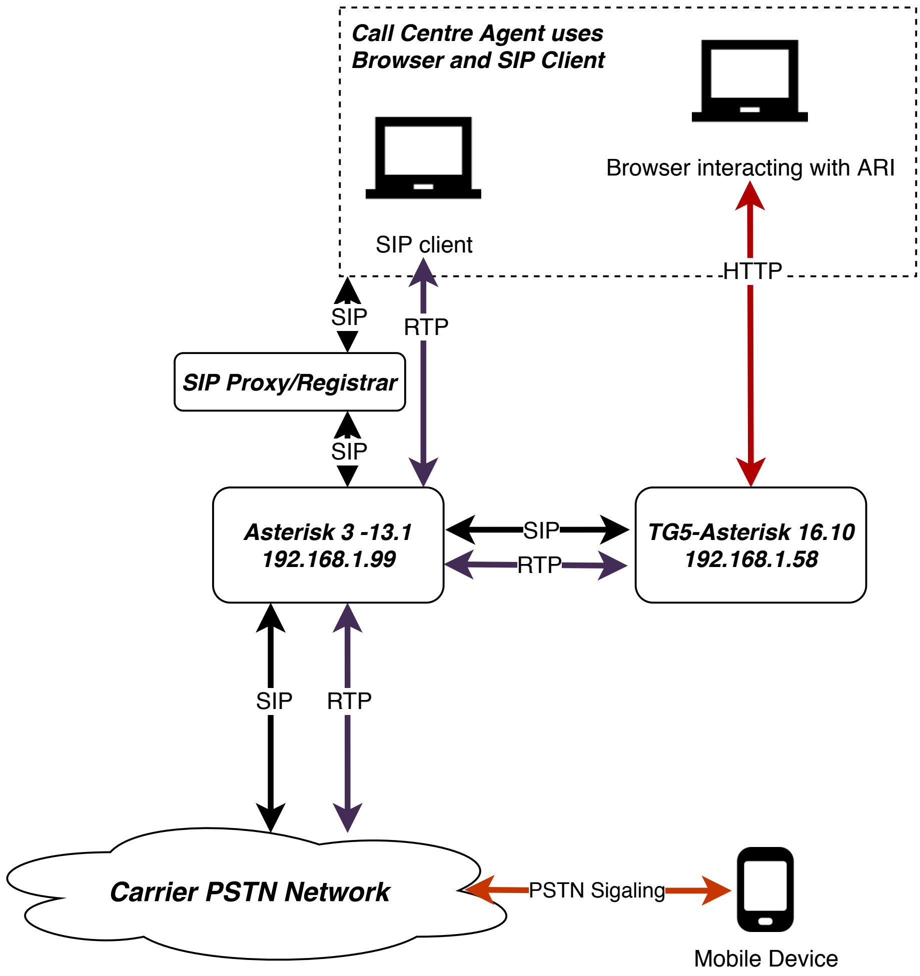
Overview Topology Call centre Agent initiates a call back to its SIP client device via browser and ARI . It then initiates a second call again from the browser out to the Mobile device and the calls are merged together with the record initiated.



RTP Flow To show where we see the issue between Asterisk 3 and TG5 Note Asterisk 3 is the edge device to carrier and SIP client for signaling and media streams involved in this issue. Note IP addresses and ports will match debug and wireshark traces - I have tried to simplify with just showing RTP and not SIP when silent RTP packets are sent. Note legs to Carrier and SIP client are not shown as issues occurs on TG5 as will be seen from wireshark traces. Issue only occurs when recording.