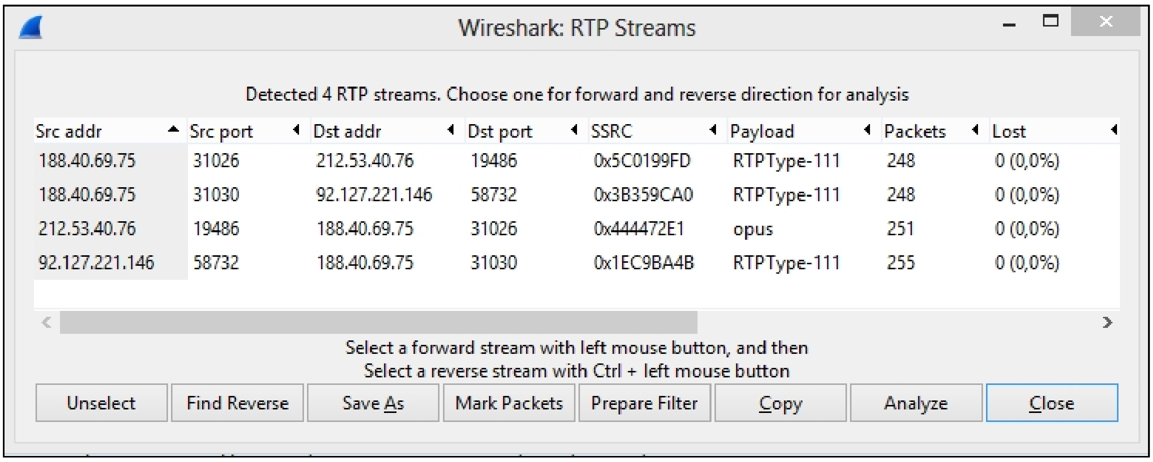
RTP

RTP stream

The traffic flows between WCS and the SIP server. You can find it in the RTP traffic dump by IP addresses (IP addresses of browser, WCS server and SIP server are known to us).


Wireshark: RTP Streams

Detected 4 RTP streams. Choose one for forward and reverse direction for analysis

Src addr	Src port	Dst addr	Dst port	SSRC	Payload	Packets	Lost
188.40.69.75	31026	212.53.40.76	19486	0x5C0199FD	RTPType-111	248	0 (0,0%)
188.40.69.75	31030	92.127.221.146	58732	0x3B359CA0	RTPType-111	248	0 (0,0%)
212.53.40.76	19486	188.40.69.75	31026	0x444472E1	opus	251	0 (0,0%)
92.127.221.146	58732	188.40.69.75	31030	0x1EC9BA4B	RTPType-111	255	0 (0,0%)

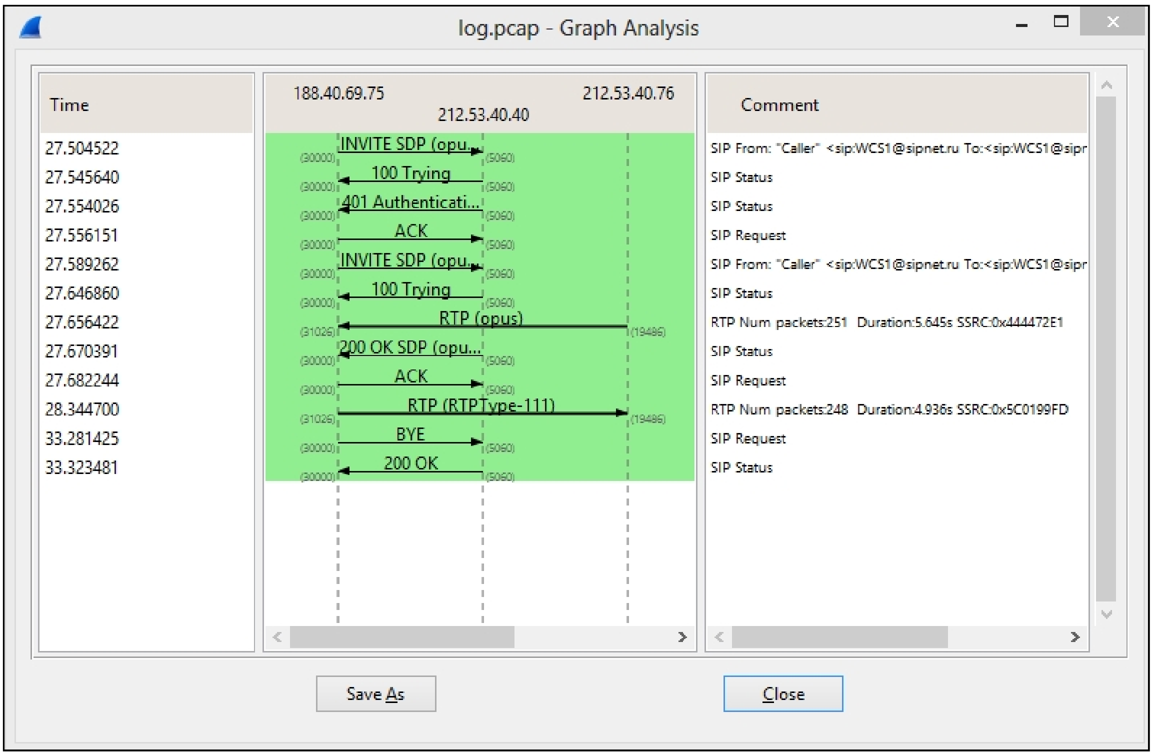
Select a forward stream with left mouse button, and then
Select a reverse stream with Ctrl + left mouse button

Unselect | Find Reverse | Save As | Mark Packets | Prepare Filter | Copy | Analyze | Close

In this specific case RTP traffic is represented by streams with SSRC 0x5C and 0x44.

SIP call traffic analysis

The following diagram shows how SIP and RTP traffic of one SIP call flows



From the analysis of the SIP call we can see that WCS receives and sends RTP audio streams with audio data encoded by Opus.

Possible problems

In most cases problems are related to RTP traffic not flowing between WCS and the SIP server.

Troubleshooting

Make sure the RTP traffic is unhindered and can flow from the WCS server and back. The default media ports of the WCS server 31000-32000 should be open to receive the incoming UDP traffic. If the WCS server is behind NAT and has an external IP address, make sure UDP packets sent to this external address are correctly routed to the corresponding ports of the WCS server behind NAT.