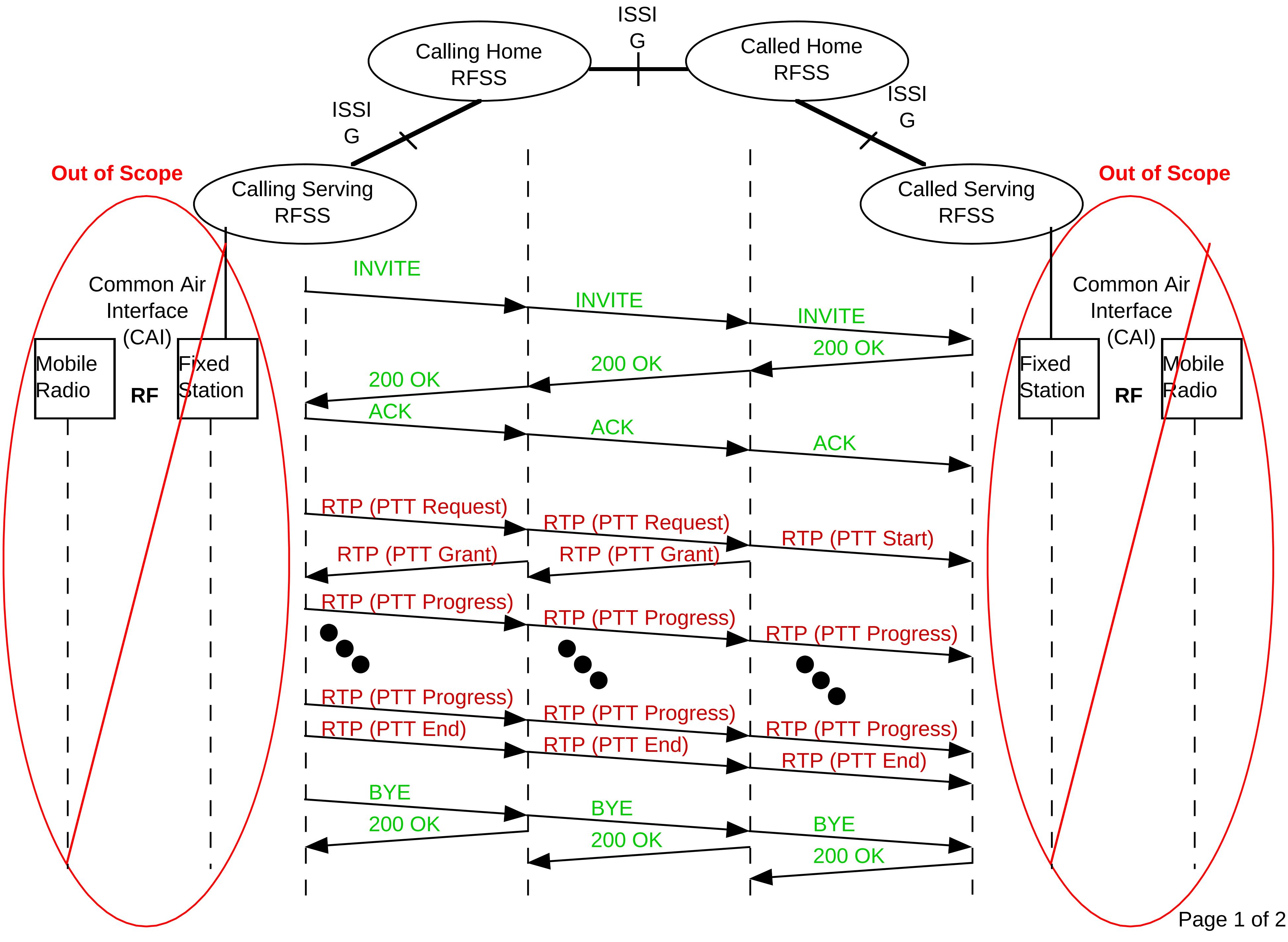
SU-to-SU voice call establishment, audio transmission, and call tear down.



SU-to-SU voice call establishment, audio transmission, and call tear down.

This network topology contains four RFSSs connected by three physical links (indicated by thick black lines). For this particular example of this particular SU-to-SU call, each RFSS is acting with a different function.

An SU within the coverage (serving) area of the Calling Serving RFSS initiates an SU-to-SU voice call. [Not shown; considered out of scope from the ISSI perspective]
The calling serving RFSS sends a SIP INVITE message to the RFSS that is acting as the home for the SU, which initiated the voice call.

The calling home RFSS receives the SIP INVITE message and ensures that the call is permitted and if so forwards on the SIP INVITE message to the RFSS that is acting as the home for the called SU.

The called home RFSS receives the SIP INVITE message and must determine the RFSS where the called SU is currently located before forwarding the SIP INVITE message on to the correct RFSS.

The called serving RFSS receives the SIP INVITE message and responds with a SIP 200 OK message. This SIP 200 OK message is forwarded back through the called home RFSS and calling home RFSS to reach the calling serving RFSS.

The calling serving RFSS upon receiving the SIP 200 OK message responds with a SIP ACK message, which is forwarded through the various functioning RFSSs to reach the called serving RFSS.

Upon the completion of this three-way exchange of SIP messages, the audio data can be requested for transmission using the RTP.

The audio data is exchanged using the RTP by exchanging a PTT Transmit Request packet. A positive response to the request is a PTT Grant packet. Once a PTT Grant packet is received, the audio data is sent using PTT Progress packets. When the talk spurt ends (i.e., no more audio), a PTT End packet is sent.

If this is the end of the SU-to-SU call, then a SIP BYE message is sent and forwarded the the called serving RFSS. The response to a SIP BYE message is a SIP 200 OK message. This two-way message exchange releases any RF resources and any RTP resources used for this call.