

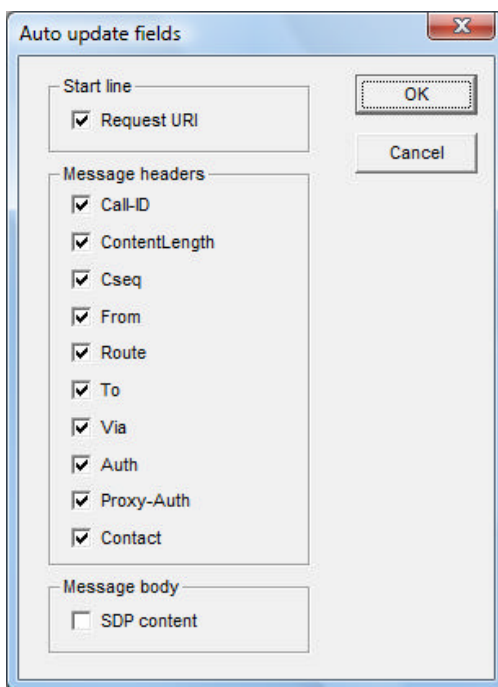
- Supported media - SDP content -

There are different levels of testing

1. First case :

You want to send unconditionally a SDP content in the INVITE, without checking the SDP parameter defined in the Emulator panel (SIP – SDP negotiator panel).

In this case, you should disable the SDP update in the AutoUpdate panel of the message.



SDP content will be transmitted "as it is", as SDP negotiation doesn't occur, the call can go to an active state but no RTP testing can be run.

This is used if you want to test the behaviour of the IUT when receiving a specific SDP. The SDP content could be erroneous, with illegal characters, it doesn't matter because there is no testing of the SDP content.

2. Second case :

You want to simulate the behaviour of a real UA with SDP negotiation.

In this case, you must :

- specify capabilities of the UA using the SDP negotiator panel (example: Payload = audio, Media = 104, Transport = RTP/AVP, Rtpmap = 104 CLEARMODE/8000)
- specify the SDP template in the message and enable AutoUpdate of SDP body

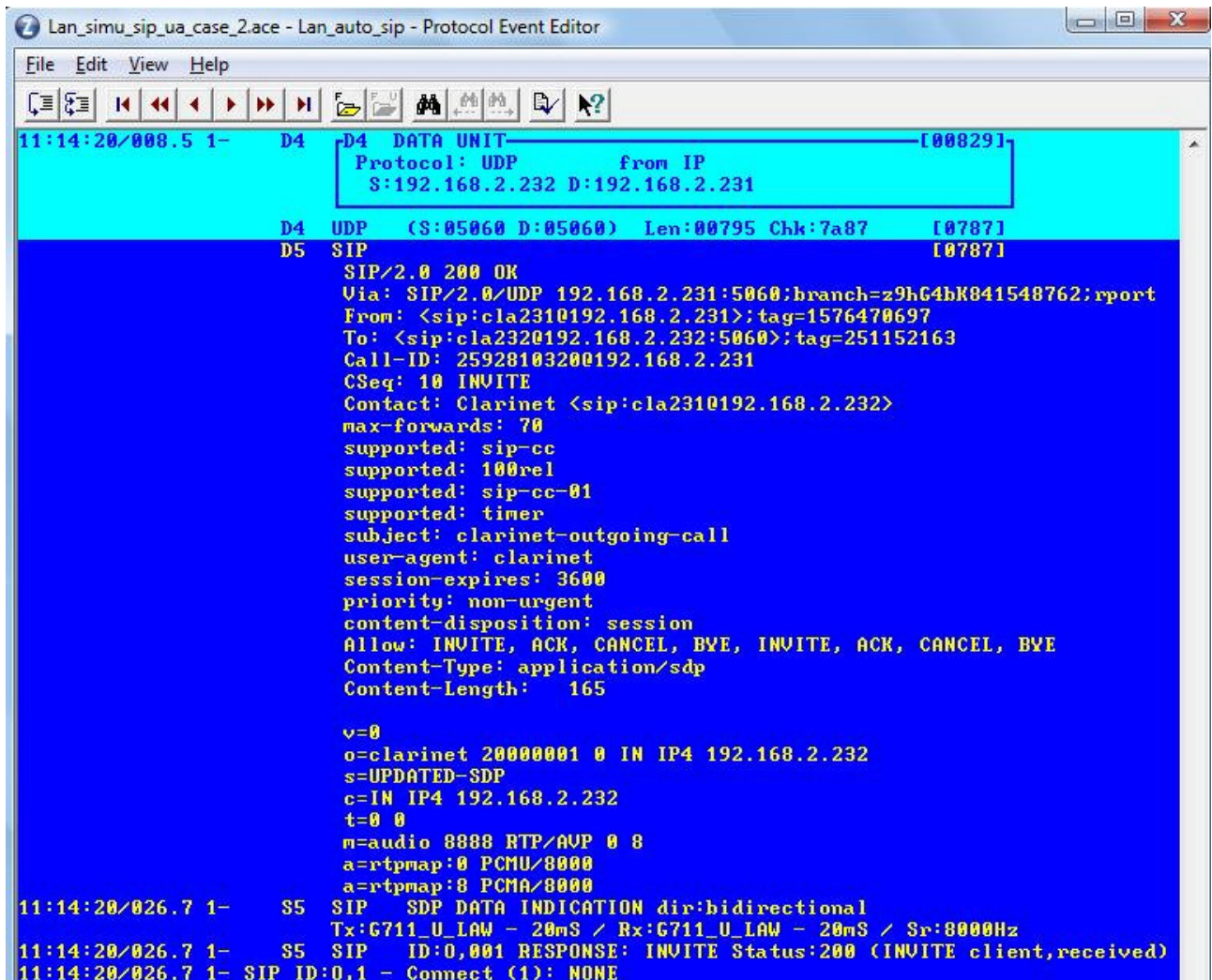
When the message is transmitted, the SDP negotiator is invoked with the SDP template to build a valid SDP offer to be sent in the outgoing message.

If negotiation fails due to invalid SDP content, or unsupported media in SDP negotiator panel, then nothing will be sent.

Alternatively, full SDP content will be updated, including addresses, port numbers ...

When an incoming call is received, the SDP is also checked with SDP parameters, this allows simulation of different behaviours : supported or unsupported media.

In this case, the SDP has been updated but the preferred media list is unchanged and the call is successful.



```

11:14:20/008.5 1- D4 D4 DATA UNIT [00829]
  Protocol: UDP From IP
  S:192.168.2.232 D:192.168.2.231
D4 UDP (S:05060 D:05060) Len:00795 Chk:7a87 [0787]
D5 SIP [0787]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.231:5060;branch=z9hG4bK841548762:rport
From: <sip:cla2310192.168.2.231>;tag=1576470697
To: <sip:cla2320192.168.2.232:5060>;tag=251152163
Call-ID: 25928103200192.168.2.231
CSeq: 10 INVITE
Contact: Clarinet <sip:cla2310192.168.2.232>
max-forwards: 70
supported: sip-cc
supported: 100rel
supported: sip-cc-01
supported: timer
subject: clarinet-outgoing-call
user-agent: clarinet
session-expires: 3600
priority: non-urgent
content-disposition: session
Allow: INVITE, ACK, CANCEL, BYE, INVITE, ACK, CANCEL, BYE
Content-Type: application/sdp
Content-Length: 165

v=0
o=clarinet 20000001 0 IN IP4 192.168.2.232
s=UPDATED-SDP
c=IN IP4 192.168.2.232
t=0 0
m=audio 8888 RTP/AVP 0 8
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
11:14:20/026.7 1- S5 SIP SDP DATA INDICATION dir:bidirectional
Tx:G711_U_LAW - 20mS / Rx:G711_U_LAW - 20mS / Sr:8000Hz
11:14:20/026.7 1- S5 SIP ID:0,001 RESPONSE: INVITE Status:200 (INVITE client,received)
11:14:20/026.7 1- SIP ID:0,1 - Connect (1): NONE
    
```

Note : for example, you had coded the clarinet SDP negotiator panel to include "Audio 5100 RTP/AVP 104", as a supported codec. The value "5100" is the port number, so it's not a part of the "supported codec" specification of the SDP negotiator panel. If you want that port 5100 is used, you must specify it as "Audio port min"

3. **Third case :**

You want to simulate the behaviour of a real UA + RTP testing.

It's the same as case 2, but the codec chosen after full SDP negotiation must be supported by Clarinet and not only defined in SDP negotiator panel.

