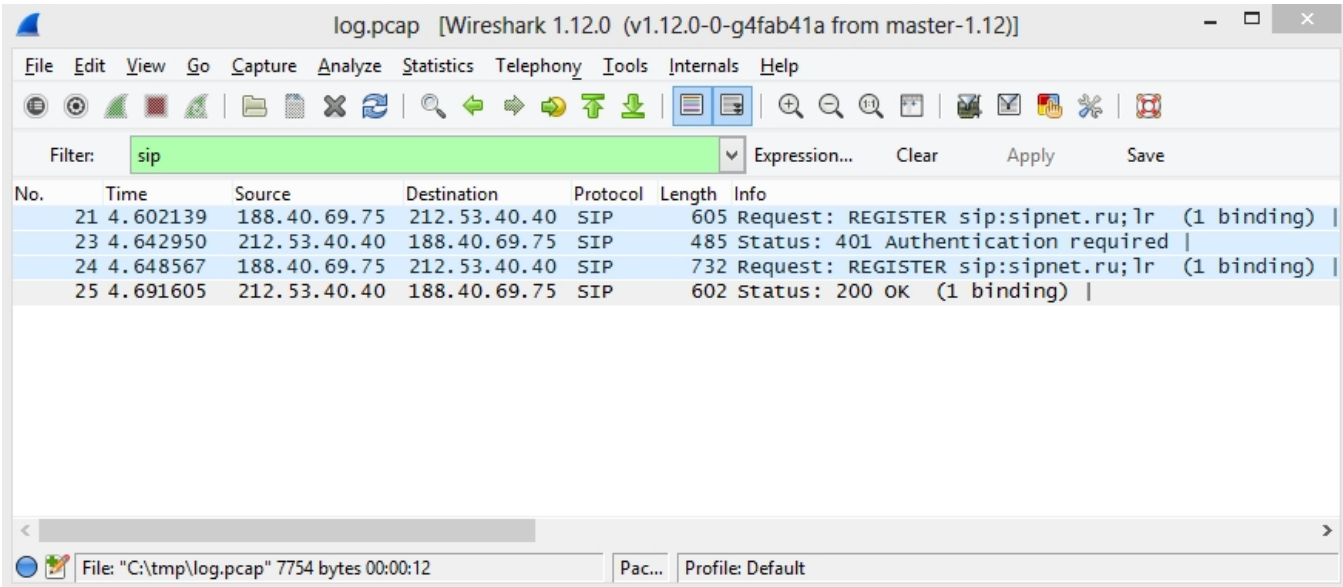


SIP

SIP registration

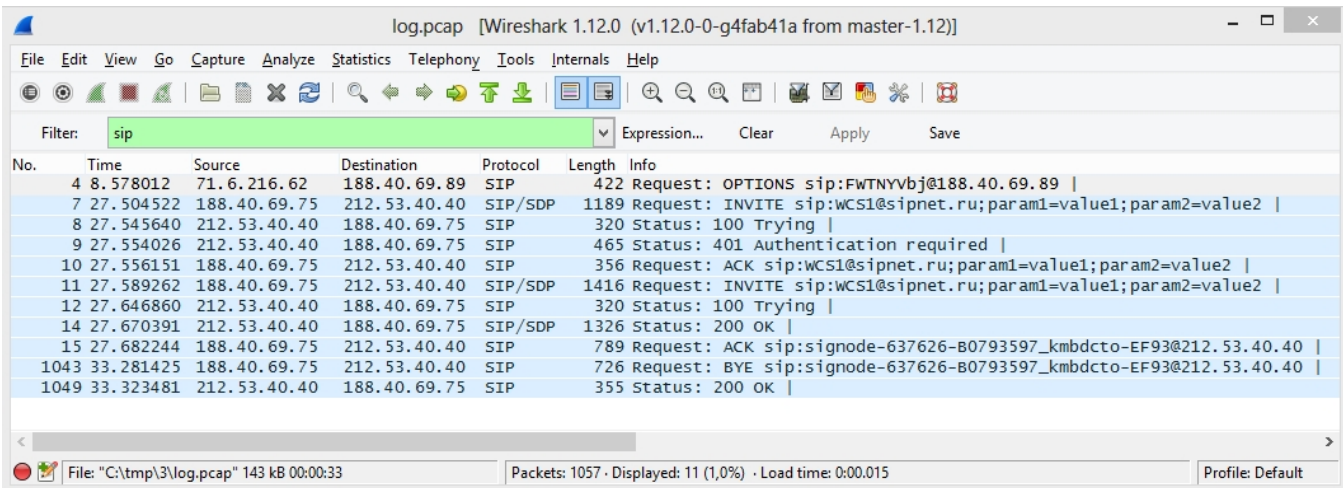
By default, Webphone attempts to register on the SIP server. A sample SIP dump below displays how successful SIP registration goes. The dump is filtered by the 'sip' protocol.



No.	Time	Source	Destination	Protocol	Length	Info
21	4.602139	188.40.69.75	212.53.40.40	SIP	605	Request: REGISTER sip:sipnet.ru;lr (1 binding)
23	4.642950	212.53.40.40	188.40.69.75	SIP	485	Status: 401 Authentication required
24	4.648567	188.40.69.75	212.53.40.40	SIP	732	Request: REGISTER sip:sipnet.ru;lr (1 binding)
25	4.691605	212.53.40.40	188.40.69.75	SIP	602	Status: 200 OK (1 binding)

Outgoing SIP call

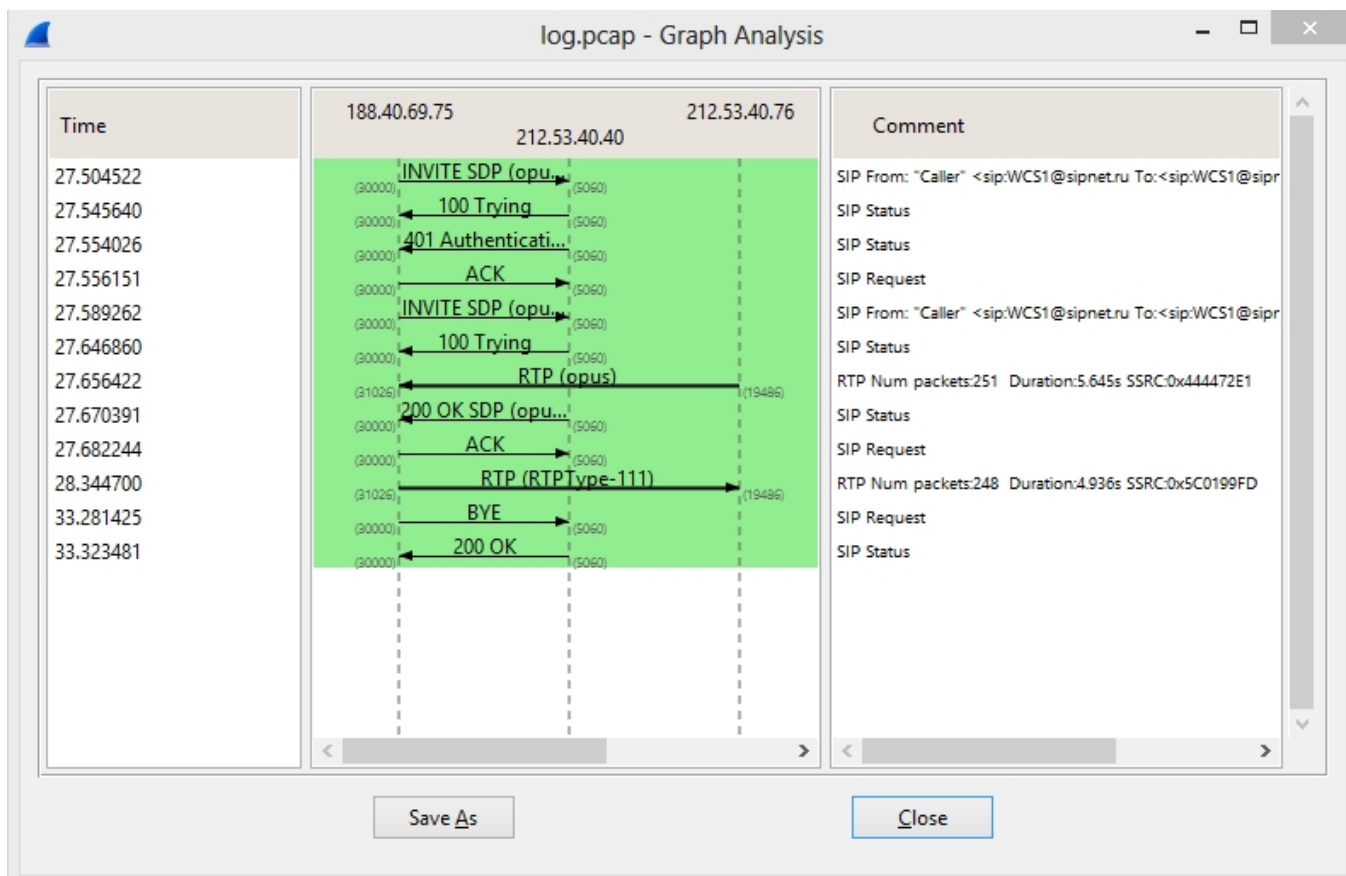
The call begins with the SIP INVITE transaction and ends with the call to BYE from the SIP server.



No.	Time	Source	Destination	Protocol	Length	Info
4	8.578012	71.6.216.62	188.40.69.89	SIP	422	Request: OPTIONS sip:FWTNYVbj@188.40.69.89
7	27.504522	188.40.69.75	212.53.40.40	SIP/SDP	1189	Request: INVITE sip:wcs1@sipnet.ru;param1=value1;param2=value2
8	27.545640	212.53.40.40	188.40.69.75	SIP	320	Status: 100 Trying
9	27.554026	212.53.40.40	188.40.69.75	SIP	465	Status: 401 Authentication required
10	27.556151	188.40.69.75	212.53.40.40	SIP	356	Request: ACK sip:wcs1@sipnet.ru;param1=value1;param2=value2
11	27.589262	188.40.69.75	212.53.40.40	SIP/SDP	1416	Request: INVITE sip:wcs1@sipnet.ru;param1=value1;param2=value2
12	27.646860	212.53.40.40	188.40.69.75	SIP	320	Status: 100 Trying
14	27.670391	212.53.40.40	188.40.69.75	SIP/SDP	1326	Status: 200 OK
15	27.682244	188.40.69.75	212.53.40.40	SIP	789	Request: ACK sip:signode-637626-B0793597_kmbdcto-EF93@212.53.40.40
1043	33.281425	188.40.69.75	212.53.40.40	SIP	726	Request: BYE sip:signode-637626-B0793597_kmbdcto-EF93@212.53.40.40
1049	33.323481	212.53.40.40	188.40.69.75	SIP	355	Status: 200 OK

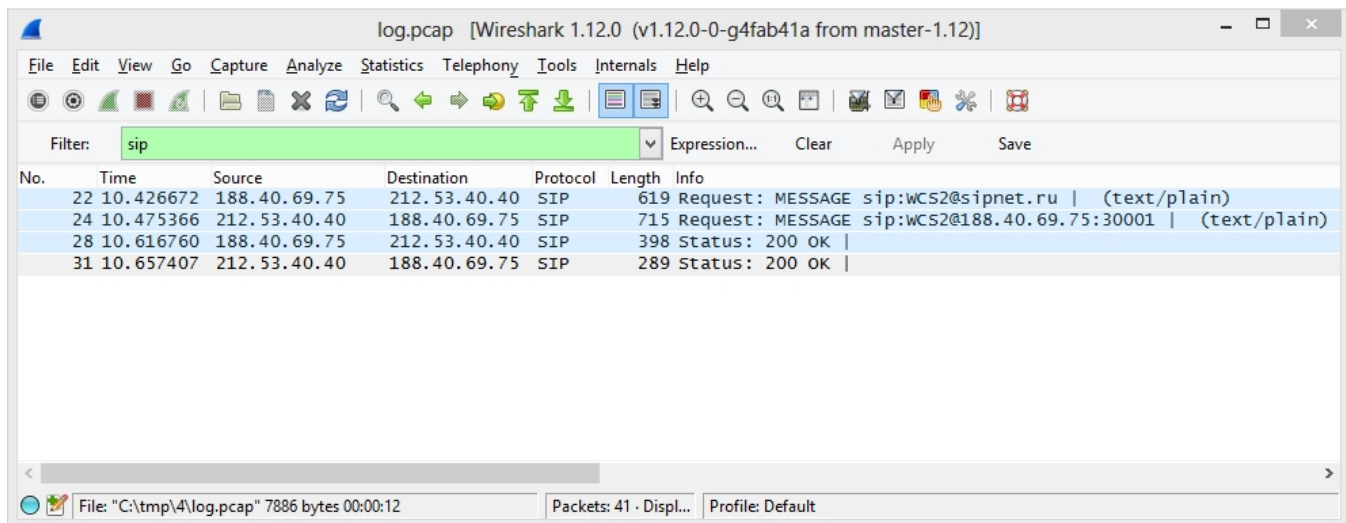
SIP call traffic analysis

Wireshark allows to analyze individual SIP calls via the menu: Telephony - VoIP Calls - Flow. Below is an example analysis of the outgoing call based on the dump. Analysis shows that 251 RTP packets were received from the server as Opus encoding, and 248 audio packets were sent.



SIP messages

Messages are sent with the SIP MESSAGE transaction in accordance with [RFC3428](#).



Possible problems

Problems with SIP traffic can lead to problematic calls, SIP messages and other functionality dependent on SIP. If SIP traffic flows normally, problems may arise when interacting with the SIP server, for example, if the SIP server returns SIP statuses of 4xx-6xx errors for unknown reason and does not allow SIP registration, SIP calls or other SIP transactions.

Troubleshooting

If SIP registration fails, make sure your SIP server correctly receives and processes SIP requests. Check operation of your SIP server using any desktop softphone, like Bria. Make sure Firewall allows receiving traffic to SIP ports [30000-31000] specified in the port_from and port_to settings of the [flashphoner.properties](#) settings file.

If SIP calls or messages do not pass through, make SIP traffic dumps and contact the administrator of the SIP server or your SIP provider for clarification. You can also contact Flashphoner technical support. If the problem is on the side of the SIP server, our specialists will help you to formulate an inquiry to the SIP provider (administrator) with proper traffic dump description and questions.