Project5: Voice over Internet

A. Bagula, and Prof. A. E. Krzesinski

Department of Computer Science

University of Stellenbosch

7600 Stellenbosch, South Africa

bagula@cs.sun.ac.za

Overview

1. *IP telephony*
2. *Protocol requirements*
3. *Using SIP*

Why use IP telephony

- *Integration of transport: different businesses run on top of different technologies*
- *Cost savings: long distance call at the price of local calls*
- *Open service market: anyone with IP connectivity can become provider*
- *Programmability: split of transport and application services allowing each person who has connectivity to become provider*
- *Integration of applications: split of applications allowing for example IP to be operated by UUNET, SIP signaling to be operated by WCOM, PSTN call termination by mypshn.com*

Technical bariers

But there are technical barriers on

- *low level of reliability*
- *voice quality limited by the bandwidth limitations experienced by the best-effort delivery in the Internet*
- *Interoperability between the different speech recognition protocols*

How it works

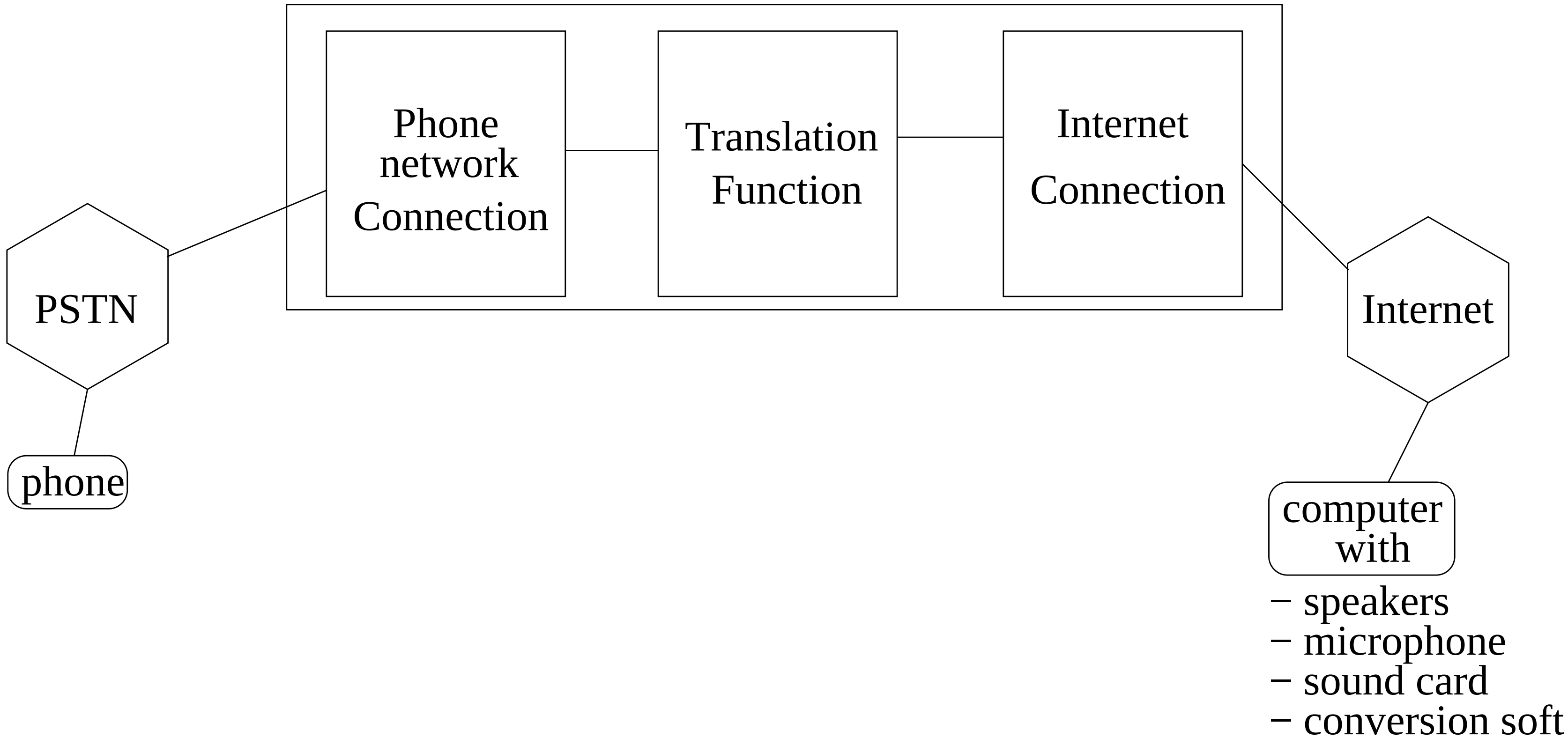

Figure 1: IP gateway

Protocol requirements

- *Signaling protocol (SIP): establish presence, locate users and initiate, update and tear down session*
- *Media (RTP) transport protocol (UDP, TCP): transmission of packetized audio/video*
- *Support protocol (RSVP, DNS, COPS): gateway location, QoS support, interdomain AAA etc...*

Understanding Session Initiation Protocol

- *Role: Initiate, update and terminate voice and video sessions across packet networks.*
- *Possible extensions for: call control services, mobility, interoperability with existing telephony systems and more.*
- *SIP entities: 4 types of logical entities, each entity having specific functions in SIP communication as a client and as a server: USER AGENT, PROXY SERVER, REDIRECT SERVER, REGISTRAR*
- *SIP messages: two types : requests from client to server and responses from servers to clients.*

Request messages

4 methods

- *INVITE: initiates a call, changes call parameters (re-INVITE)*
- *ACK: confirms a final response for INVITE*
- *BYE: Terminates a call*
- *CANCEL: cancels searches and “ringing”*
- *OPTIONS: queries the capabilities of the other side*

Response messages

- *REGISTER: registers with the location server,*
- *INFO: sends mid-session information that does not modify the session state*

Sip Session Establishment

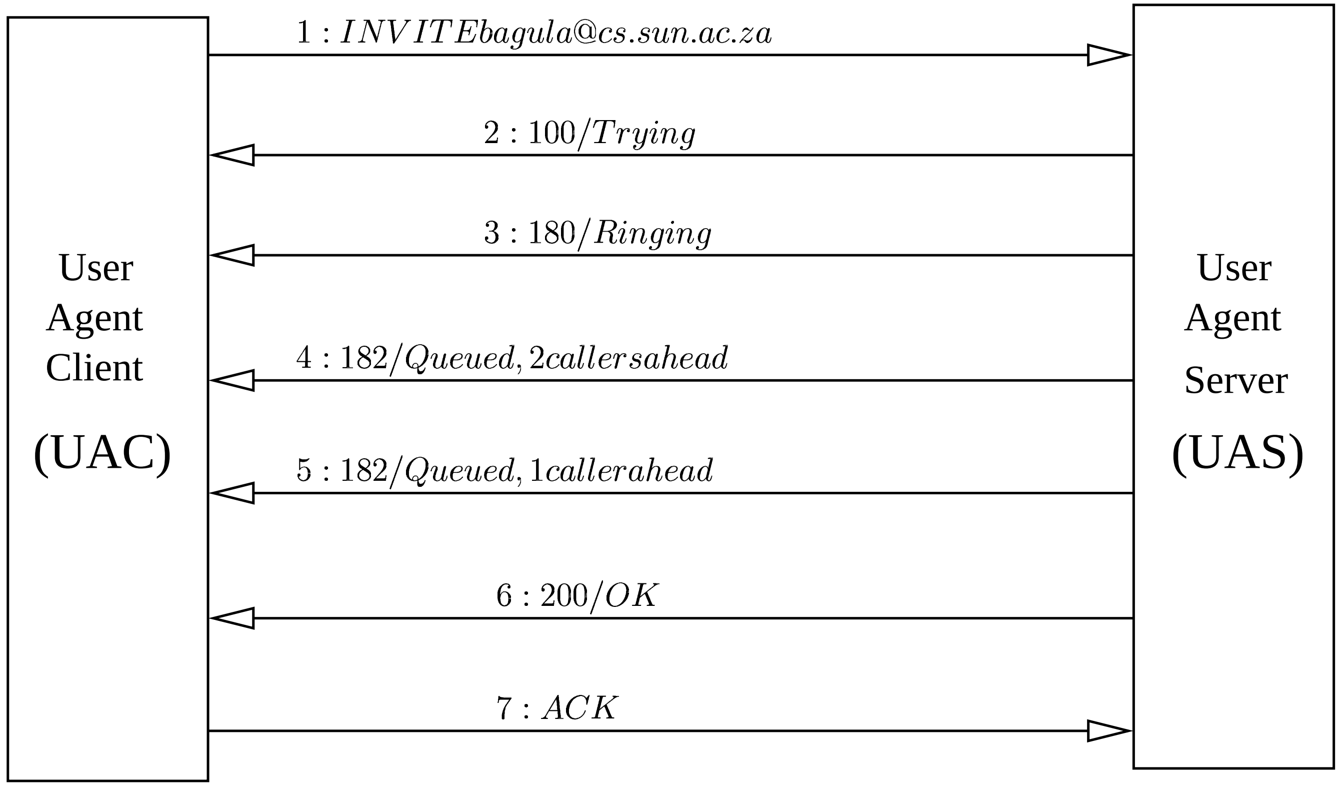

Figure 2: Sip Session Establishment

Sip Call Termination

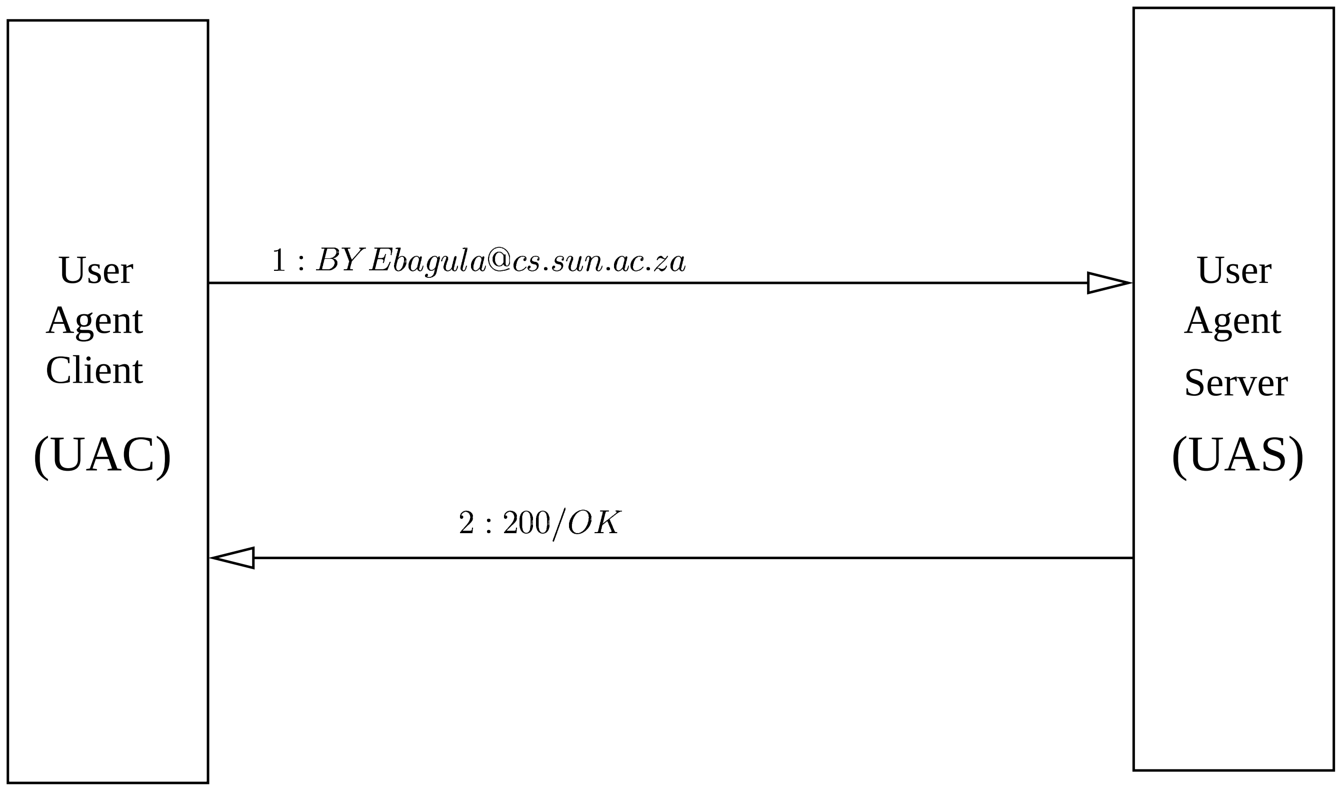

Figure 3: Sip Call Termination

Sip Proxy Server

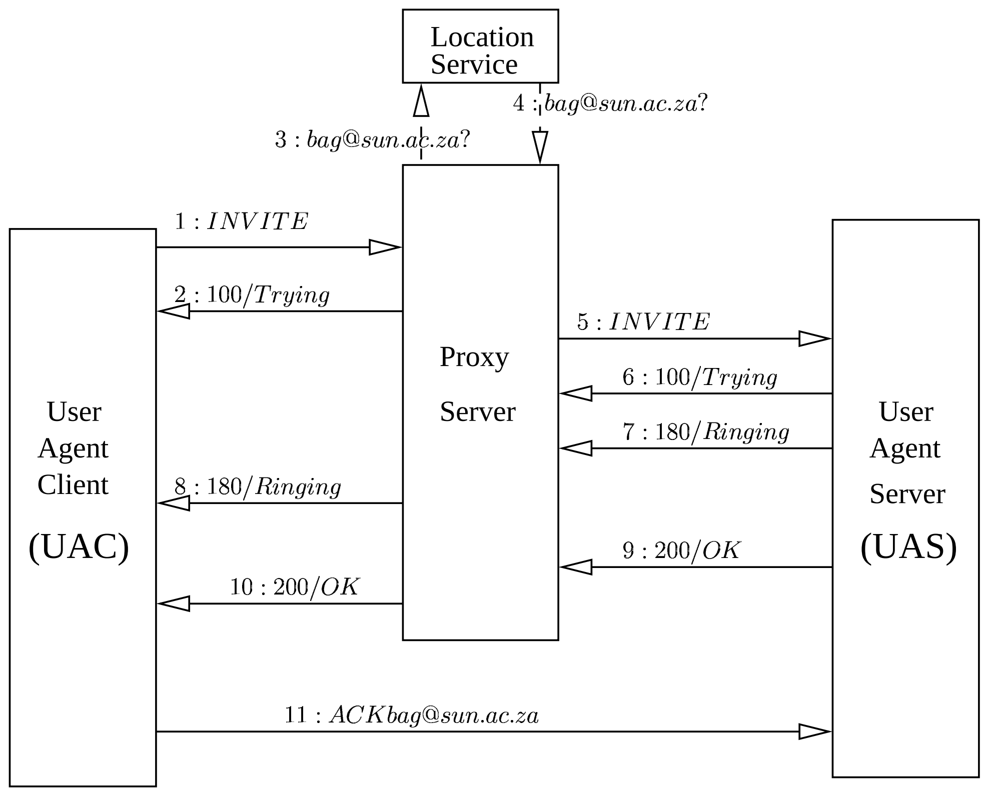

Figure 4: Sip Proxy Server

Sip Redirect Server

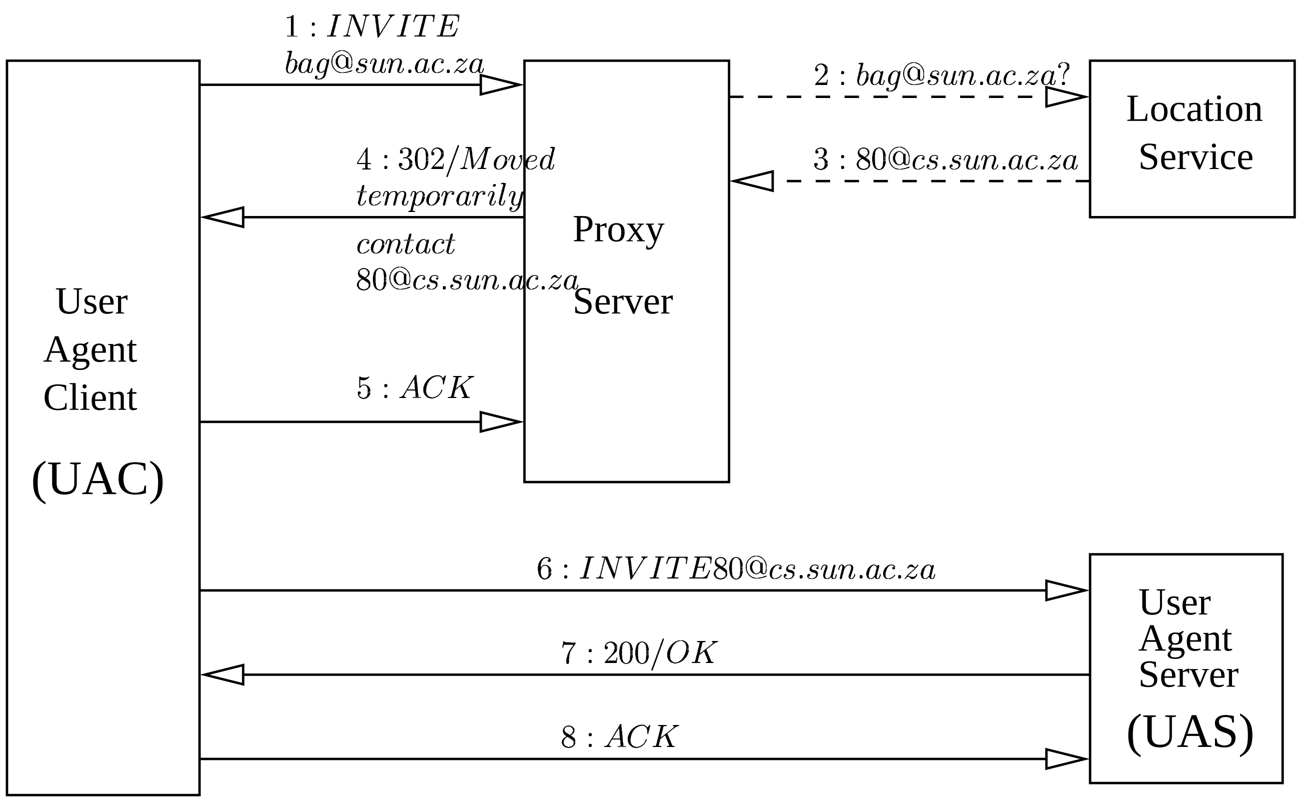

Figure 5: Sip Proxy Server