VoIP and Videoconferencing: are they the same?

Dr. Saverio Niccolini
Research Staff Member
now @ Network Laboratories, NEC Europe Ltd.
(saverio.niccolini@netlab.nec.de)
VoIP and Videoconferencing

1st remark:
- I would like to speak about IP Telephony instead of VoIP (without changing the title)
  - in my opinion they are not exactly the same and there is much confusion about it
    - please refer to what I wrote on the GNRT on such a topic
Differences?

• Are they so different?
  – With IP Telephony (or VoIP, as you want), the main focus is always on voice services
    • this is not the complete story
    • talking only about voice is misleading regarding the support of video
      – IP Telephony standards and protocols have the capability to signal and are able to initiate multimedia communication
        » what about H.323?
        » what about SIP?
  – IP Telephony technologies/standards and videoconferencing solutions may be seamlessly integrated
    • actually they are based on the same recommendations
H.323 protocol

• What about H.323 HW/SW? (Netmeeting, Polycom systems, etc.)
  – they are based on the H.323 protocol
    • you can call other endpoints using them (voice and video depending on endpoint capabilities)
    • if you have a gateway you can use them to make PSTN calls (voice and video, depending on endpoint and gateway capabilities)
    • videoconferencing capabilities are built-in in the protocol itself
SIP protocol

• What about SIP HW/SW (eyebeam, Polycom systems, miniSIP)
  – it is based on SIP protocol
    • you can call other endpoints using them (voice and video depending on endpoint capabilities)
    • if you have a gateway you can use them to make PSTN calls (voice and video, depending on endpoint and gateway capabilities)
    • videoconferencing capabilities are built-in in the protocol itself
What should we achieve then?

• The goal should be to provide the user with:
  – a global architecture derived from IP Telephony standards (H.323, SIP, ENUM, TRIP, etc.)
  – enabling videoconferencing systems to be used on top of it seamlessly

• perfect videoconferencing sessions are achieved when the technology is no longer noticeable (neither before nor during the session)
  – improvements in the architecture
    » it has to be as simple as make a PSTN phone call
    » application-specific devices to deliver the content to the end-user
Where do we need improvements?

• Global architecture to enable seamless integration (still some conflicts between the PSTN world and the IP one to be solved)

• Accessibility and usability of the system
• Interoperability among different technologies
• Value-added services, such as data/application-sharing and voice mail
What does it mean?

• Discussions at architectural level

• Gateways to provide interoperability with different access types
  – example: not all gateways support video nowadays

• Gateways/bridges to provide interoperability among different IP Telephony protocols
  – example: H.323 users may not be able call SIP users
  – example: H.323 additional services may not be compatible with SIP additional services

• Conference bridges and multipoint conferencing units to provide capabilities for multipoint conferencing
  – example: this is a service already available in H.323, SIP has not yet good application servers for this task, need bridging from SIP to H.323 to achieve this
What is the state of the art?

- Example (SURFNET office):
What is the state of the art?

Example (NEC Network Laboratories):

- **SIP Proxy** (Default Outbound Proxy and B2BUA for NAT/FW traversal with video support using rtpproxy)
- **STUN Server for NAT traversal**
- **Asterisk PSTN-IP Gateway**
- **PBX**
- **office phone**
SIP Deployment at NEC Europe Ltd.

- Network Laboratories in Heidelberg (http://www.netlab.nec.de/)

Inbound call
0. Someone decides to call me at my office number (0049-6221-9051118)
1. The call arrives at NEC PBX in Heidelberg and forwarded to my phone
2. I am away, I have configured an entry to forward the call to my SIP client (my phone does not ring)
3. The call is redirected to the SIP gateway (to the number associated to my SIP client (be it my software client or the WiFi pool phone)

We have not yet configured ENUM, thus no ENUM entry for such a number (gateway just configured statically to handle such redirected calls)
**SIP Deployment at NEC Europe Ltd.**

- Network Laboratories in Heidelberg (http://www.netlab.nec.de/)

---

0. No matter where I am I register to my SIP proxy and I dial the numbers on my SIP clients as I was in the office (0 to exit, etc.)
1. I can call both SIP address, internal numbers and PSTN numbers (configuration of SIP proxy takes care of this)
2. The SIP proxy authorize me and routes my call
   - 2a. SIP-to-SIP: I am only using Internet connection
   - 2b. SIP-to-internal: I am calling my colleagues using internal numbers (65, 39, 32, etc.)
   - 2c. SIP-to-PSTN: I can call all the world as I was in my office (I make a local call in Germany thus my company is happy)
What is the state of the art?

SPADA project
domain: spadatel.org

- Inbound Proxy
  - no rtpproxy yet
- Web server
- Master DB
- Slave DNS
- SIP User Agents
  - GnuGK in full proxy mode
- MGCP User Agents
- Asterisk translation server
- Outbound proxy with nathelper
- VoiceMail
- Conference

- Sip server
- Billing Server
- (B2BUA under development)
- H.323 User Agents
- To/From H.323 GDS

- Master DNS
- Slave DNS
- Other SIP domains

- Local database
- PSTN GW
- pstn

VoIP WS @ TNC2005 - POZNAN - JUNE 2005
What is the state of the art?

• Another example:
  – PolyPhone service at ETH Zurich
  – lots more (not mentioned)
That’s all!

Questions?