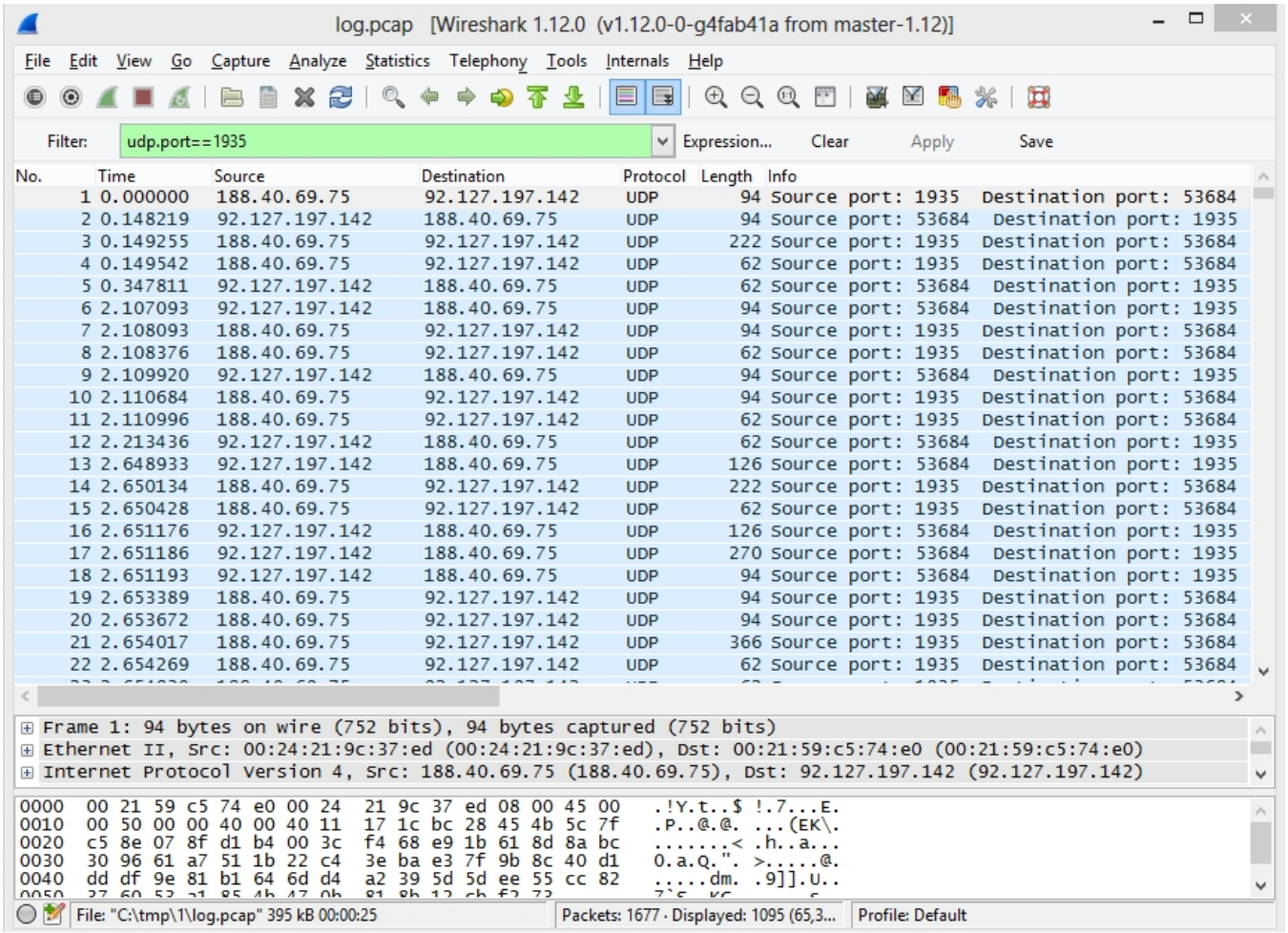
RTMFP

RTMFP traffic analysis

The RTMFP protocol is only used to transfer audio and video data if the WebRTC technology is unavailable. All signal data and messages are sent via the Websocket protocol.
Here, we filter the dump by the UDP protocol, 1935. This is a standard RTMFP port. As seen from the dump, packets flow in both directions, from the web client to the server and back. RTMFP uses AES encryption on the protocol level, so decoding and decomposing the protocol in Wireshark is not possible.



Possible problems

In most cases problems are related to RTMFP traffic not flowing between the web client and the WCS server. If RTMFP traffic does not pass through, the web client can establish connection and register at SIP, because signaling works via Websocket. During calls Flash Player will produce errors, or there will be simply no audio.

Troubleshooting

Make sure the UDP port 1935 is open and available. If the WCS server is behind NAT, make sure UDP packets send to the external IP address does reach that port of the WCS server.