

# Introduction to SIP

Russ Clark  
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# IP Telephony

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

# VoIP Protocols

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- SDP - Session Description Protocol - RFC 2327
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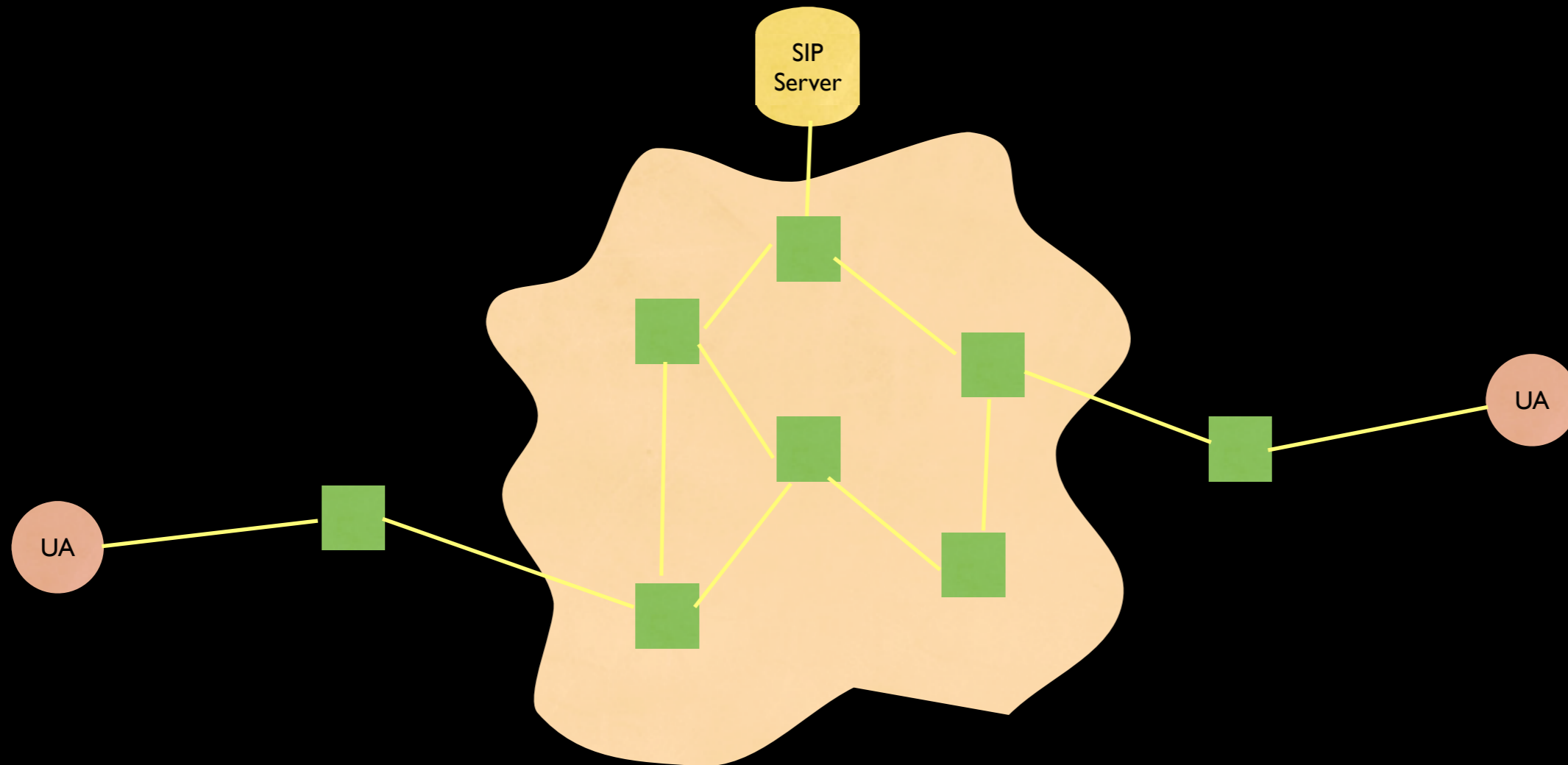
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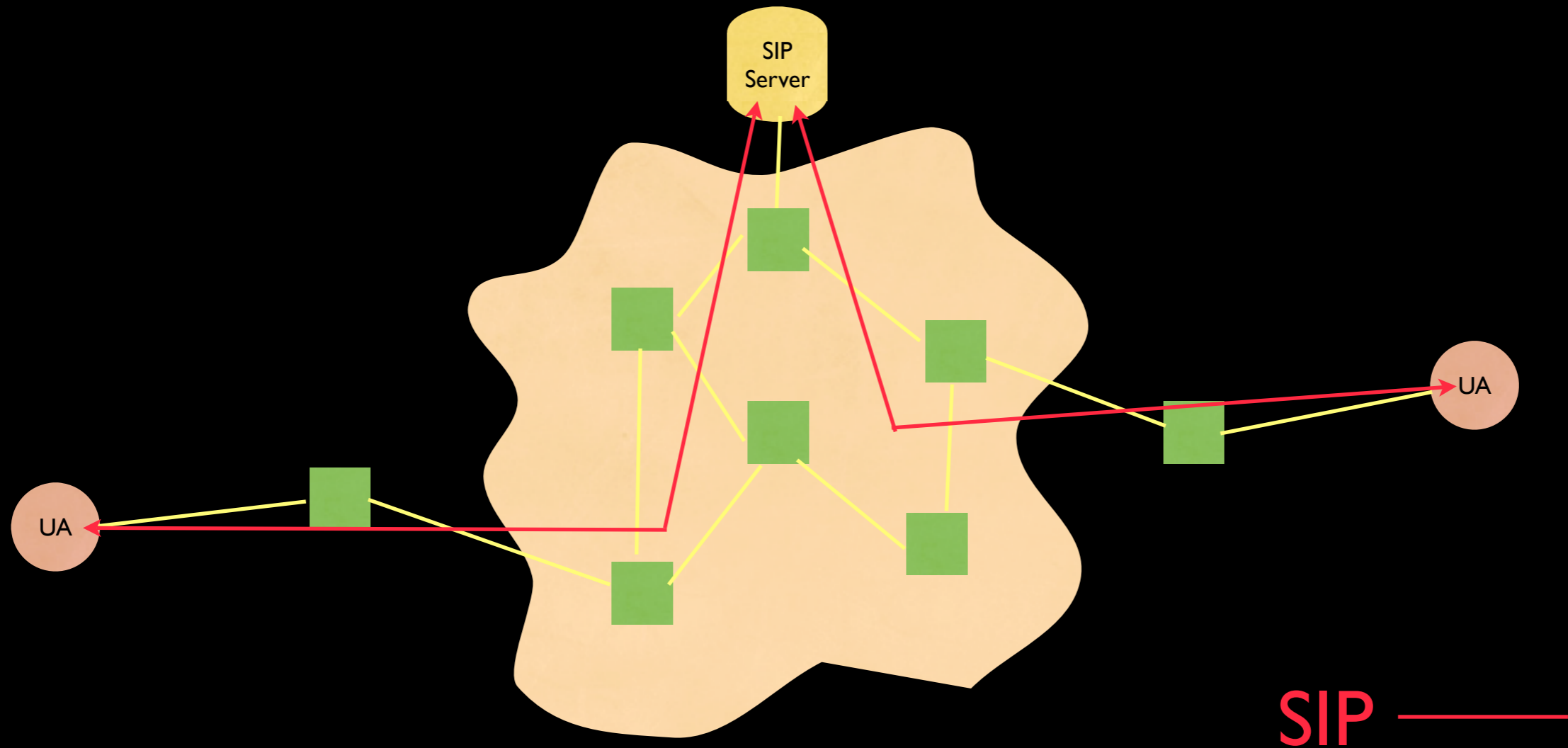
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- RSVP - ReSerVation Protocol - RFC 2205
  - Establish Priority and Reservations INSIDE the network

# VoIP Protocols



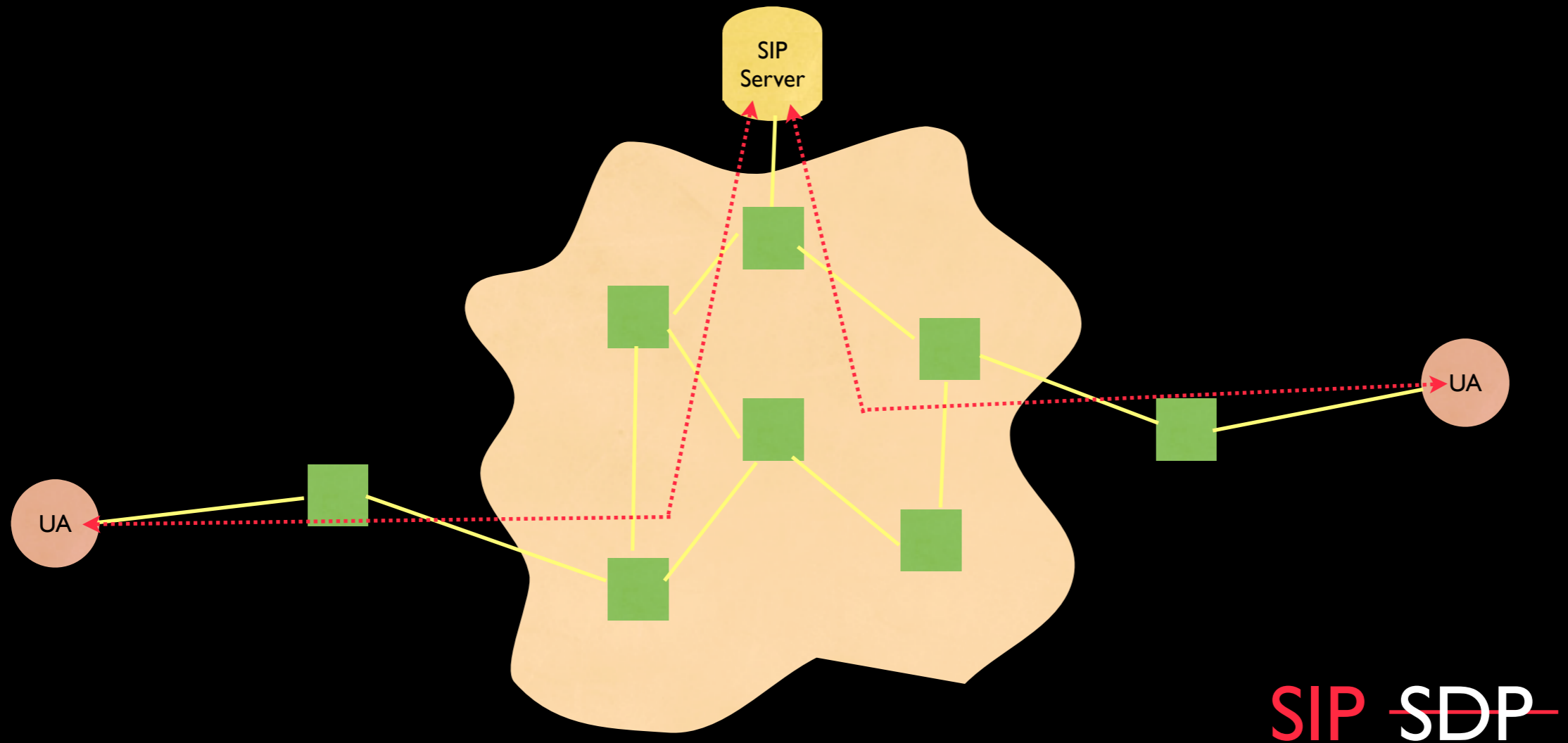
- User Agents (UA) connected over an IP network

# VoIP Protocols



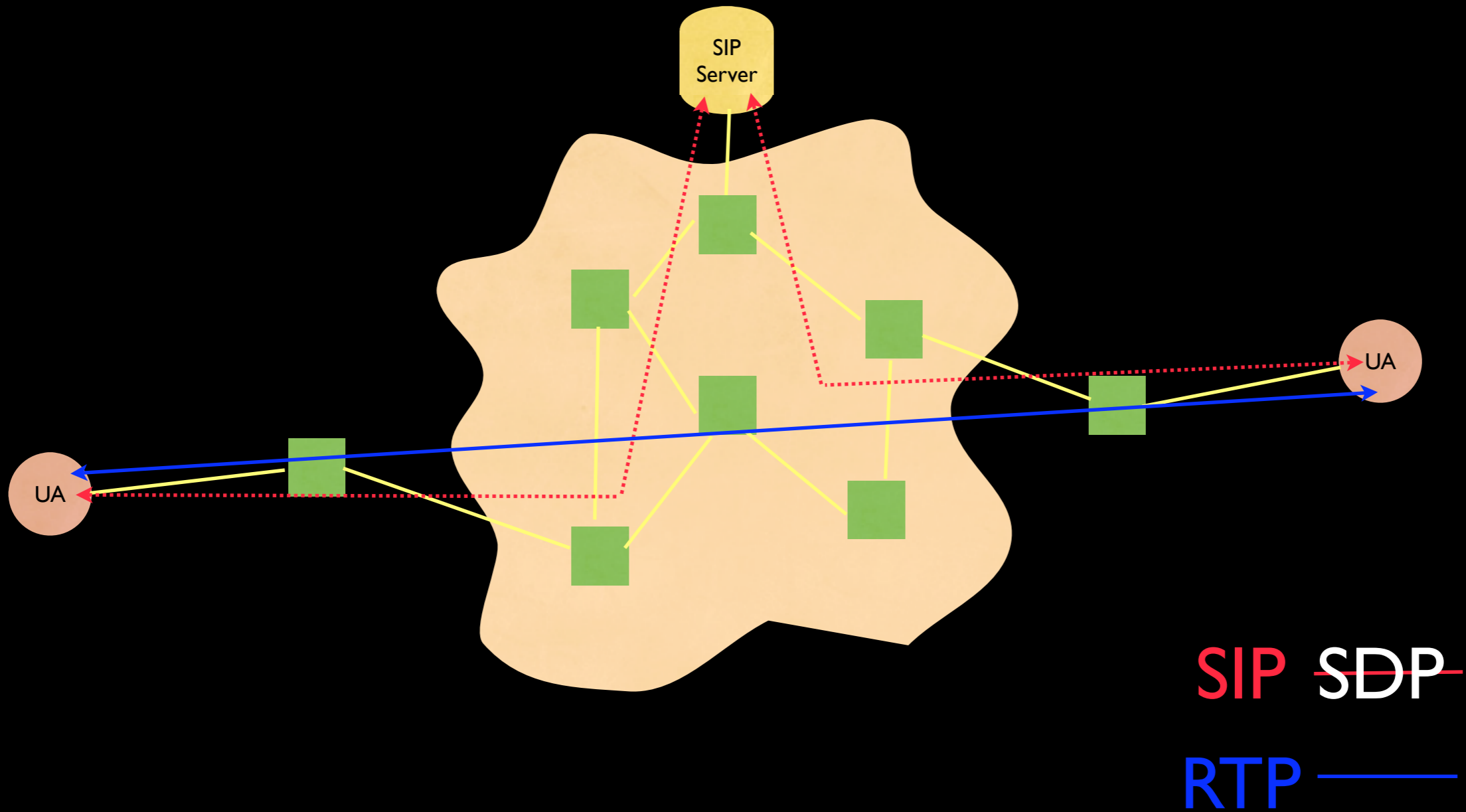
- A SIP server isn't technically required.
- But we always use it.
- Why? To find you.

# VoIP Protocols



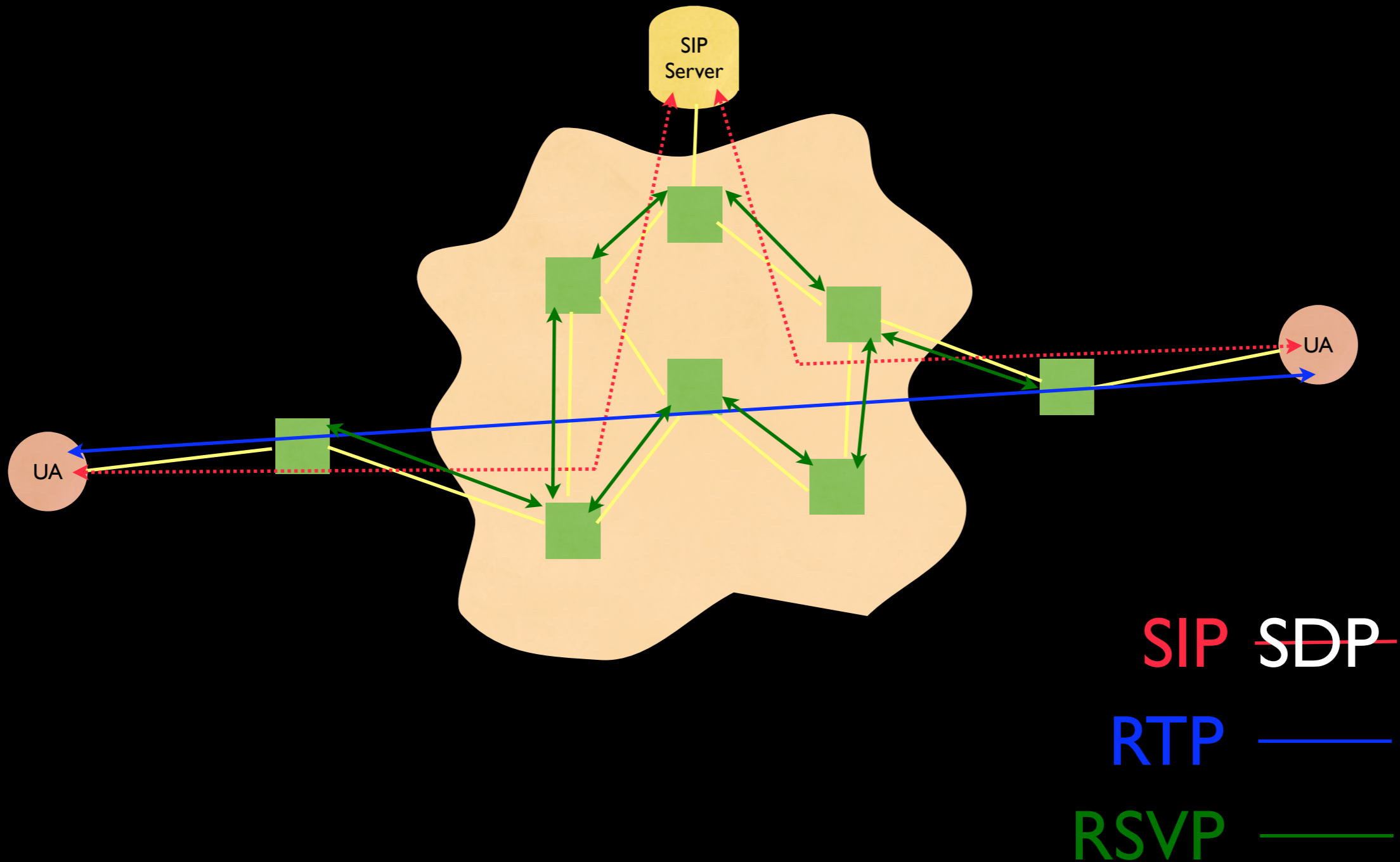
- SDP is carried as payload in the SIP INVITE and response

# VoIP Protocols



- RTP may not follow the same path as SIP

# VoIP Protocols



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