RTP

RTP stream

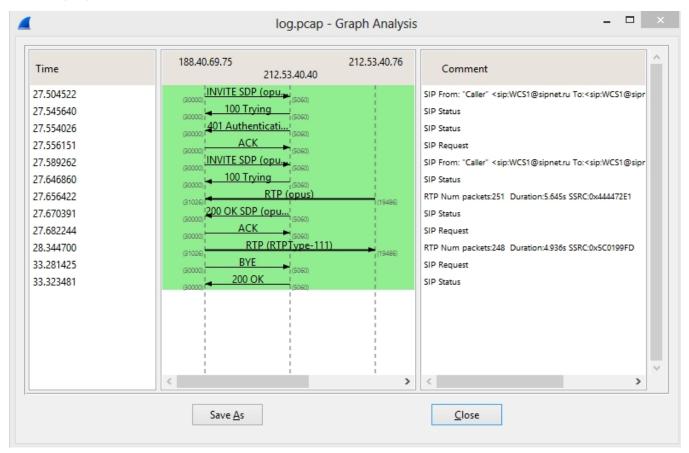
The traffic floes beetween WCS and the SIP server. You can find it in the RTP traffic dump by IP addresses (IP addresses of browser, WCS server and SIP server are known to us).

[Wireshark:	RTP Streams				
	Detec	ted 4 RTP streams.	Choose one for	forward and rev	erse direction for a	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%)	
<								3
				h left mouse butt th Ctrl + left mo				
Unselect	Find Reverse	Save <u>A</u> s	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.