

SIP Basics and Beyond

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What is SIP (Session Initiation Protocol)

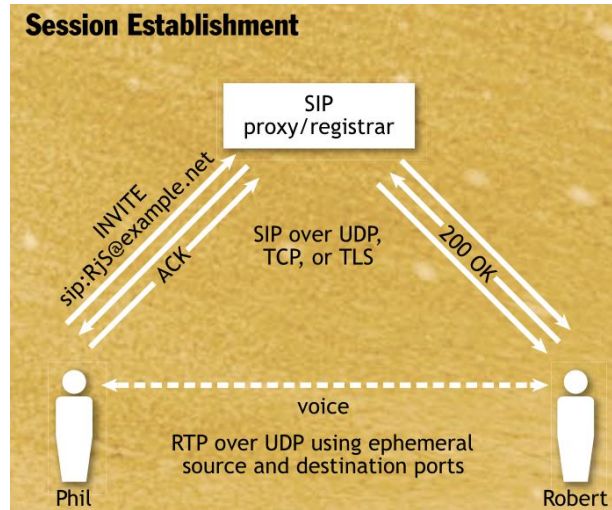
- In the traditional **circuit-switched** system, telephones were required to have essentially the same set of capabilities
- The mechanics of reaching them were based on their being at the end of a particular **fixed section of copper wire**

What is SIP (Session Initiation Protocol)

- In the **Internet-based** systems, the phone can have a wide range of capabilities. ex. codecs
1. It helps two parties wanting to communicate find each other on the Internet
 2. It allows those parties to negotiate how they are going to communicate

How SIP Works

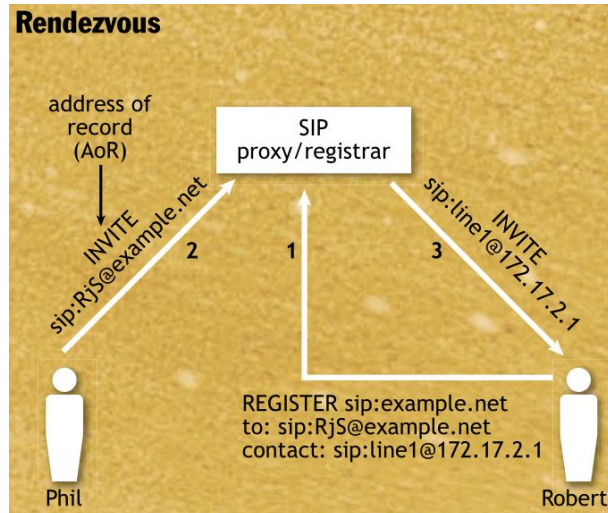
- SIP endpoints, also known as UAs (user-agents), can both generate and answer requests.
- A proxy/registrar is an endpoint when handling registration and an intermediary when forwarding requests



★ Note that the SIP signaling and the media traverse the network independently

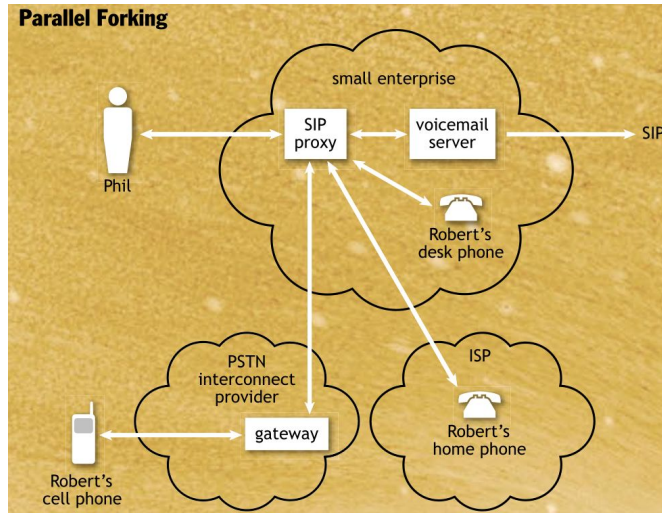
How SIP Works

- Rendezvous is established using these proxy/registrars and is based on users having a well-known AoR (address- of-record)



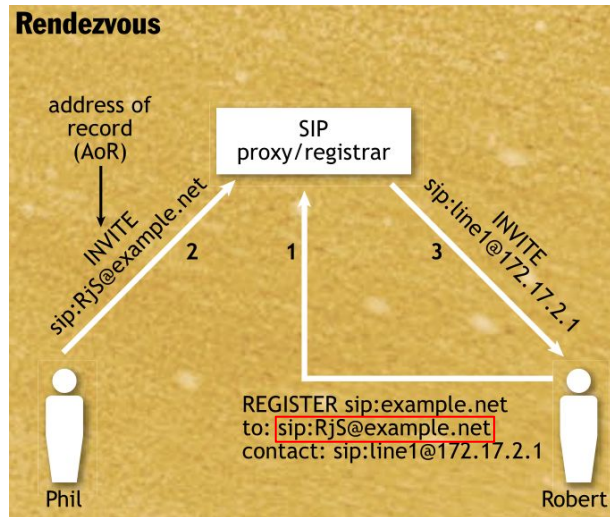
How SIP Works

- More than one endpoint can be registered with an AoR
- With many deployed services, it will simply forward the request to each contact simultaneously (parallel forking), which is a primary source of power in SIP's rendezvous function



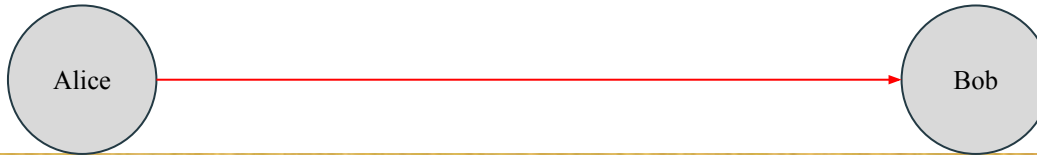
SIP URI (Uniform Resource Identifier)

- sip:RjS@example.net
- RjS: username
- example.net: DNS



SIP Message

- RURI (Request-URI) determines where the request is going to go



URIs in a SIP Message

request-URI (RURI) — INVITE sip:bob@biloxi.example.com SIP/2.0
to URI — Via: SIP/2.0/TCP pc33.atlanta.example.com;branch=z9hG4bK776asdhds
from URI — Max-Forwards: 70
contact URI — To: Bob <sip:bob@biloxi.example.com>
From: Alice <sip:alice@atlanta.example.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.example.com
CSeq: 293482 INVITE
Contact: <sip:alice@pc33.atlanta.example.com>
Content-Type: application/sdp
Content-Length: 138

SIP Message

- **To** header field was intended to carry whom the request was originally targeted toward (some proxy **retargeted** the **RURI** before it got to its final destination)
- **From** header field was intended to identify who sent the request
- In the **absence of extensions** to the protocol, they are best treated as opaque bits. opaque 不透明



URLs in a SIP Message

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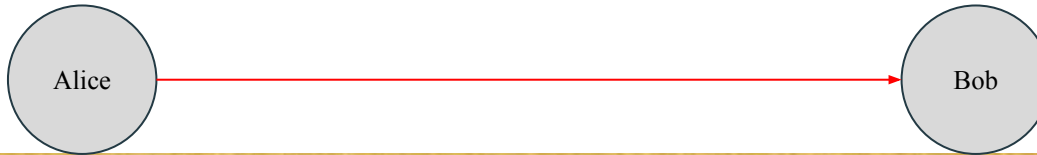
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SIP Message

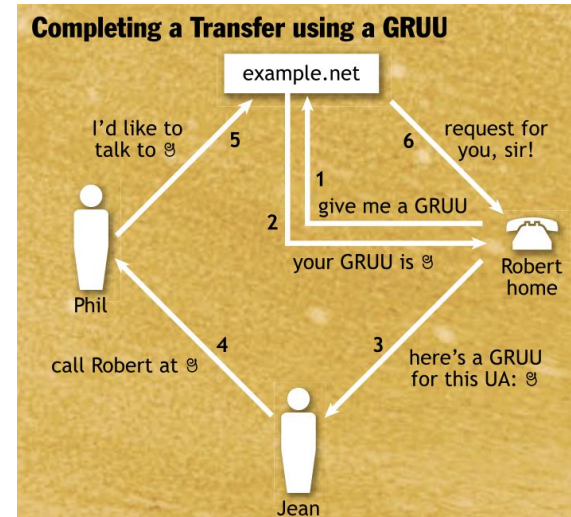
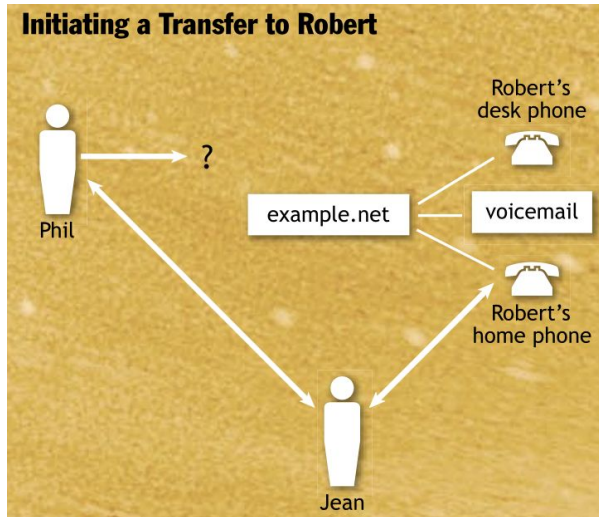
- Contact URI tells the recipient where to target future requests in the dialog this message might establish



URIs in a SIP Message

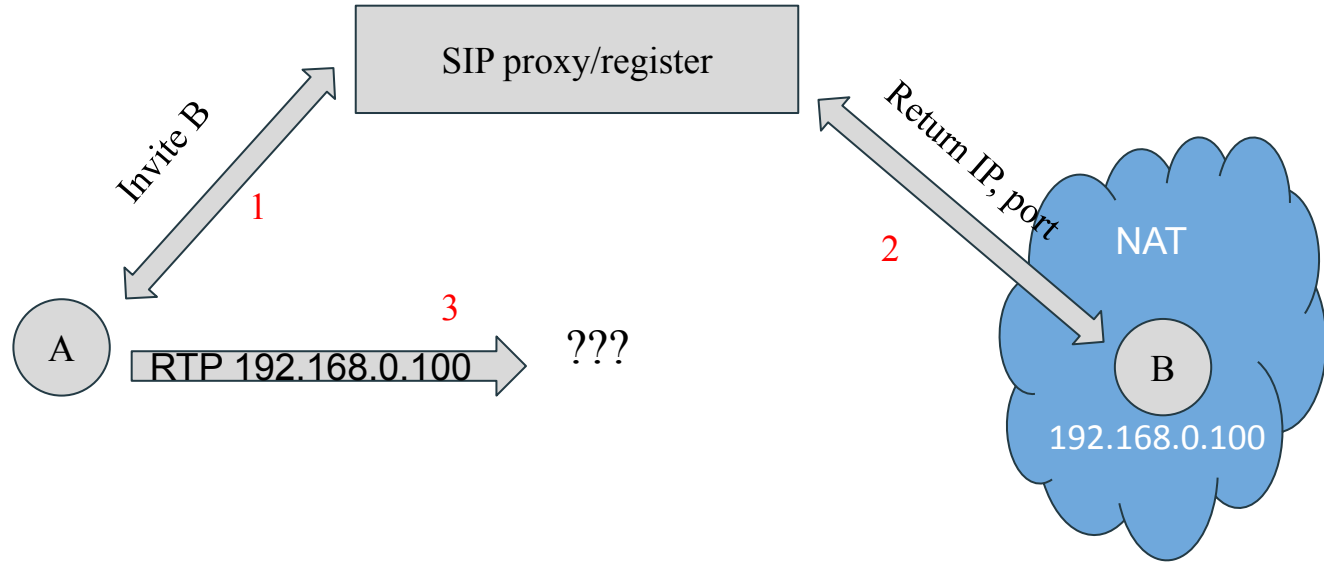
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GRUUs (Globally Routable User agent URIs)



Current Issues

Communicating through NATs



★ Solution: STUN, TURN, ICE

Current Issues

Emergency services

- Providers of emergency services in the traditional PSTN network. They used the property that the calling phone was at the end of a fixed, provisioned copper wire
- On the Internet, there is no such luxury for identifying the calling location
- In the IETF, this work is taking place in the ECRIT and GEOPRIV working groups

Thank you for your listening