Asterisk RadIo Architecture

VoIP Based Campus Announcement System

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Outline

Problem

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

Conventional System

1. **No Flexibility** in selection of audience.
   No way of communicating with a part of audience.
2. Requires **heavy cabling** around the campus.
3. Limited **scalability and extendability**.
   - *System cannot grow beyond a point, power and load problems.*
   - *No scope of extending the system further. (Things like video, two way paging etc.)*

Use an existing network!!

VoIP

Voice Over Internet Protocol *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, Accessed on February 1, 2012

Advantages

1. Can utilize the existing IP network in campus.
   - A major cost in implementing the system can be made practically nil.
2. Each client can be addressed individually.
   - User can create groups to address on-the fly.
3. Scalable - Adding a new client is simple as long as there is network connectivity
   - No load problems.
4. Have an option to build advanced systems.
   - Video transmission.
   - Two way communication systems.
5. Provision for remote access.
6. Easy modification of system - No hard circuits.

Products Currently Available

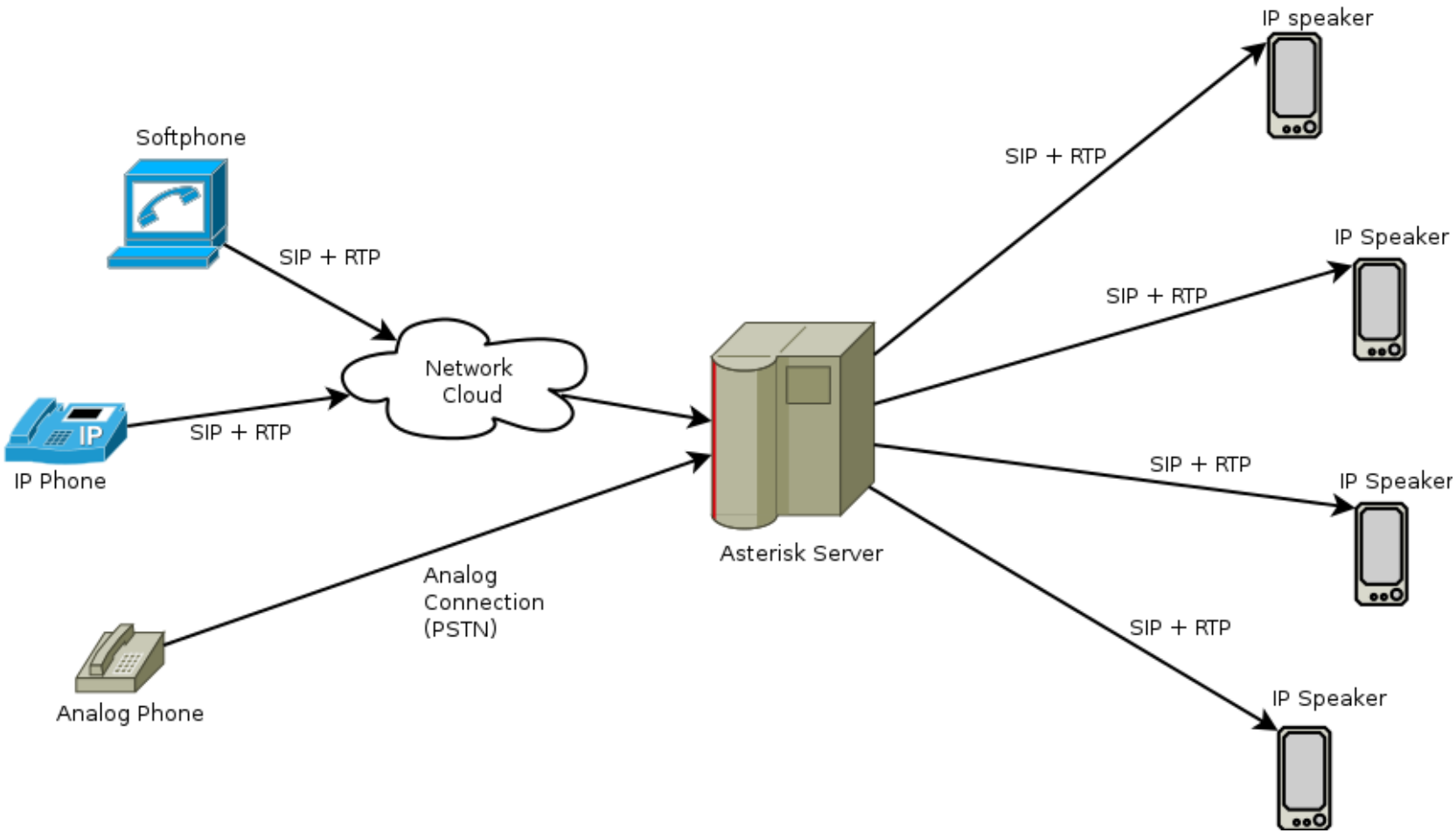
1. LanTone Systems. - `http://www.voip.com.sg/voip-products/ip-pa-system.html`
2. AbleTEK IP-PA System - `http://www.abletek.co.uk/ip_public_address.php`
3. TalkMaster System - `http://www.digac.com/ii3_talkmaster.htm`

All these products cost ~**$1500** for the software itself, and comes with a minimum **device purchase limit** and **no inter-operability**.
No Free Software products exist, Though almost all core components are available in a compatible license

ARIA

Asterisk **R**ad**I**o **A**rchitecture

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.
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)*

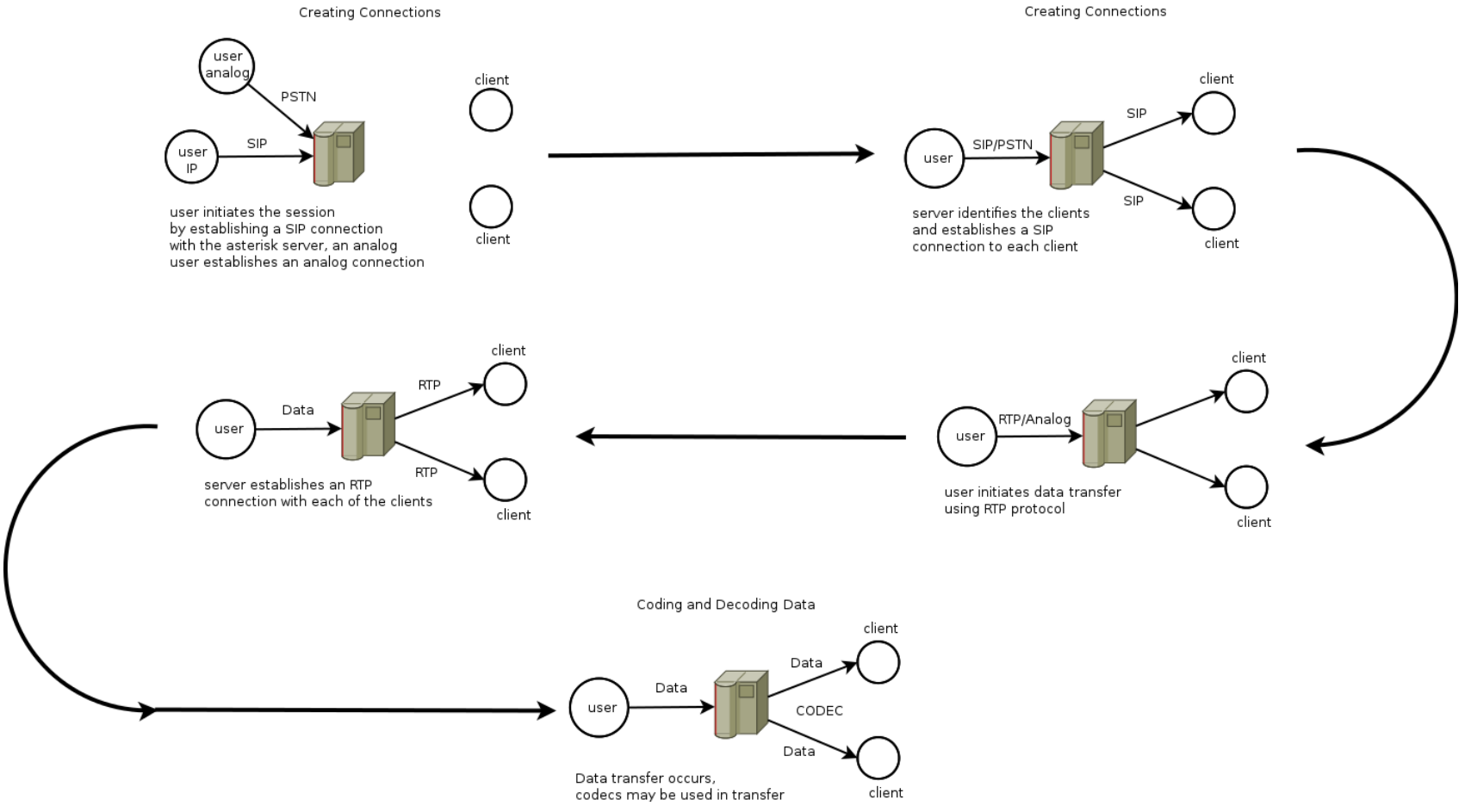
Asterisk

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, *.*

- Wikipedia.org. accessed February 3, 2012

Asterisk thus can act as a proxy for routing the **IP multicast transport** we needed to implement.

Working


Creating Connections
user analog
PSTN
user IP
SIP
client
client
user initiates the session
by establishing a SIP connection
with the asterisk server, an analog
user establishes an analog connection
Creating Connections
user
SIP/PSTN
SIP
SIP
client
client
server identifies the clients
and establishes a SIP
connection to each client
user
RTP/Analog
client
client
user initiates data transfer
using RTP protocol
user
Data
RTP
RTP
client
client
server establishes an RTP
connection with each of the clients
Coding and Decoding Data
user
Data
Data
CODEC
Data
client
client
Data transfer occurs,
codecs may be used in transfer

Challenges

1. Development of software for transmission and receiver.
2. Development of a streamlined approach for configuring Asterisk PA System.
3. Implementation and Testing.

Expenditure

- Consumables
  - Network equipment Rs. 1500
  - Import charges on equipment Rs. 7000
  - Misc Charges: Rs. 1000
- Equipment
  - IP Phone Rs. 5000
  - IP speakers x2 Or Analog Gateway+ Speakers 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.

Total Cost: Rs.42500/-
Real World Implementation: Add cost of each client needed.

Conclusion

1. Provides easy and streamlined approach to install, configure and manage a system of any size.
2. Uses open systems and protocols wherever possible.
3. The system can be accessed remotely.
4. Only **Open Source** Final Product.

References

1. Imran, A.; Qadeer M.A.: Conferencing, Paging, Voice Mailing via Asterisk EPBX, 10.1109/ICCET.2009.209
2. Montoro, P.,Casilari, E. -A Comparative Study of VoIP Standards with Asterisk 10.1109/ICDT.2009.8
3. Subramanian, S.V.; Dutta, R.; Comparative Study of Secure vs. Non-secure Transport Protocols on the SIP Proxy Server Performance: An Experimental Approach 10.1109/ARTCom.2010.90