Introduction to SIP

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Mobile Applications and Services

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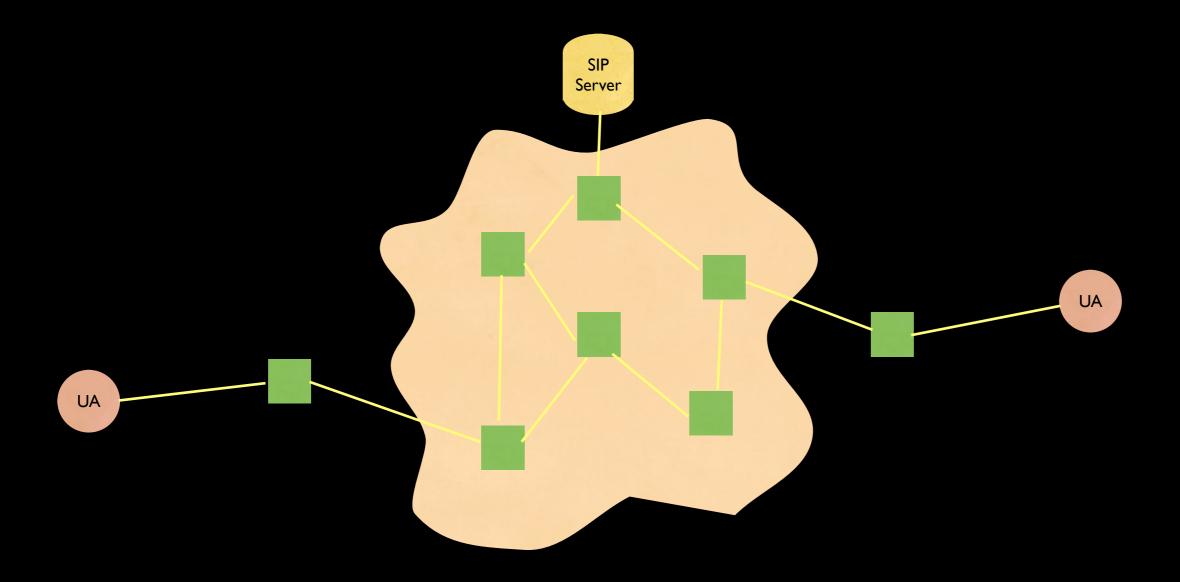
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- Must create the illusion of a call session
 - need a session protocol

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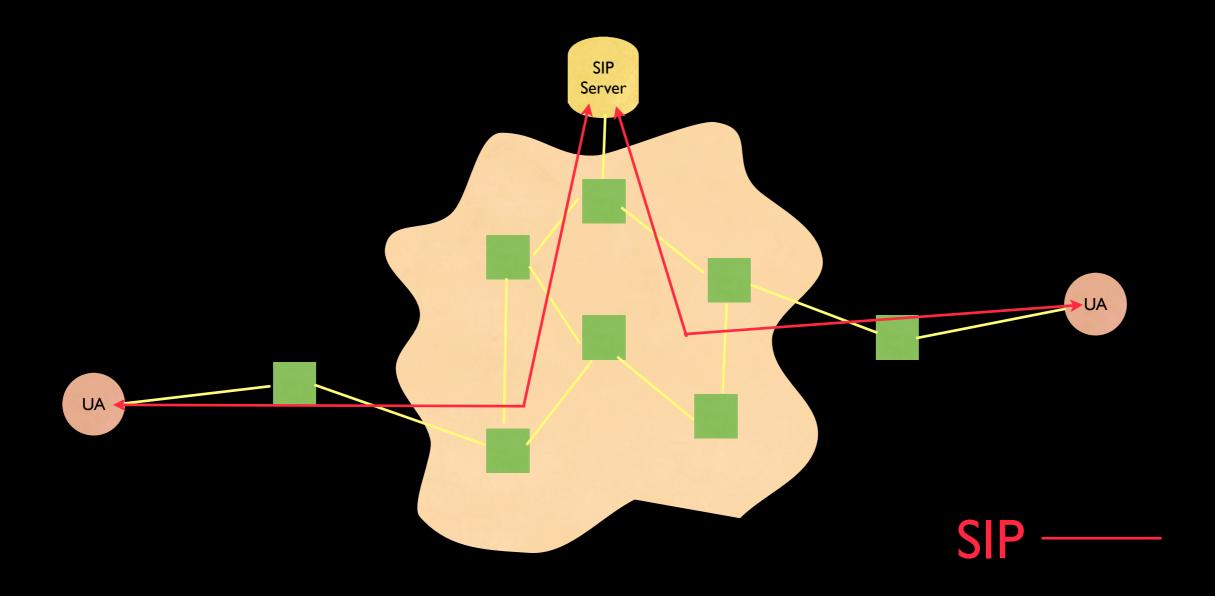
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- RTP Real Time Transport Protocol RFC 3550
 - For the data exchange, a stream of encoded voice packets

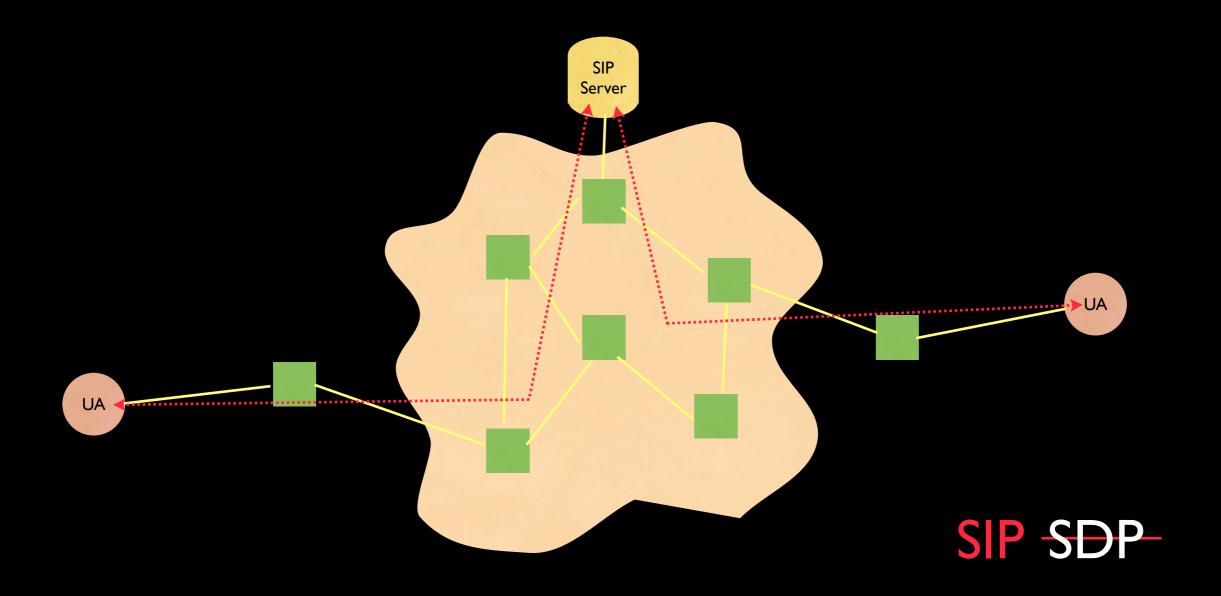
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- RSVP ReSerVation Protocol RFC 2205
 - Establish Priority and Reservations INSIDE the network



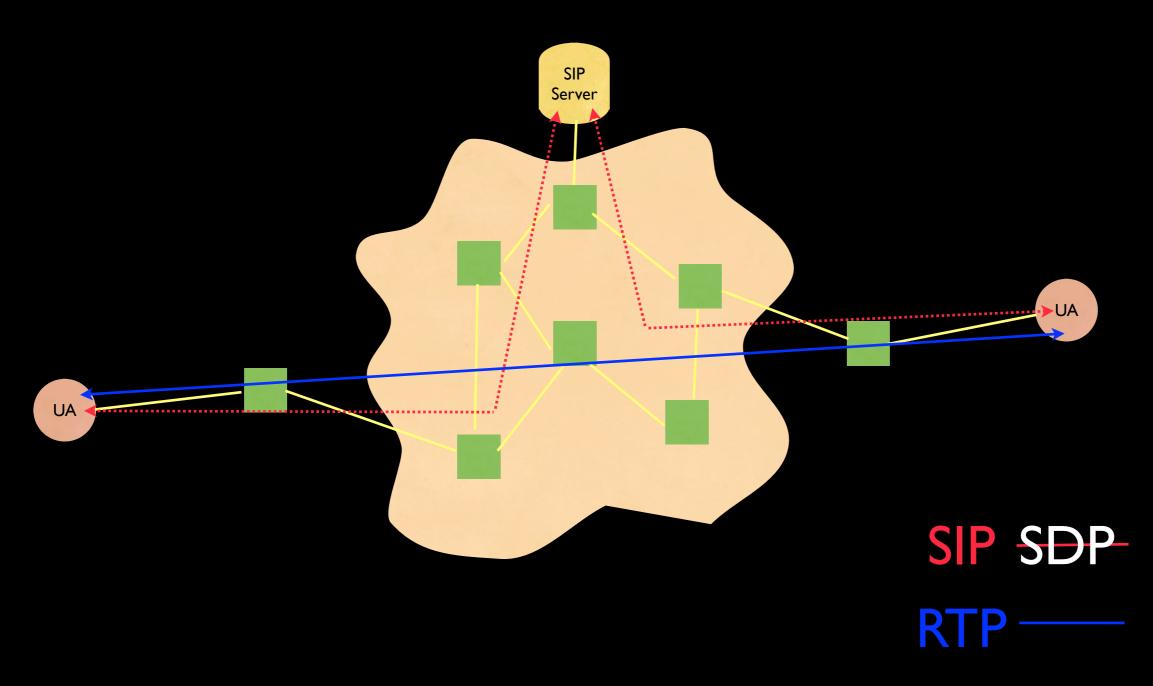
•User Agents (UA) connected over an IP network



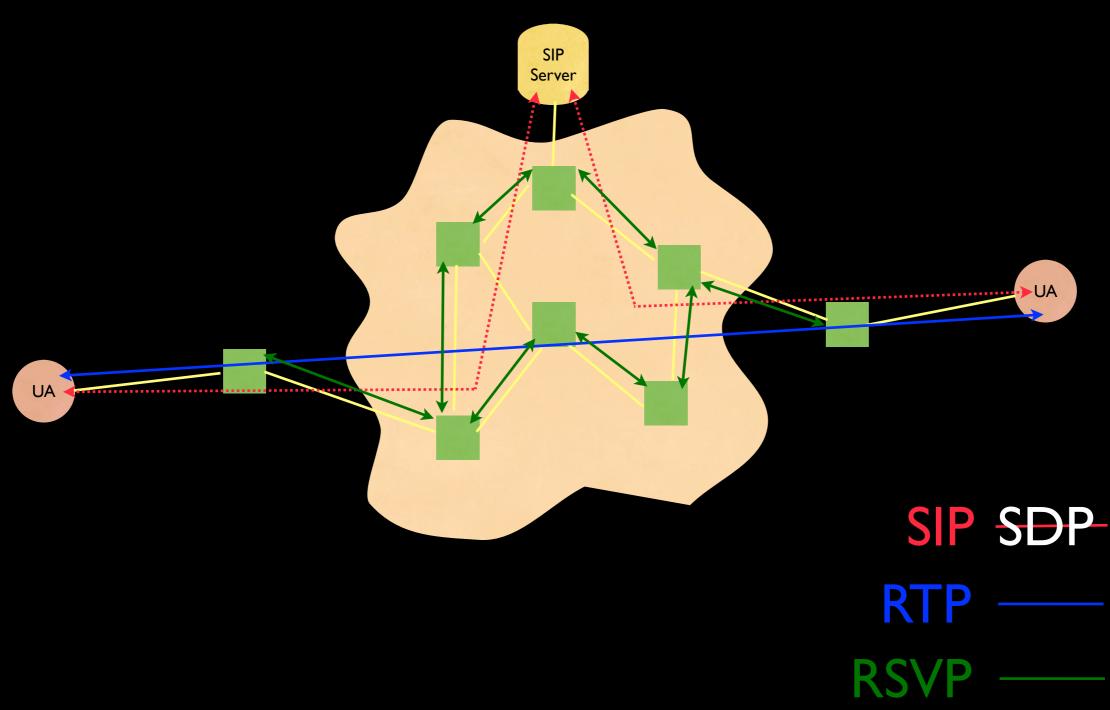
- A SIP server isn't technically required.
- But we always use it.
- •Why? To find you.



•SDP is carried as payload in the SIP INVITE and response



•RTP may not follow the same path as SIP



- •RSVP is internal to the network
- Current IMS implementations are limited here