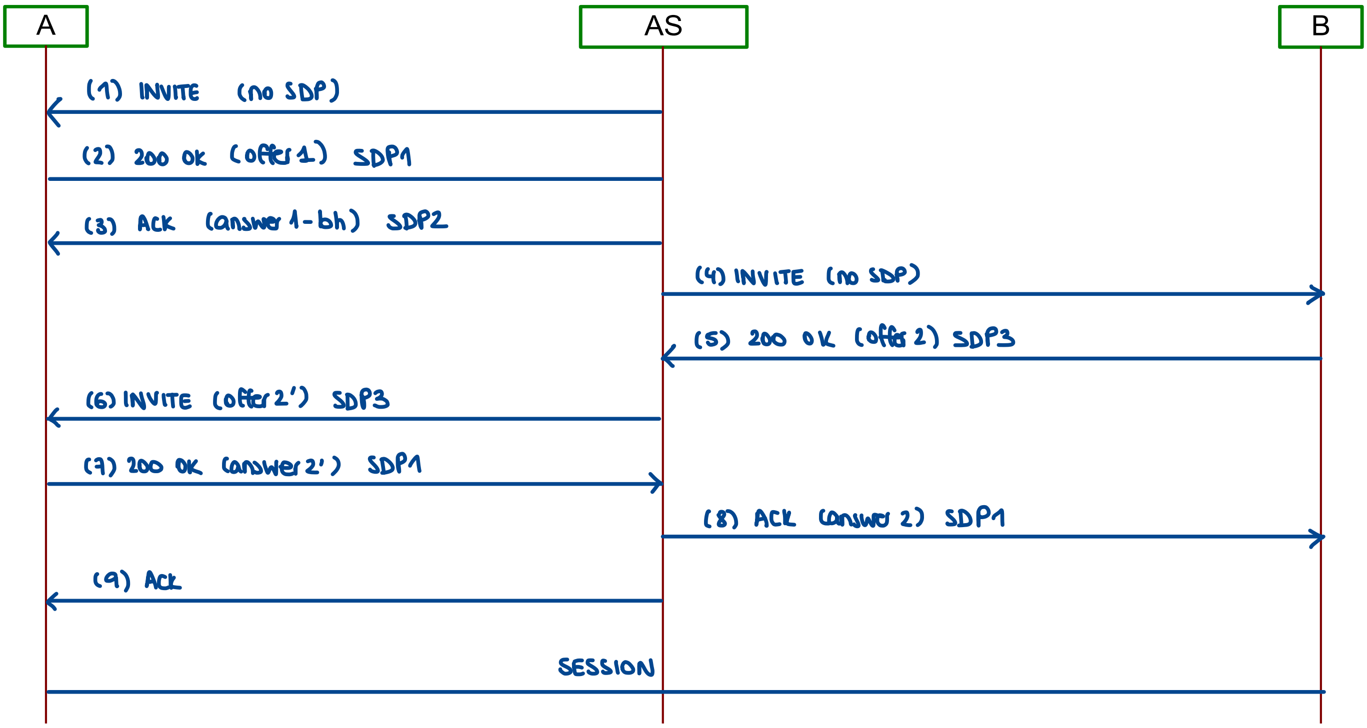
ACTIVITY 1 (3 PTS)

Consider the following multimedia communications scenario:

- There are 2 User Agents (A and B) and an Application Server (AS) that acts as a B2BUA
- A y B represent human users
- The AS manages the web calling service. It is NOT an RTP entity (it does not handle media traffic)
- 100rel is not supported
- Neither A nor B generate early media
- A and B only support PCMA voice (payload type=8)
- A's IP address=1.1.1.1, A's media port=13000
- B's IP address=2.2.2.2, B's media port=14000

It is requested to represent the signaling flow indicating the name of the SIP messages and the SDP content (only c and m line) for the following scenario:

- A requests via web that the AS sets up a call between A and B
- AS uses 3PCC mechanisms in order to establish the call between A and B on behalf of A



I have used flow 3.
Flow 1 would not be recommended in this case because A and B are human and if they not respond immediately, we would have timeout problems.

(1) The AS sends an INVITE with no SDP to A. This is used to alert A that it should generate an offer
(2) A responds to the INVITE and generates an offer (offer 1)
(3) AS sends the answer to offer 1 (answer 1) in the ACK indicating a bh IP address
(4) AS sends an INVITE with no SDP to B. This is used to alert B that it should generate an offer
(5) B responds to the INVITE and generates an offer (offer 2)
(6) Before sending the ACK to B, the AS sends B's offer to A. There's no timeout problem because this INVITE is already within the dialog
(7) A answers to offer 2'
(8) AS sends the ACK to B including A's answer to its offer.
(9) AS sends an ACK to A to finish the transaction initiated in message (6).

Flow 3 sometimes poses an incompatibility problem between A and B. In case they don't support the same media, the AS might need to add or trim m-lines. These SDP modifications are not efficient. However, in this case, since both A and B only support voice, no modifications will be needed.

SDP 1 → c= IN IP4 1.1.1.1
m= audio 13000 RTP / AVP 8

SDP 2 → c= IN IP4 0 0.0 0
m= audio 13000 RTP / AVP 8

SDP 3 → c= IN IP4 2.2 2 2
m= audio 14000 RTP / AVP 8

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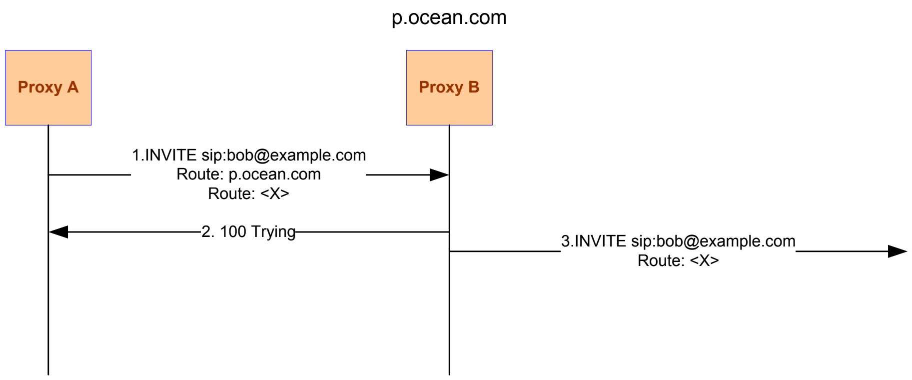
ACTIVITY 2 (2 PTS)

The following flow shows a scenario that involves several SIP proxies:



INVITE in step 1 contains two Route headers. There is a DNS , reachable by all the nodes, that contains the following entries:

Name	AddresType	RR Type	Order	Pref.	Flags	Service	Reg Ex	Replace
example.com	IN	NAPTR	20	50	“s”	“SIP+D2T”		_sip._tcp.example.com
example.com	IN	NAPTR	30	50	“s”	“SIP+D2U”		_sip._udp.example.com

Name	Address Type	RR Type	Priority	Weight	Port	Target
_sip_udp.example.com	IN	SRV	10	1	5080	proxy4.example.com
_sip_udp.example.com	IN	SRV	20	1	5080	proxy5.example.com
_sip_tcp.example.com	IN	SRV	10	1	5090	proxy6.example.com
_sip_tcp.example.com	IN	SRV	20	1	5090	proxy7.example.com
example.com	IN	SRV	10	1	5090	example.com
example.com	IN	SRV	20	1	5090	backup.example.com

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Name	Addres Type	RR Type	Address
proxy4.example.com	IN	A	192.168.74.4
proxy5.example.com	IN	A	192.168.74.5
proxy6.example.com	IN	A	192.168.74.6
proxy7.example.com	IN	A	192.168.74.7
example.com	IN	A	192.168.74.8
backup.example.com	IN	A	192.168.74.9

It is requested to indicate the value of the IP address, port and protocol used to generate the INVITE message in step 3, based on the value of the Route header:

Route header	IP	puerto	protocolo
sip:192.168.74.2;lr	192.168.74.2	5060	UDP
sip:192.168.74.2:5085;lr	192.168.74.2	5085	UDP
sip:example.com;lr	192.168.74.6	5090	TCP
sip:example.com;transport=udp;lr	192 168.74.4	5080	UDP
sip:example.com:5070;lr	192.168.74.8	5070	UDP

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ACTIVITY 3 (2 PTS)

Please, answer the following questions related to RTP:

(1) Let us asume that a RTP sender splits a coded video frame in 3 packets before sending it. If the previous frame timestamp contained an integer value of 25, which will be the RTP timestamp value of the third packet in the video frame? We assume that the frequency of the media clock at the sender is 90KHz and FPS=30.

(2) A RTP sender generates PCM coded audio at 16 bits per simple. How many octets does the RTP payload have if we asume that P=0 in the RTP header? Information: In the SDP exchange, the offer included the following lines:
- ptime=40
- maxptime=40

(3) A host has two connected video cameras and two connected microphones. It is transmitting the respective signals towards another host in the Internet by using the RTP protocol. The host is using two RTP sessions, one for audio and another one for video.
- How many different synchronization sources exist?
- How does the RTP receiver know how to correlate the audio signals with the corresponding video signals in order to achieve lipsync?

(4) A RTP transmitter of PCM coded voice at 8-bit has received a RTCP RR packet with the following values: LSR=1200, DLSR=400. If the time as indicated by the media clock in the RTP transmitter when it receives the previous RTCP RR is 2000, which is the RTT in milliseconds?

(1) When a frame is fragmented, all fragments must have the same timestamp as the original frame

$$TS1 = TS0 + \frac{\text{Media Clock Rate}}{\text{Video Frame Rate}} = 25 + \frac{9000\cancel{0}}{3\cancel{0}} = 25 + 3000 = 3025$$

- TS0 = 25
- FPS (Video Frame Rate) = 30
- Media clock Rate = 90 KHz = 90 000 Hz

(2) 1 sample = 16 bits
P = 0 → No padding. Media doesn't include "extra" zeros.
ptime = 40 → media length is 40 ms
maxptime = 40 → max media length is 40 ms
$f = 8000\ Hz \rightarrow T = \frac{1}{8000} = 0{,}000125\ s = 0{,}125 \cdot 10^{-3}\ s \rightarrow$ 1 sample every $0{,}125 \cdot 10^{-3}\ s$

$$\frac{40 \cdot \cancel{10^{-3}}\ s}{0{,}125 \cdot \cancel{10^{-3}}\ s} = 320$$ → The media contains 320 samples

1 sample = 16 bits = 2 octets → 320 · 2 = 640 octets → The media contains 640 octets

(3)

	A → B	
CAM 1	$\xrightarrow{SSRC1}$	SESSION 1
CAM 2	$\xrightarrow{SSRC2}$	
MIC 1	$\xrightarrow{SSRC3}$	SESSION 2
MIC 2	$\xrightarrow{SSRC4}$	

- There are 4 synchronization sources
- The association between audio and video is achieved with the CNAME parameter found in SDES RTCP messages

(4) LSR = 1200
DLSR = 400
Current time = 2000

RTT = current time − LSR − DLSR = 2000 − 1200 − 400 = 400

$$RTT\,(ms) = \frac{400}{fs} = \frac{4\cancel{00}}{80\cancel{00}} = 0{,}05\ ms$$

ACTIVITY 4 (2 PTS)

Please answer the following questions related to SDP:

(1) Indicate if the following SDP exchange is correct or not. Why?.

OFFER

```
v = 0
o = - 2890844526 2890844526 IN IP4 host1.ocean.com
s = -
c = IN IP4 host1.ocean.com
t = 0 0
```

ANSWER

```
v = 0
o = - 2890877711 2890822211 IN IP4 host2.ocean.com
s = -
c = IN IP4 host2.ocean.com
t = 0 0
m = audio 42000 RTP/AVP 0
a = rtpmap:0 PCMU/8000
```

(2) Indicate if the following SDP exchange is correct or not. Why?

OFFER

```
v = 0
o = - 2890844526 2890844526 IN IP4 pc.ocean.com
s = -
c = IN IP4 host.ocean.com
t = 0 0
m = audio 49170 RTP/AVP 0
a = rtpmap:0 PCMU/8000
a=sendonly
```

ANSWER

```
v = 0
o = - 2890897755 2890899432 IN IP4 host.sea.com
s = -
c = IN IP4 host.sea.com
t = 0 0
m = audio 42000 RTP/AVP 0
a = rtpmap:0 PCMU/8000
a = inactive
```

(3) Indicate if the following SDP exchange is correct or not. Why?.

OFFER

```
v = 0
o = - 2890844526 2890844526 IN IP4 host.ocean.com
s = -
c = IN IP4 host.ocean.com
t = 0 0
m = audio 49170 RTP/AVP 0
a = rtpmap:0 PCMU/8000
m = video 51372 RTP/AVP 31
a = rtpmap:31 H261/90000
```

ANSWER

```
v = 0
o = - 2890897755 2890899432 IN IP4 host.sea.es
s = -
c = IN IP4 host.sea.es
t = 0 0
m = audio 42000 RTP/AVP 0 100
a = rtpmap:0 PCMU/8000
a = rtpmap:100 telephone-event/8000
```

(4) Given the following SDP offer generated by host A, which is the maximum data rate generated in the session?

```
v = 0
o = - 2890844526 2890844526 IN IP4 host.sea.es
s = -
c = IN IP4 host.ocean.com
t = 0 0
m = audio 49170 RTP/AVP 0
a = rtpmap:0 PCMU/8000
b = AS:64
b = RS:800
b = RR:2400
```

(1) Not correct because offer and answer must have the same number of m-lines

(2) Correct

(3) Not correct because offer and answer must have the same number of m-lines

(4) MAX RATE = AS + RS + RR = 64000 + 800 + 2400 = 67200 bit/s

- AS = 64 kbit/s = 64000 bit/s
- RS = 800 bit/s
- RR = 2400 bit/s