Inter-working of P2P-SIP and Traditional SIP Network

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40 ans la révolution de l'information

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Outline

- Introduction
  - What is P2P-SIP?
  - Why P2P-SIP?
  - Different Approaches of P2P-SIP
- Inter-working of P2PSIP & Traditional SIP
- Integration in EMANICS VoIP testbed
- Conclusion

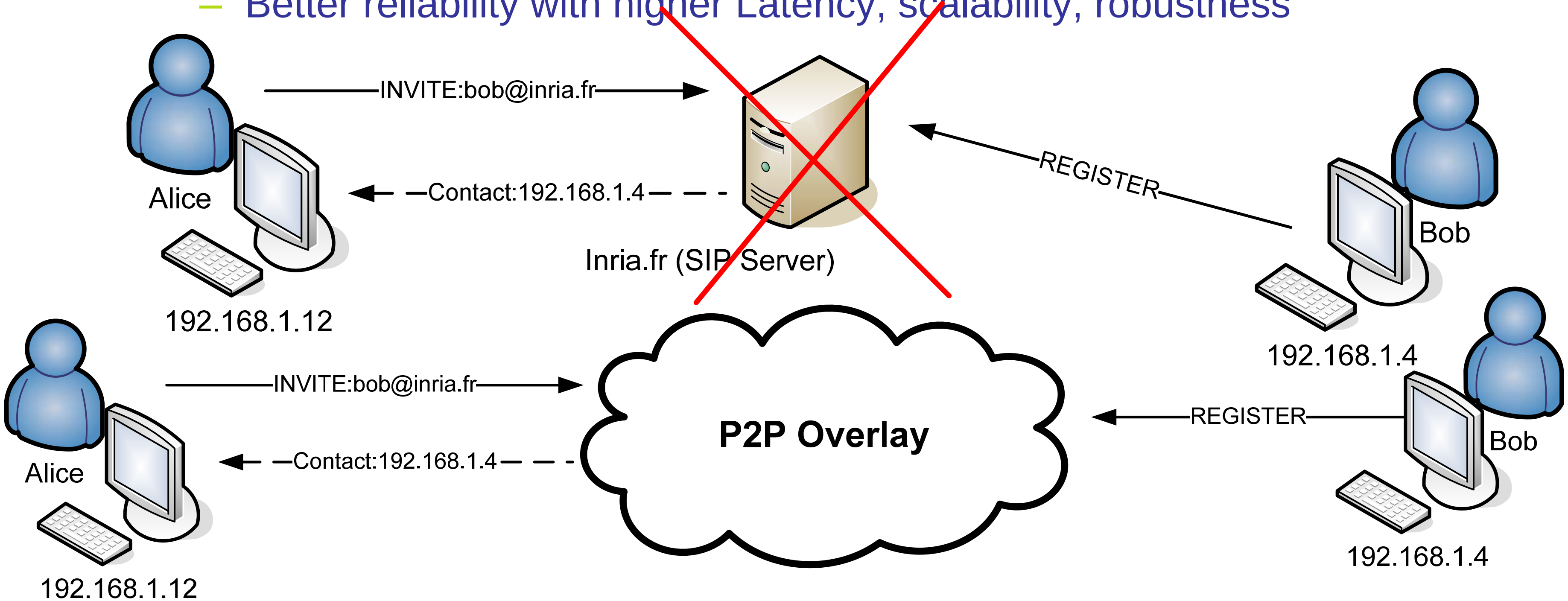
Introduction

•What is P2P-SIP ?

- Use the P2P techniques (like DHT) to resolve the SIP related functions
- Leveraging the advantages of distributed network (P2P) for SIP network
  - Reliability, scalability etc.,

•Why P2P-SIP ?

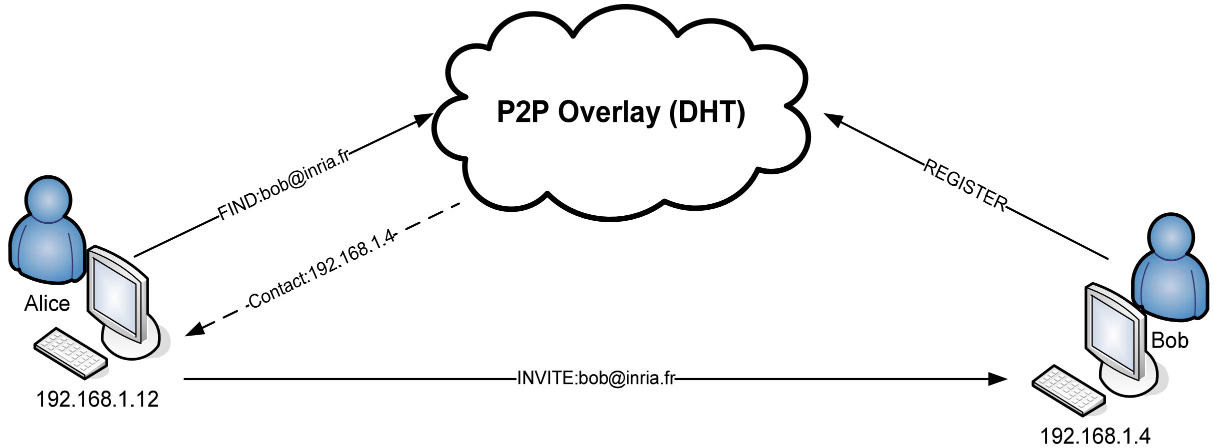
- NO centralized server
  - Lower Maintenance cost, configuration
  - Better reliability with higher Latency, scalability, robustness



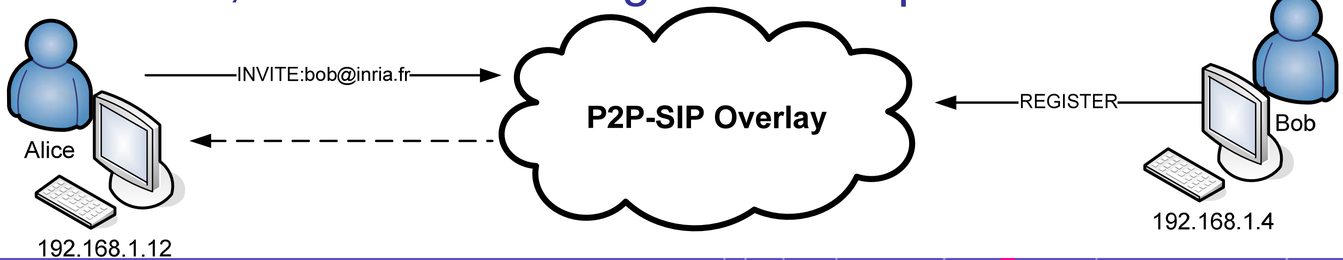
Introduction (cont..)

Different Approaches of P2P-SIP

- SIP using P2P
  - Replace the SIP location service with a P2P protocol



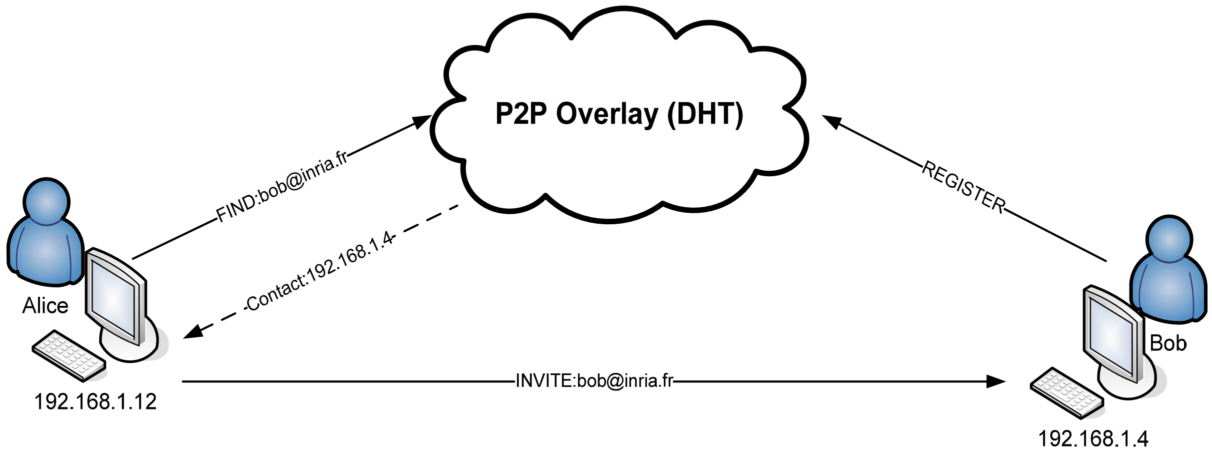
- P2P using SIP
  - Implement P2P using SIP messaging
    - i.e., Use SIP messages to transport P2P traffic



Introduction (cont..)

How is P2P-SIP works?

- Replace the SIP location service with a P2P protocol
- SIP URI are used for Hash key generation in DHT.
  - Eg.,Sip:bob@test.bamboo => Hash(Key, identifier)
- Communication between the P2PSIP client and DHT uses P2P protocol



Inter-working of P2PSIP and Traditional SIP networks

- **Problems**
  - **Finding the exact location of the clients (both P2P-SIP and conventional)**
    - Because P2P-SIP domain uses overlay identifier rather than DNS entries.
    - Whereas conventional SIP uses DNS to locate domains
  - Solution
    - Need a gateway (proxy) kind of approach
- **Different Approaches**
  - P2P-SIP Peer can act as Standard SIP Proxy and Registrar for conventional SIP UA
    - The peer can insert the UA in the distributed database
    - Can retrieve contact information from the distributed database when proxying

Inter-working of P2PSIP and Traditional SIP networks (cont.,)

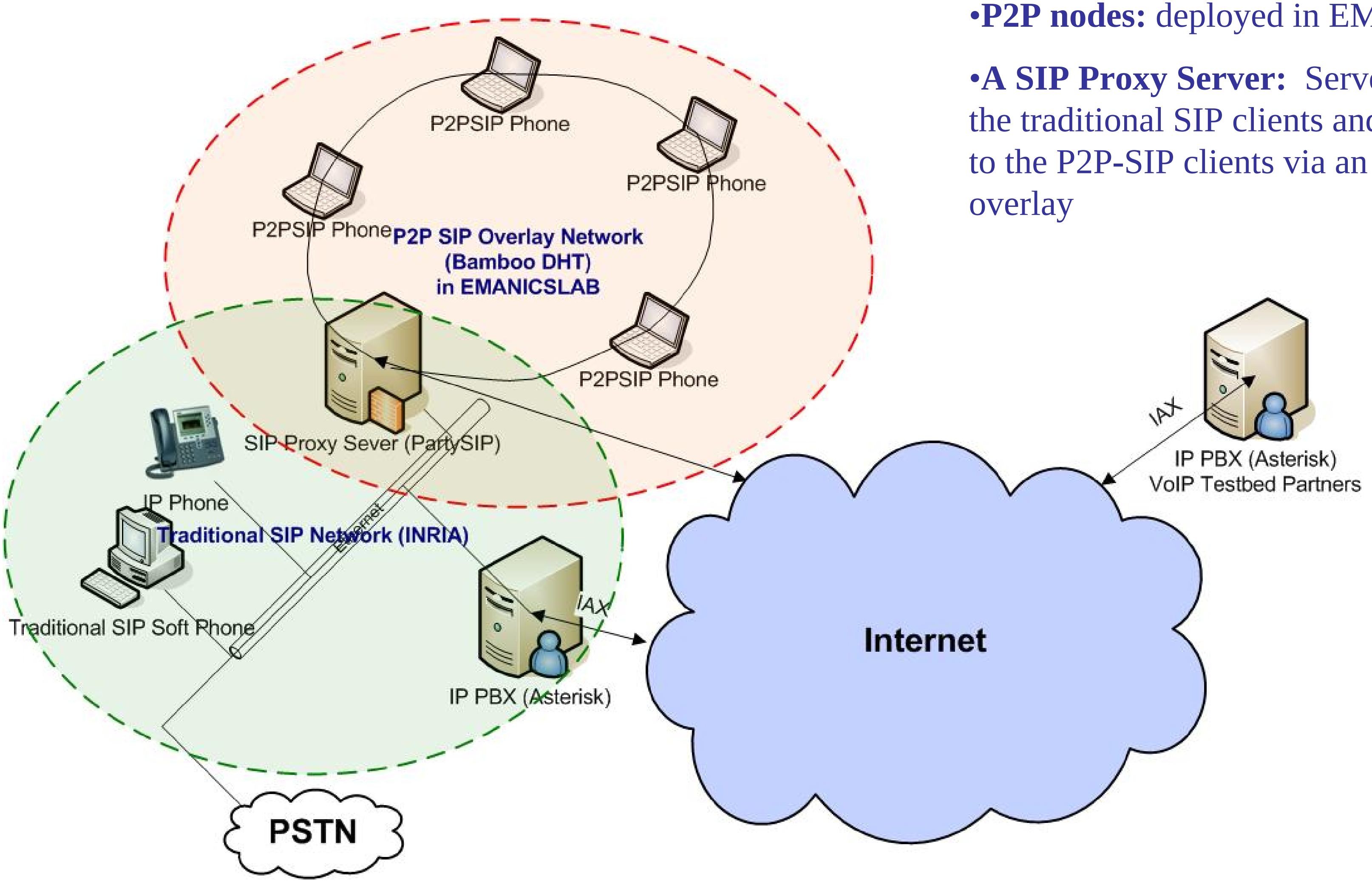
•Different Approaches.

- Act as a SIP Proxy for the overlay
  - P2P-SIP peer has a fully qualified domain name (FQDN) that matches the overlay name .

- **Use Hierarchical overlay architecture**
  - **Has two level of overlays (Higher and Lower)**
  - **Higher level overlay will serve as the gateway for the lower level**
    - **i.e., the higher level overlay will be connected to the internet and lower level is just a P2P-SIP overlay**

P2P-SIP in EMANICS Testbed

P2P-SIP Architecture

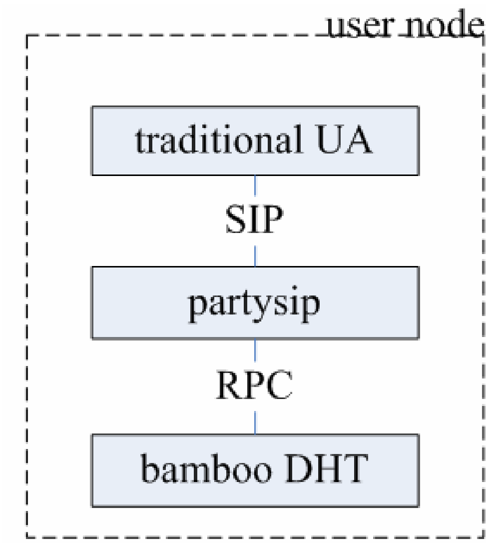


Components

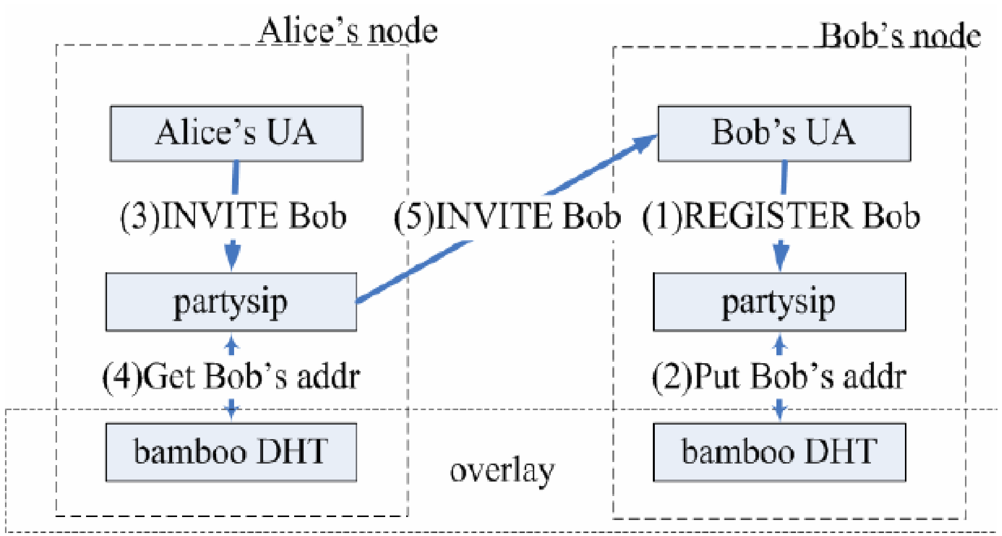
- **P2P nodes:** deployed in EMANICSLAB
- **A SIP Proxy Server:** Serves as a Proxy for the traditional SIP clients and also connects to the P2P-SIP clients via an upper level overlay

P2P-SIP in EMANICS Testbed (cont.,)

- **How is it deployed?**
  - We use an existing open source implementation called Olyo P2P-SIP based VoIP system (http://code.google.com/p/olyo/)
  - Uses Bamboo DHT
  - A P2P-SIP adaptor (Modified PartySIP)
    - connects the SIP clients and P2P overlay



Internal Structure of P2P-SIP user node



Working of REGISTER and INVITE of two nodes in the P2P-SIP overlay

Conclusion

- About P2P-SIP
  - P2P-SIP is still in the beginning stage of its development and needs a standard architecture specification for interworking with traditional SIP networks.

- Some open Issues (Research to be done)
  - How do we connect many domains in the overlay?
  - Should we register one domain in many overlays ?
  - Define User management or buddy list?
  - How are the authentication, authorization, accounting to be handled?
  - Provide traditional services like Voicemail, conferencing, emergency services, etc.,
  - P2P-SIP security (DoS attacks, etc.,)

Thank you

Any Questions?

1st EMANICS Peer-to-Peer Management Workshop

Zurich, march 2008

Peer-to-Peer SIP

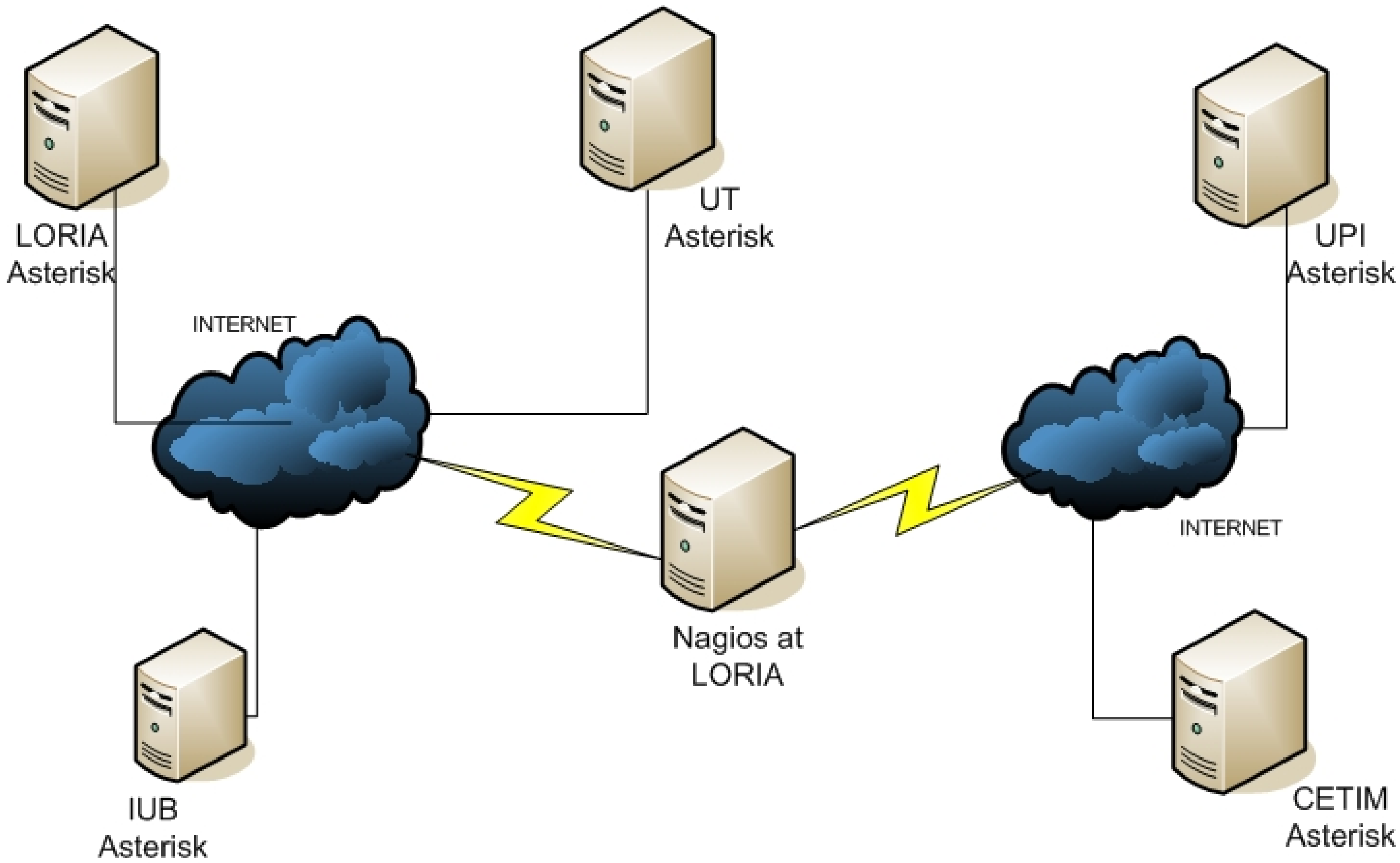
Implementation and monitoring

UPI Research Team

Existing infrastructure

- VoIP testbed based on Asterisk servers
- servers are interconnected through IAX
- system is hierarchical and secured
- authentication is needed to join in
- latency is low, voicemail and conferencing are active; PSTN connectivity is enabled

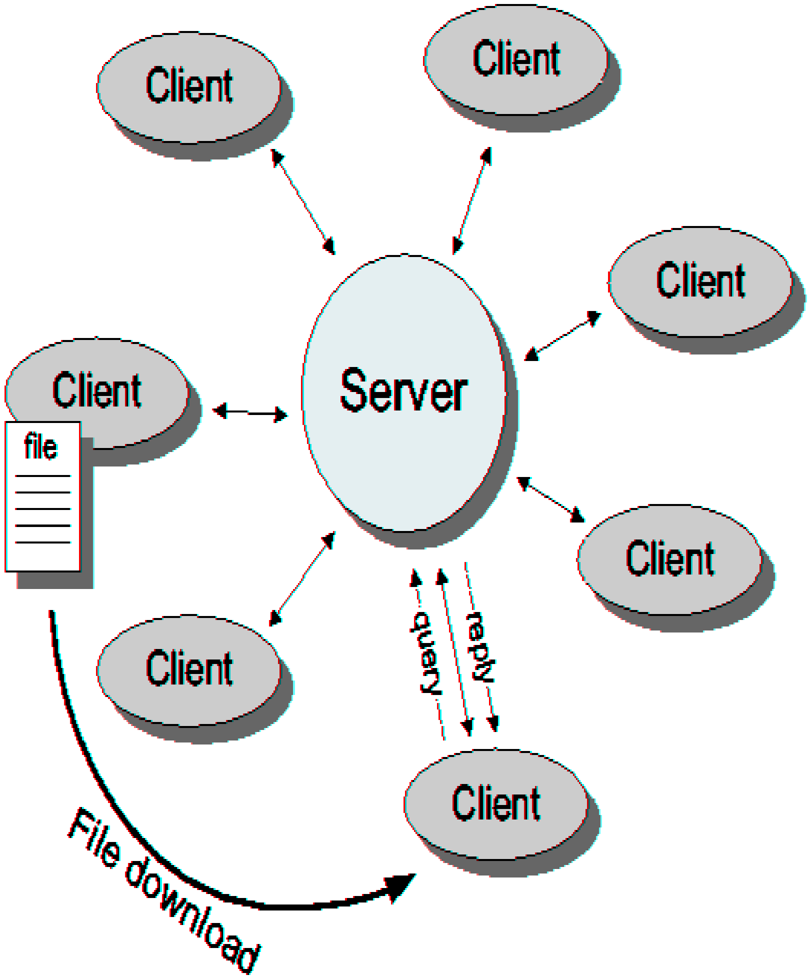
Map of the VoIP testbed

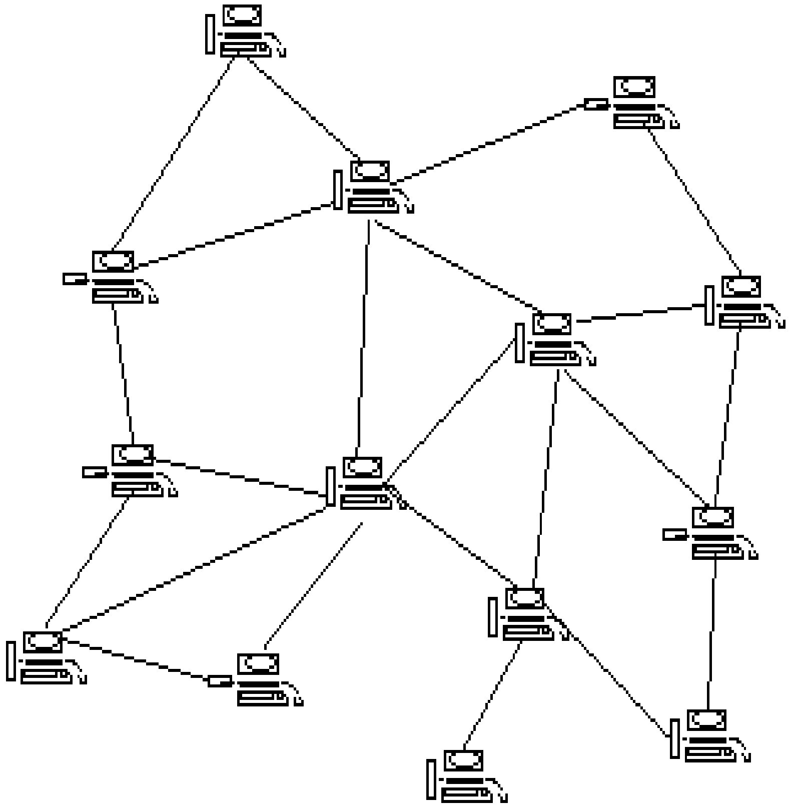


Peer-to-peer architectures

- P2P is not new, file sharing services are using it for quite some time with success
- every user installs a copy of the network's client and it announces the others of it's presence, location and shared content when connecting
- data flows directly form one peer to another
- system is reliable and scalable, without any server being a single point of failure

Client/server vs P2P





Commercial solution

- P2P + Voice = Skype
- free for now, but closed source and proprietary protocols, highly encrypted; no compatibility
- hybrid system, uses Global Index Server for authentication, friends list and user lookup
- nodes with enough resources and bandwidth become super-nodes, routing traffic (act as proxy) for users behind NAT and firewalls

P2P SIP architecture (1)

- SIP used as communication protocol between peers, and an existing P2P (preferably OSS) protocol for user registration (REGISTER) and lookup;
- a ring-structured DHT like Chord or Bamboo reduces latency to *O(log(n)),* because every peer keeps information about log(n) of its neighbors
- a SIP ID (sip:user@domain) is associated with a hash key, based on the client's IP address
- security may be an issue: privacy is not insured (call logging), and DoS attacks are possible
- solution: a PKI combined with a web of trust gives results

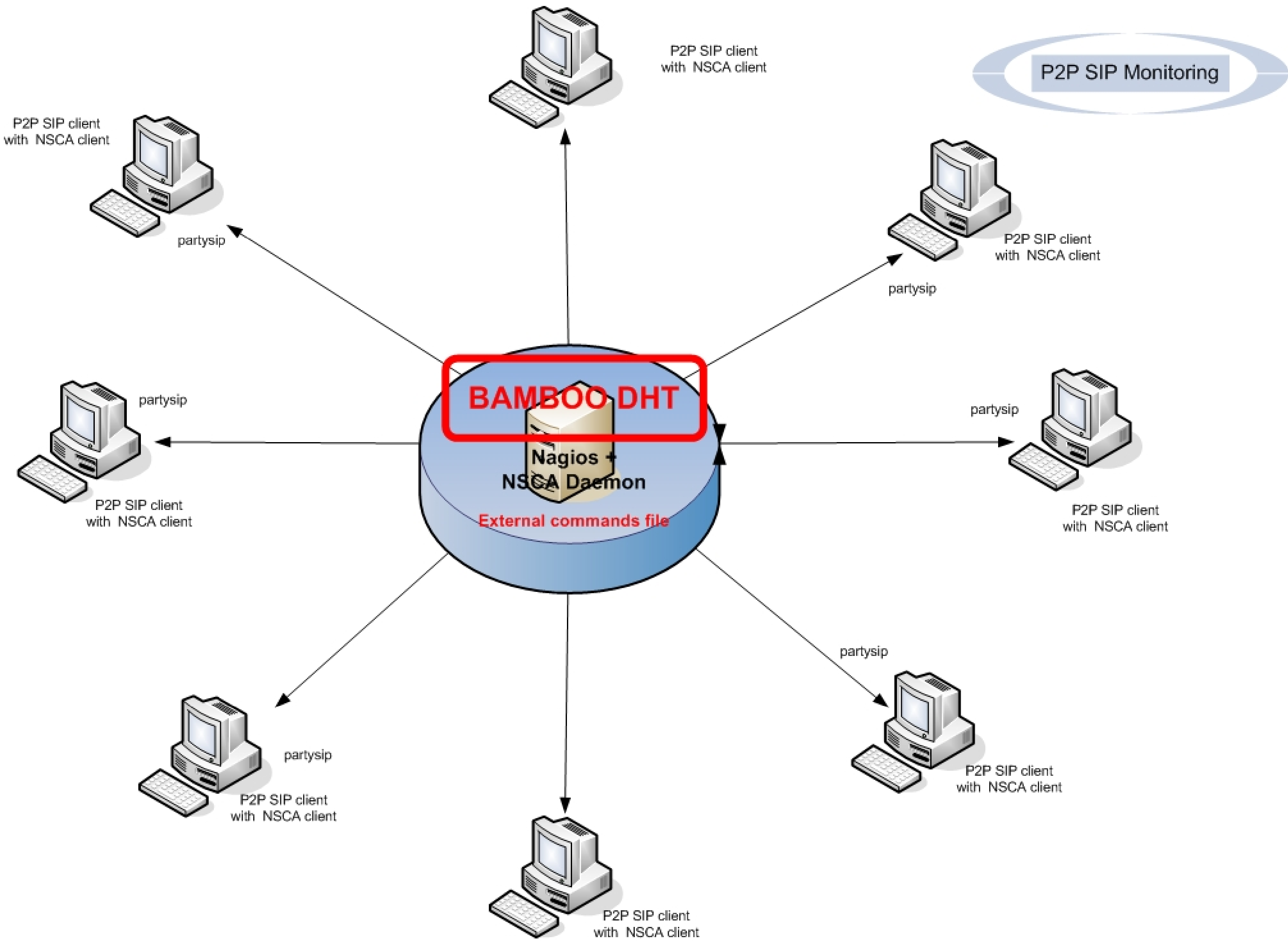
P2P SIP architecture (2)

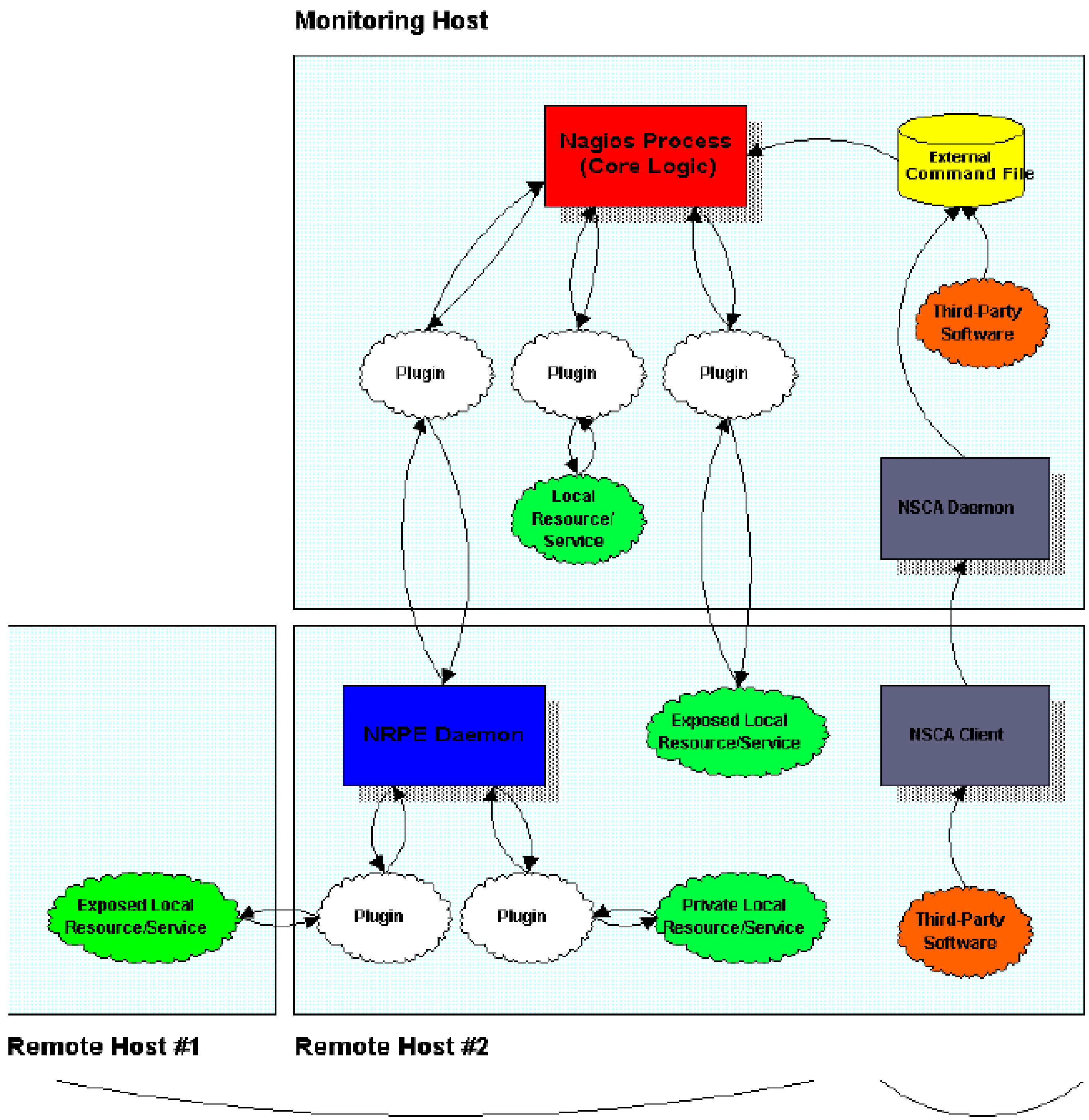
- reliability: when a peer or a super-node shuts down, its absence is detected by the REFRESH messages that remain unanswered and Chord structures of the neighbors are updated accordingly
- voicemail can be stored either at the sender, either on a super-node and delivered later, when the client connects
- a super-node can become a mixer for multi-party conferencing between its clients
- ICE can be used for NAT traversal, as every peer has integrated STUN and TURN server
- Tested implementations: Olyo and P2P SIP Monitor

Monitoring P2P SIP

- **Nagios**: widely used open-source monitoring platform, running on Linux as a *daemon*
- open, plug-in-based architecture, with a web-based GUI for graphs and reports
- periodically launches plug-ins against selected hosts (**defined IPs**) to check service availability
- in case of failures, it can automatically alert administrators by e-mail, Jabber or even SMS
- has the ability to automatically run error-correcting scripts without intervention, by *SSH*

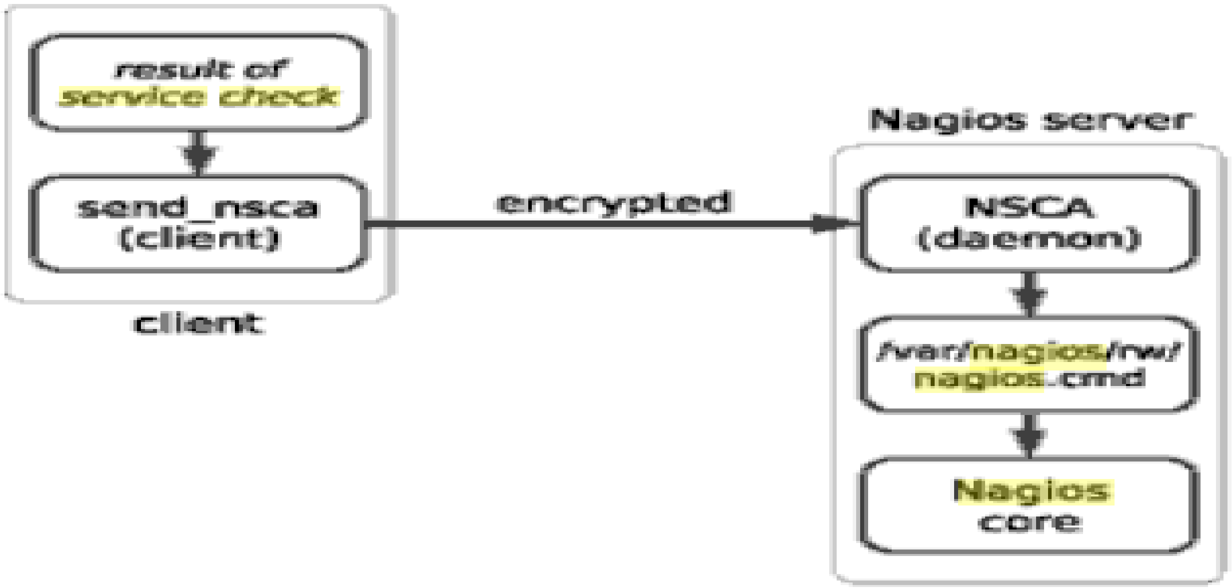
P2P monitoring diagram



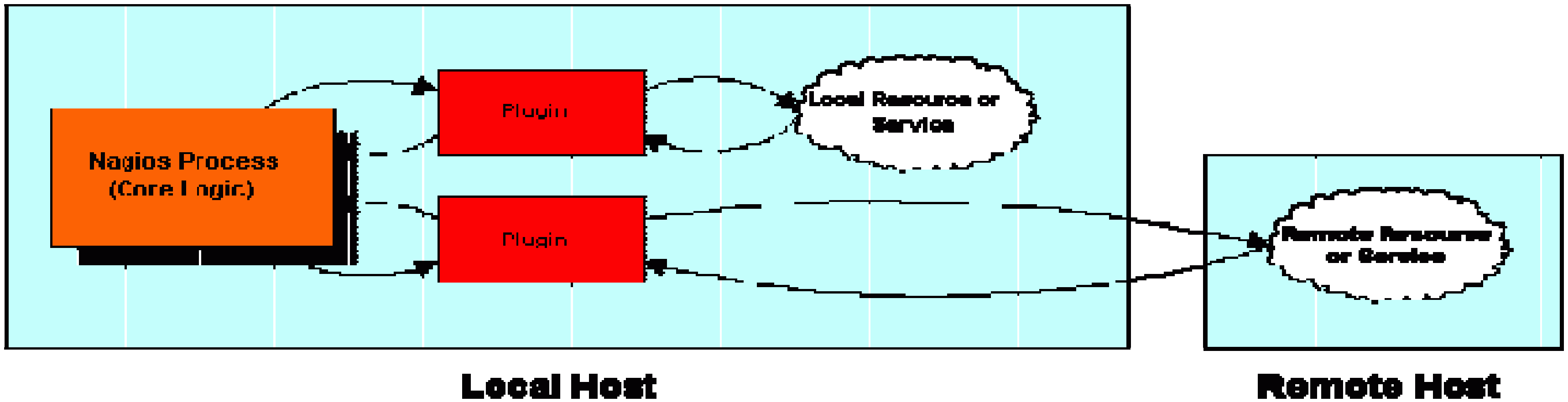


Monitoring Host
Nagios Process (Core Logic)
External Command File
Plugin
Plugin
Plugin
Third-Party Software
Local Resource/ Service
NSCA Daemon
NRPE Daemon
Exposed Local Resource/Service
NSCA Client
Exposed Local Resource/Service
Plugin
Plugin
Private Local Resource/Service
Third-Party Software
Remote Host #1
Remote Host #2

NSCA Communication



check_p2p plug-in for Nagios



- Perl script that sends an OPTIONS message to every P2P SIP client on the standard port (5060), or a standard check_http against 6301 port of the partysip client (Olyo) => client’s IP needed; **solution**: modify client to send IP to a java server that updates *hosts.cfg*
- another option is that the client sends messages to the Nagios Service Check Acceptor, calling *send_nsca* from the client code:
- *$ /usr/sbin/send_nsca -H localhost -c /etc/send_nsca.cfg*

Thank you for your attention!

Questions & Answers

UPI Research Team