Asterisk RadIo Archetecture

VoIP Based Campus Announcment System

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Problem

A geographically large campus with many groups of students have to implement an announcment system.

Conventional System

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VoIP

VoIP *is a family of technologies, methodologies, communication protocols, and transmission techniques for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.*

- Wikipedia.org, Accesed February 2, 2012

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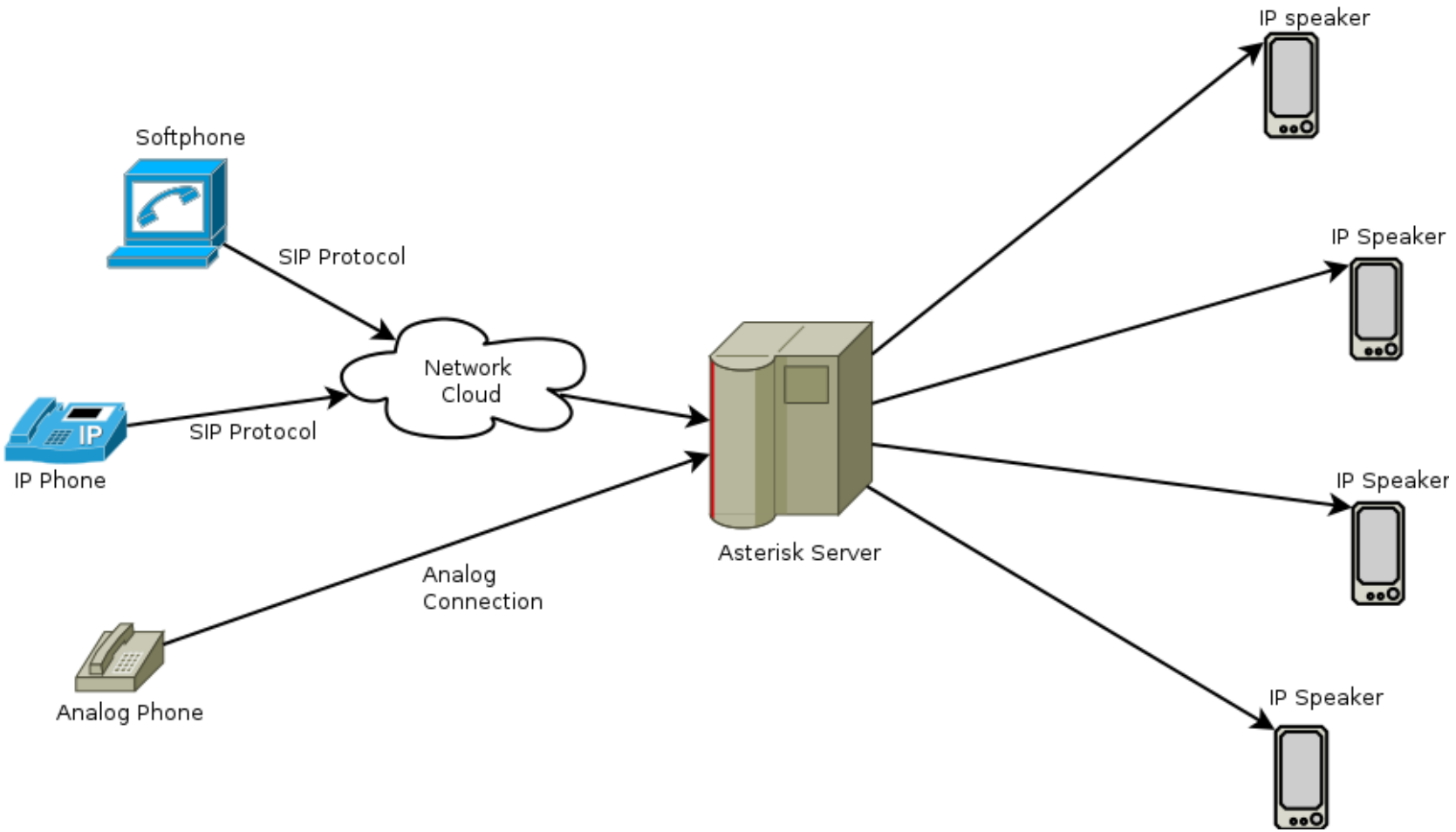
Products Currently Available

1. xyz

No Free Software products exist, Though almost all core components are available in a compatible licence.

Block Diagram



Protocols

1. The **Session Initiation Protocol (SIP)** is an IETF-defined signaling protocol widely used for controlling communication sessions such as voice and video calls over Internet Protocol (IP). The protocol can be used for creating, modifying and terminating two-party (unicast) or multiparty (multicast) sessions.

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2. **RTP** provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. *(RFC 3550)*

Asterisk

Asterisk *is a software implementation of a telephone private branch exchange (PBX); it was created in 1999 by Mark Spencer of Digium. Like any PBX, it allows attached telephones to make calls to one another, and to connect to other telephone services including the public switched telephone network (PSTN) and Voice over Internet Protocol (VoIP) services. Its name comes from the asterisk symbol, *.*

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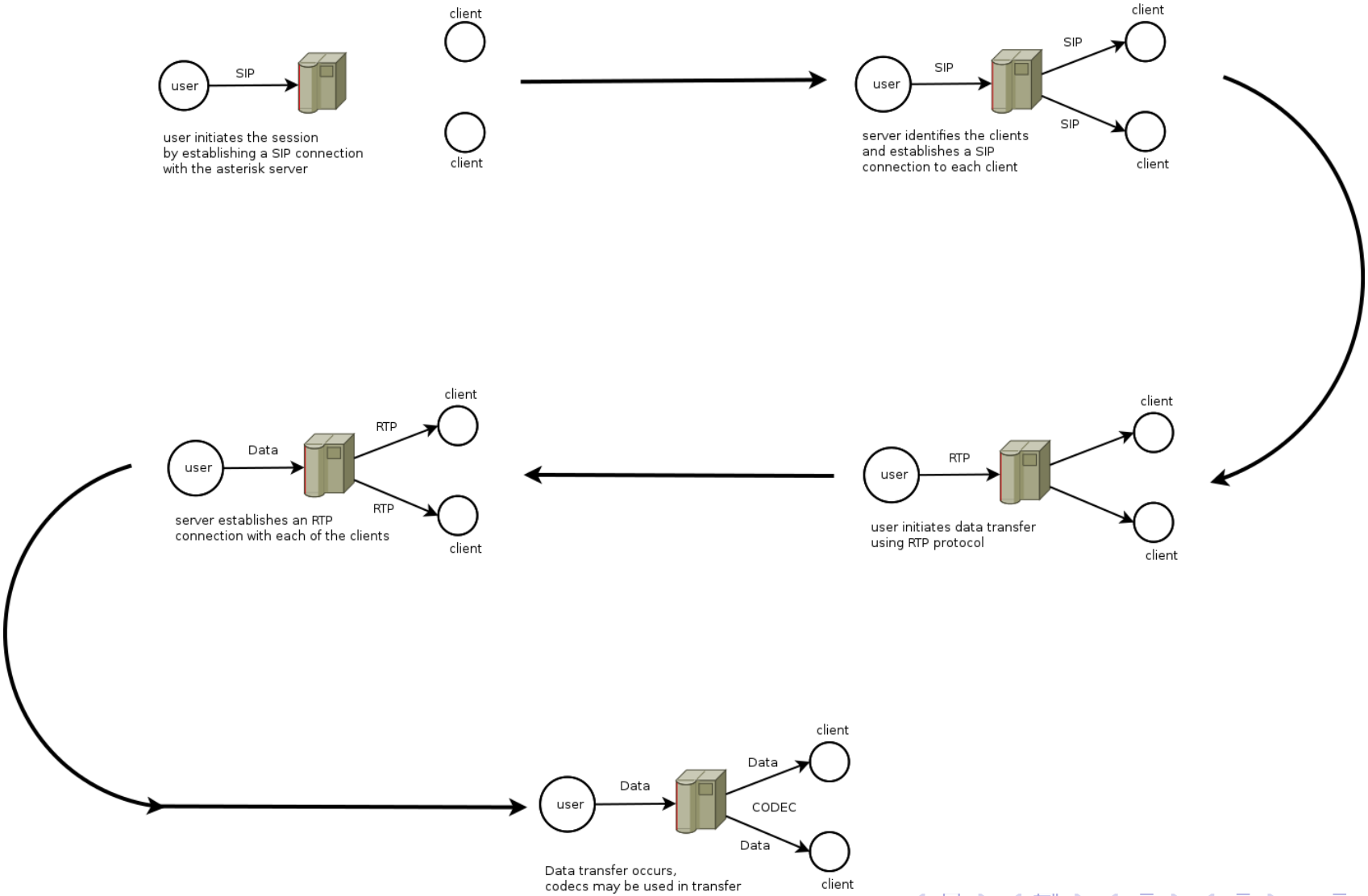
Asterisk thus essentialy can act as a SIP proxy for routing the IP multicast trasport we needed to implement.

Working


client
user
SIP
user initiates the session
by establishing a SIP connection
with the asterisk server
client
client
user
SIP
SIP
SIP
server identifies the clients
and establishes a SIP
connection to each client
client
client
user
RTP
user initiates data transfer
using RTP protocol
client
client
user
Data
RTP
RTP
server establishes an RTP
connection with each of the clients
client
client
user
Data
Data
CODEC
Data
Data transfer occurs,
codecs may be used in transfer
client
client

Challenges

1. Development of software for transmission and reciever.
2. Development of a streamlined aproach for configuring Astrisk PA System.
3. Implementation and Testing.

Expenditure

- Consumables
  - Network equipments Rs. 1500
  - Import charges on equipment Rs. 7000
  - Misc Charges: Rs. 1000
- Equipments
  - IP Phone Rs. 5000
  - IP speakers x2 Rs. 10000
  - Digium FXO cards - 1TDM410PLF Rs. 10000
- Research Literature - Rs. 3000
- Others
  - Uplink to telephony provider to test remote link. (college PBX)
- Contingencies Rs. 1000.
  - Rs. 4000 in case IP speakers are not available.