MTA   2003ba:

Unresolved question
18.b.v. SIP is syncronizing RTP streams ? do they mean mixer ?

1. Not in the material
2. b
3. a+c
4. c
5. a
6. d
7. d
8. b
9. d
10.b+ d (very similar to question 6, from 2009 moed a in TAU, but notice to answers b & d which are different)
11. b
12. b
13. d (or b or c, don't know what is MFAS and MFC)
14. c+e
15. d
16. d

17.
a.
i.   presentation 2, slide 26.
ii.   In the request: according to the Request-URI and "Route" field.
        In the response: according to the first "via:" header which determines the next route.
iii.   A request determines the path all its' response messages (which should go in the same path).
        Future requests can be sent in other path.
        Usually they will go directly without a proxy because the
        destination location is already known, unless Record-Route" was added by some proxies in the beginning of the dialog,
        in such case, each message will have "Route" field for each of these proxies and the message will go through tem.
iv.   The "contact:" header.
v.   This is done in SDP protocol.
        The SIP header related to it is: "Content-Type: application/sdp" and Content-Length"
        If the meaning in the question was the SDP header,
        this is the Media header, which starts with "m=".
b.
i.   We want 350 channels, PCM-30 has 30 channels, so we need 12 lines (11 and 2/3).
ii.   Channel 140 is in the 5th line.
        The channels in this line are 121 to 150, so 140 is channel number 20.
        Channel 140, Track 20 is in TS 21.
        The signaling (from source to destination) is on the right 4 bits of TS 16 in Frame 5 in each MF.
        The signaling (from destination to source) - is on the right 4 bits of TS 16 in Frame 5 in each MF, on the

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        E1s from Haifa to Tel Aviv.
iii.    Now each PCM has 31 channels, but we still need 12 lines
(at least more than 11).
iv.     Channel 140 is in the 5th line.
        The channels in this line are 125 to 155, so 140 is
channel number 16.
        Channel 140, Track 16 is in TS 16.
        The signaling location depends on implementation, it
doesn't concern us.

18.
a.
i.      in time 4ms.
ii.     sin wave with height of 7 and length of 4.
iii.    7
iv.     frequency is f=1/T, T=0.004sec, so f = 1/0.004 = 250 Hz.
v.      180 degrees. (presentation 1, slide 13)
vi.     the line x=0.25, from height 0 to 7.
        (X is frequency in Khz, Y is amplitude) - (presentation 1,
slide 16)
b.
i.      It is used to order the received packets in the
de-jittering buffer and to composed multiple RTP streams to one
stream (by the mixer).
ii.     The TS units are "decoder pulses", i.e, it is a counter of
how many decoder
        pulses has been since the beginning of the session to the
        time of the first sample in the current RTP message
        The 2 sides sync on the TS in RTCP message. The sender
sends
        NTP time (which is a global syncronized time) and its' TS
in this time.
        The receiver gets it, compares to his time, and can see
the delta, and syncronize itself.
        The 2 sides know the relation between "decoder-pulses" and
real time, by the "a=..." header of SDP,
         which is contained in INVITE and 200 OK messages (of
SIP). This line determines the amount of samples per seconds.
iii.    SN is the serial number of the RTP message.
        We can predict TS, if we have a constant rate of messages,
we can know the delta time between
        2 messages.
        It is not always applicable, because sometimes there is
"quiet",
        and there is no need to send messages, so 2 messages will
have
        different delta in the TS.
        On the other hand, we can't use TS without SN, because we
need to know
        if we the resason that there was a certain delta TS is
because a
        message was lost, or because there was "quiet".
iv.     Jitter (?) - (presentation 2, slides 22-23)
v.      SIP is not syncronizing RTP, SIP is for signaling (unless
we count the mixer as SIP vertex).
        The mixer is doing it, by using the NTP and TS fields
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of SR (RTCP) message from each source.

19.
a.
i. - iii. (presentation 1, slides 35-38).
iv. base = 512 (PCM 80), delta = 406, factor = 32
delta = 406/32=12.68 -> 12.
PCM - 92.
b.
i. (presentation 1, slides 63-65).
ii. There is a field in LAPD to indicate if channel is used for CS signaling or PS.
The field name is SAPI.
iii. Channel B is CS, layer 2-3 is irelevant for it. Only layer 1.
iv. No, it took alot of time to develope it (heavy), and than the ADSL was developed and no one needed ISDN.
The ISDN development was important because the cellular technology used
some of its' principles.

20. Same as question 20 in 2005bb.

21.
a.
i.
- Pair of lines and telephone at each end-point.
- Multinet notes - lecture 1 (first page).
ii-iv. Same as question 21 in 2005bb, section ב
b.
i-iii. (presentation 1, slides 53-60).
iv. Question 20, ב, i
c. Same as question 21 in 2005bb, sections ו.ז
d.
i. SIP and H.323
ii. (presentation 2, slide 33).
iii. Yes. When the caller knows the exact and updated current address of the desination.
In this case, the message can be sent directly.
iv. RTP over UDP (used with both SIP and H.323 ?).
SIP is not restricted to work with RTP, but we haven't learned any other protocol which is used with it.