IP Telephony Research at Networking Lab 1997 -2001

Basic Technology Evaluation

SIP signaling and XML-services

Interoperability of VOIP and IN/ISDN

Delay

Diploma thesis by Tomi Yletyinen

1) coding delay
2) workstation processing delay
3) packetization delay
4) propagation delay
5) buffering delay in network
6) processing delay in routers
7) receiver buffering delay

1),2),3) 5),6) 4),5),6) 5),6) 7),3),2),1)

Synchronization

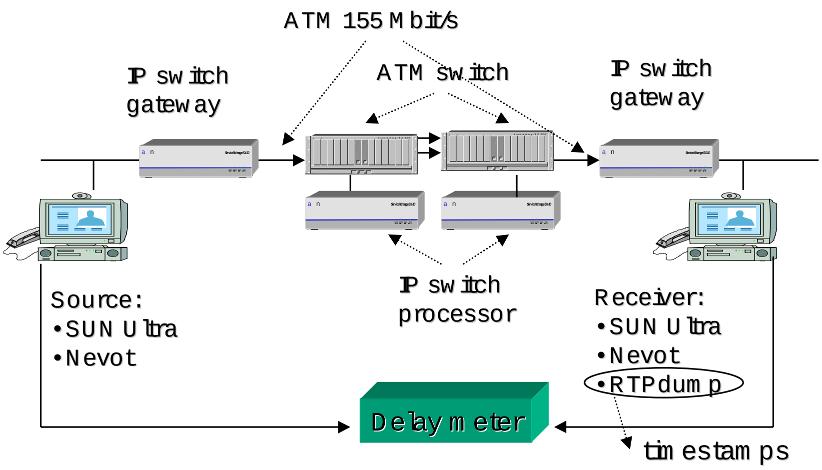
- Adjusting the playout time
- Using RTP-sender timestamps:

  $p_k^i = t_k +$ fixed delay estimate + variable delay estimate
- RTP: Interarrival jitter the mean deviation of the difference in packet spacing at the receiver compared to the sender for a pair of packets
- Difference in packet spacing:

  $D(k^i, k^j) = (a_k^j - a_k^i) - (t_k^j - t_k^i)$
- Various algorithms exist for estimating jitter from D

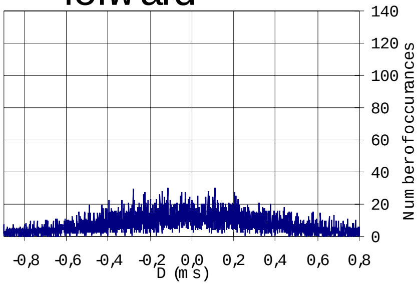
Laboratory setup

- e-to-e delay: 115 ms, network delay (ping): 1-2 ms

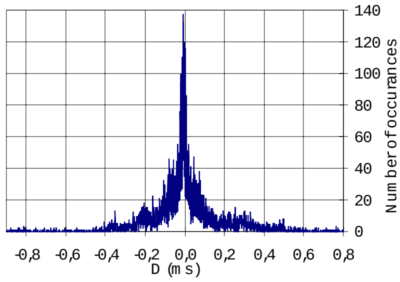


Four processes

- IP store and forward

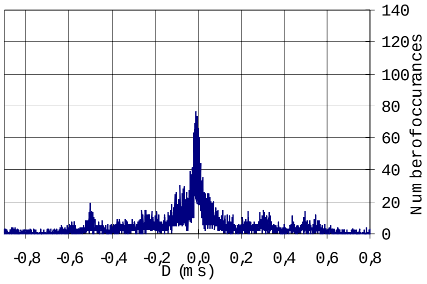


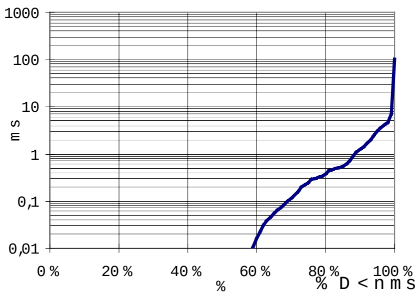
- IP switching



Mixed load

- IP switching





Conclusions on Delay

- Significant differences not expected
  - only four nodes
  - edge-router performance critical in our measurements
- Packet spacing differences for both IP switching and IP store and forward small
  - small non-adaptive playout buffer enough
  - e-to-e delay large, caused by terminals
- IP switching slightly better
  - IP switching increases throughput
  - packet spacing differences smaller
- Long packets increase packet spacing differences
  - bursty traffic and small packets not a problem
- WS are poor phones, network is just a dimensioning matter

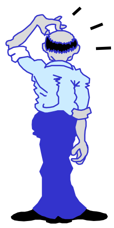
Delay breakdown in a Nevot SunOS Workstation

Diploma thesis by Harri Marjamäki

- End to end delays of 30… 40 ms in a campus intranet are achievable.
- A buffering software bug caused most of the 100ms delay in previous work.
- Processing delay is 1 - 10% of CPU time depending on the coder.

Objectives on SIP signaling and Service Development

M Sc Thesis work by Inmaculada Espigares del Pozo, instuctor: Jose Costa



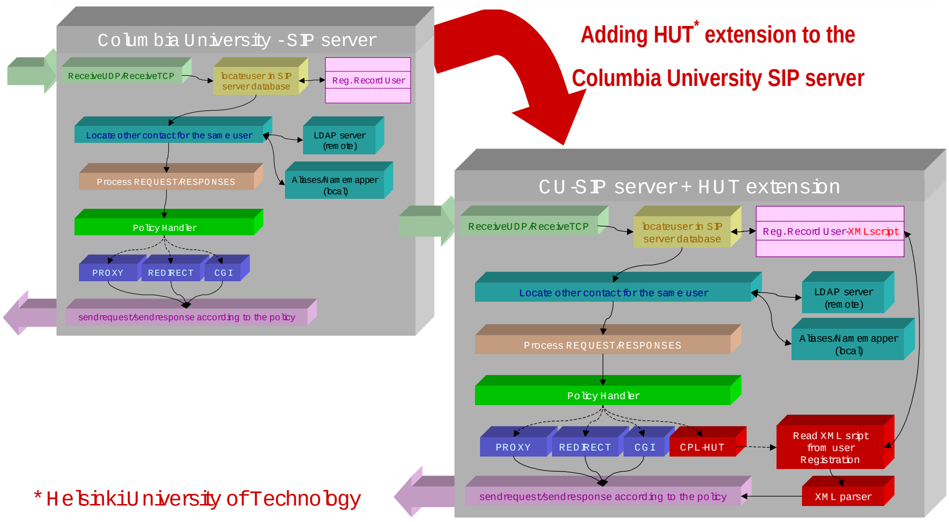
How can new services be defined

from end points???????

Definitely… using NEW TOOLS !!

- SIP for signalling features
- JAVA for building the SIP client
- CPL for describing the services
- XML for defining the service files
- XML parser for interpreting the scripts

HUT extension



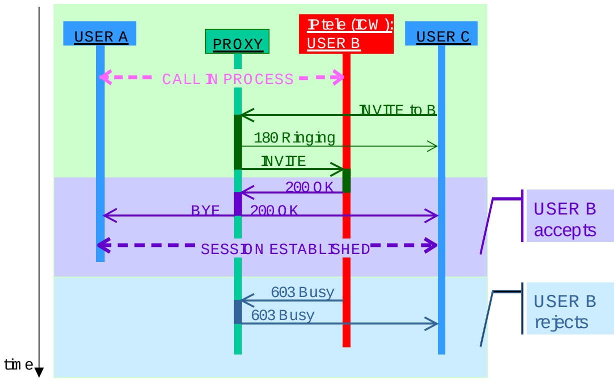
*Helsinki University of Technology

IPtele program

The example service was Internet Call Waiting

IPTele program
is the client
implemented in JAVA

The Service itself
was implemented in
XML



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Service definition

- **<u>XML advantages</u>**
  - **Simplicity: basic structures**
  - **Extensibility**
  - **Interoperability**
    - On a wide variety of platforms
    - Interpreted with a wide variety of tools
  - **Openness: anyone can do it!**
  - **Applications:**
    - Data exchange: Machine-Machine
    - Data interchange: Human-Machine

```
<?xml version="1.0"?>
<call Type="ICW">
  <proxy>
    <icw>
      <forward>
        <link ref="voicemail"/>
      </forward>
      <success>
        <location url="queca@pc2.tct.hut.fi"/>
      </success>
      <reject>The user is Busy now</reject>
    </icw>
    <busy/>
    <noanswer/>
    <failure/>
  </proxy>
  <response status="busy"/>
</call>
```

Circuit Switched Network/VoIP Interoperability

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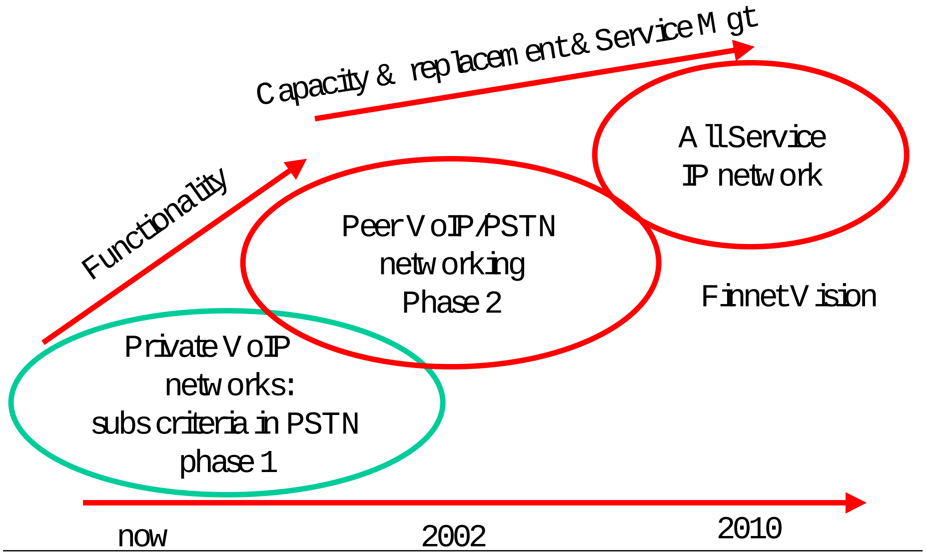
Outline

- Roadmap and Interoperability Issues
- Signaling and QoS overview
- Routing information problem
  - Requirements
  - Locating GWs from the IP Telephony network
  - Locating a SG from the ISDN network angle.
  - Number portability across the technology boundary.
- The Solution to Telephony Routing over SCN/IP - hybrid network.

Roadmap to the Future



Interoperability Issues

- Signaling and Call control
- Quality of Service

Phase 1
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- Telephony Routing and addressing
  - Input Information gathering
  - Alternative routing over IP

Phase 2
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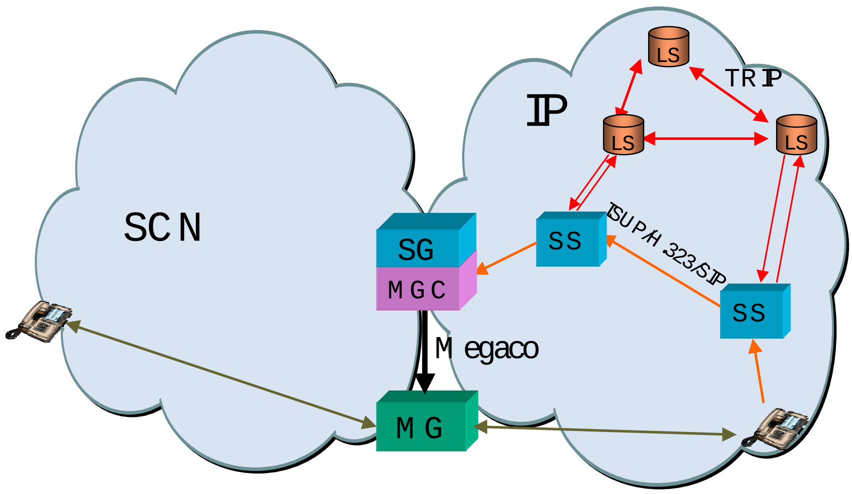
- Service Management in the hybrid network

Phase 3

Phase 2 Requirements

- Efficient routing and numbering infrastructure across the emerging hybrid network is a necessity
  - Delay and jitter highly depend on call path
- In all call scenarios at all cost we must avoid unnecessary conversion between IP and PSTN.
  - Call Forwarding, Number Portability, Roaming, 800-numbers ...

Current Architecture



TRIP = Telephony Routing over IP, SG - Signaling Gateway, MGC - Media Gateway Controller
MG - Media Gateway, SS = Signaling Server, LS = Location Server

For Telephony routing we must choose optimal Gateway

- The IP Telephony view:
  - LS provides info about Next hop Signaling server e.g. a Signaling Server or an MGC in the same domain
  - TRIP keeps information in LSs updated across IP Telephony systems
  - MGCs are registered e.g in LS (this information may be local to an Adm in Domain)
  - SS can use LS to locate MGC and MG

How does the SCN choose a GW?

- ISDN, GSM, PSTN view
  - Good news: SGs are large - easy to locate
  - Bad news: I do not hear Any Body working on the problem of Gateway location from the ISDN point of view
    - From the SCN it is equally important to select the most suitable Gateway for SCN to IP calls
    - Btw, we are working on this problem in Telelab

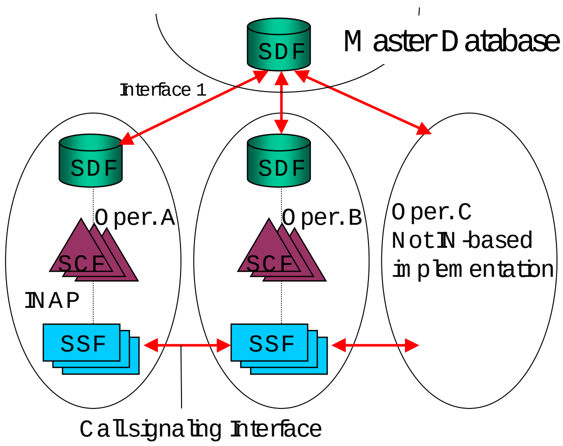
Numbering Issues

- What if an IP Telephony Number is ported to another ITSP operator?
  - ISDN side may need to choose another SG for calls to that number
- What if an ISDN number is ported to another ISDN operator?
  - IP side may need to choose another set of SG, MGC, MG
  - LSs need to know about the change
- What if a number is ported SCN to IP or vice versa

Current situation at the ISDN side

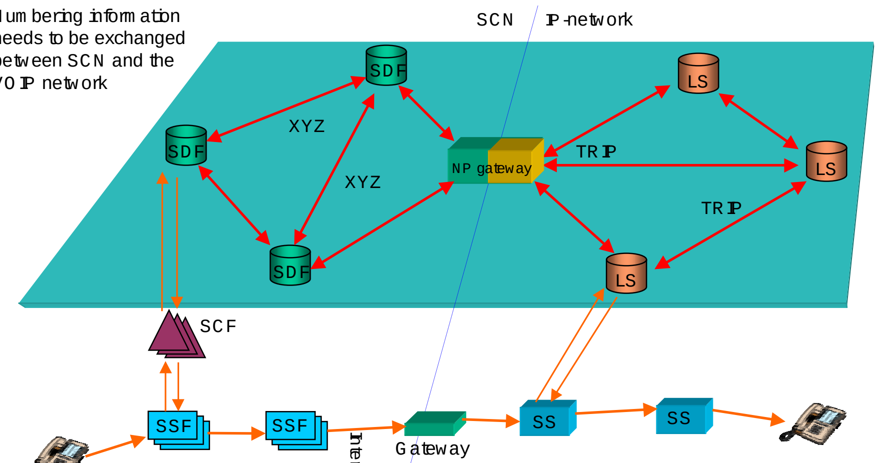
Number Portability is mandated by regulators in Europe and the US

Typical solution is based on IN



ISDN needs a pair to TRIP

Numbering information needs to be exchanged between SCN and the VOIP network



TRIP = Telephony Routing over IP, NP-gateway = Gateway for sharing information between XYZ and TRIP, SS = Signaling Server, LS = Location Server, SDF = Service Data Functions, SCF = Service Control Function, SSF = Service Switching Function

Requirements for Numbering & Routing

Number portability for IP subscribers

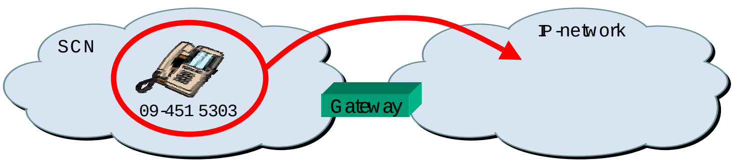
Number portability within IP-networks.

Number portability between the SCN- and IP-networks.

Integration with the TRIP (Telephony Routing over IP) protocol for location of gateways and signaling servers. Integration with TRIP and DNS (enum) for location of IP terminals.

Optimisation of routing between SCN- and IP-networks for portable numbers.

- Location of nearest or most suitable gateway
- Support for several geographical areas



Architecture for Numbering & Routing

Distributed architecture

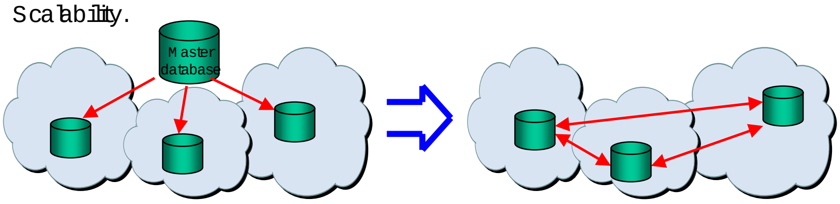
A distributed database instead of a single master database.

No single point of failure.

Master DB to SDF Interface replaced by a distributed database based on e.g. SCSP (Server Cache Synchronisation Protocol).

Database updates made directly by the operators. Support for subscriber-initiated updates possible.

Scalability.



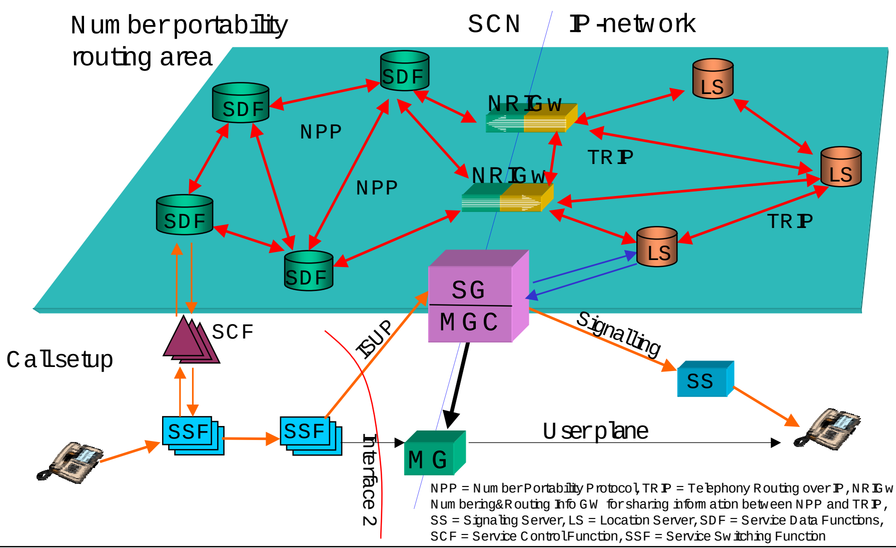
Requirements for 800- and GSM numbers

- IP Telephony view
  - an 800-number and a Cellular Mobile Number may be located anywhere in the ISDN/PSTN cloud or the Cellular cloud respectively
  - additional round of indirection for choosing the GW is needed to ensure adequate quality voice
  - LS needs to cascade the request to an SDF or to an HLR or return the address of an SDF or HLR so SS can make a subsequent query

Requirements for 800- and GSM numbers

- SCN view
  - an 800-number (and a Cellular Mobile Number - only a matter of time!) may be located anywhere in the IP cloud
  - additional round of indirection for choosing the GW is needed to ensure adequate quality voice
  - SDF needs to cascade the request to an LS
  - It is not efficient to flood Mobile numbers among LSs when a mobile number is in an IP cloud - a solution scalable to frequent location changes is needed

The solution is NPP + NRIgateway

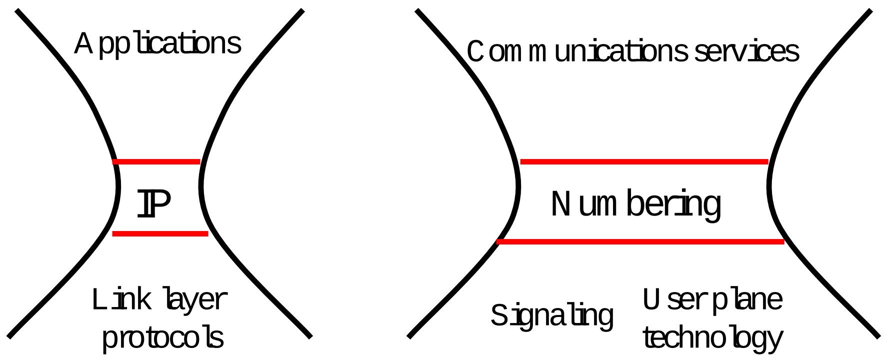


NPP = Number Portability Protocol, TRIP = Telephony Routing over IP, NRIGw = Numbering&Routing Info GW for sharing information between NPP and TRIP, SS = Signaling Server, LS = Location Server, SDF = Service Data Functions, SCF = Service Control Function, SSF = Service Switching Function

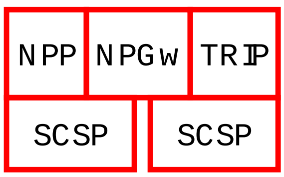
An analogy

Protocol centered view
"How"

Reachability view
"To whom you can call"



Conclusions on Routing Information



- Gateway model needs to be complemented by Numbering & Routing Information gateways
- SCSP can be the common Numbering infrastructure component for both SCN and IP Telephony networks

- Location servers need to be able to cascade requests to cater for 800-numbers, any service specific routing methods and for mobility
- How to avoid exhausting link capacity by real-time services and starving BE service is still open!