The Fundamentals of IP Telephony

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The Fundamentals of IP Telephony …

Terminology:

- Internet Telephony
- IP Telephony
- Voice Over IP
- Third Generation (3G) Telecommunication Networks
- Packet cable telephony
- Others …

A bit of history (Telephony)...

First generation cellular networks (70s – 80s)

- Analog systems, circuit switching based
  - Total Access Communications Systems (TACS) – UK
  - Advanced Mobile Phone Systems (AMPS) – USA/Canada
  - Nordic Mobile Telephone System (NMT) – Scandinavia

Second Generation (90s – early 00s)

- Digital systems, circuit switching based
  - GSM – Europe mainly – However, gaining ground in North America
  - D-AMPS (Digital version of AMPS)
  - PDC (Japan)

A bit of history (Telephony) ...

Third Generation (early 00s –)

- Still digital, but more capacity
- Packet switching based
- Two main standards for mobile networks
  - UMTS
  - CDMA 2000

Fourth Generation (early 10s –)?

- At research stage
- No standard yet

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IP Telephony …

Distinctive characteristics

1. Packet switching (instead of circuit switching in today's 2G networks)
2. QoS enabled (unlike the Internet best effort)
3. Voice + data (unlike today's 2G networks which focus on voice)

A bit of history (Packet switched based Telphony) …

Telephony over packet switched networks …

- Late 70s:
  - First two party voice calls over Internet (Network Voice Protocol (NVP - RFC 741 - November 1977)
- 80s:
  - Emergence of proprietary systems for Internet Telephony
- 90s:
  - Emergence of standards (e.g. SIP, H.323, Megaco/H.248)
- Early 00s:
  - Backing by telcos (e.g. 3GPP specifications)
  - Backing by other new players (e.g. cable industry)

The Fundamentals of IP Telephony …

Part I: Networking

Part II: Value Added Services

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The Fundamentals of IP Telephony …

More information: Slides on my URL
http://www.ece.concordia.ca/~glitho/

1. Graduate course at Concordia University, Montreal
Value Added Services in Next Generation Networks

2. Course at IMSP Porto-Novo, Benin
Reseaux et Telecommunications I

3. Tutorials at various conferences, IEEE distinguished lecturer tours

.

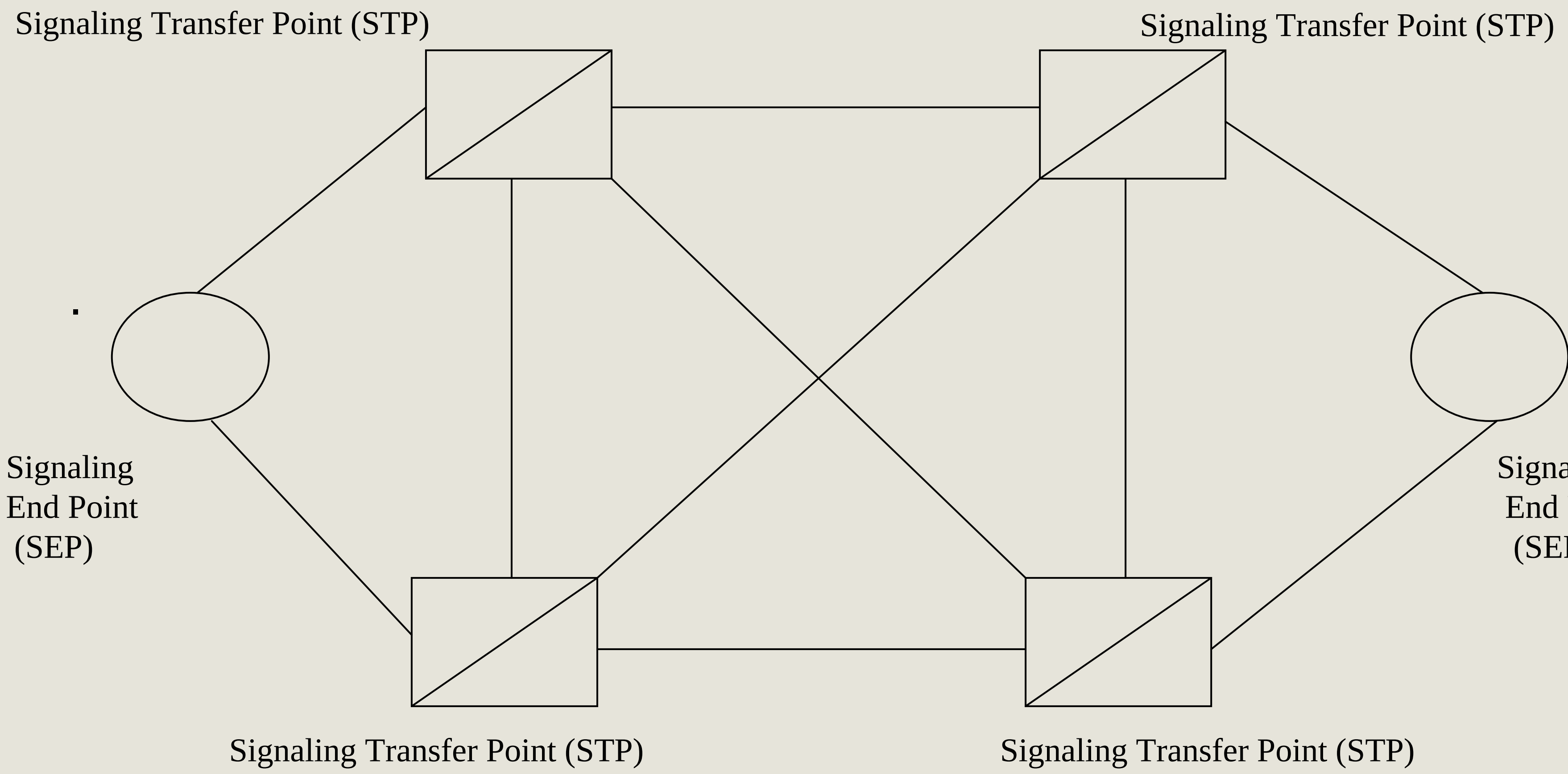
Part I: The networking Aspects Of IP Telephony

- Signaling
- Media handling
- QoS
- Inter-working with legacy

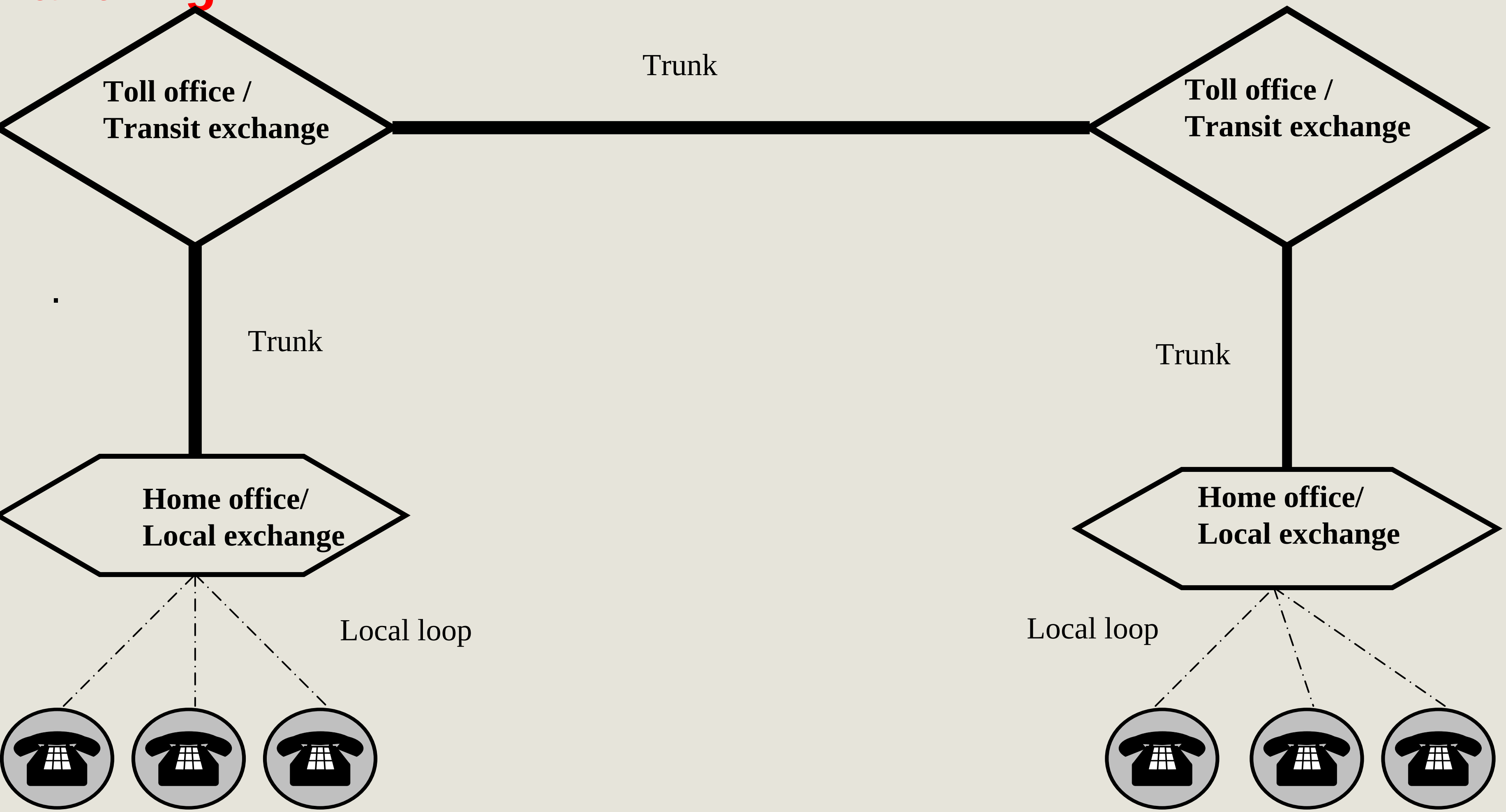
Circuit switching vs. packet switching

Principal Criteria	Circuit switched	Packet switched
Dedicated Physical path	Yes/No	Yes/No
Derived criteria	Circuit switched	Packet switched
Call set up required	Yes/No	Yes/No
Possibility of congestion during communication	Yes/No	Yes/No
Fixed bandwidth available.	Yes/No	Yes/No
Non optimal usage of bandwidth	Yes/No	Yes/No

A simplified circuit switched telephony network: signaling ...



A simplified circuit switched telephony network: media handling



Inherent weaknesses ...

Signalling

- Two party voice call focused
  - Little / no room for multiparty sessions (e.g. conferencing)
  - Little / no room for multimedia sessions

Media handling

- Inefficient usage of resources

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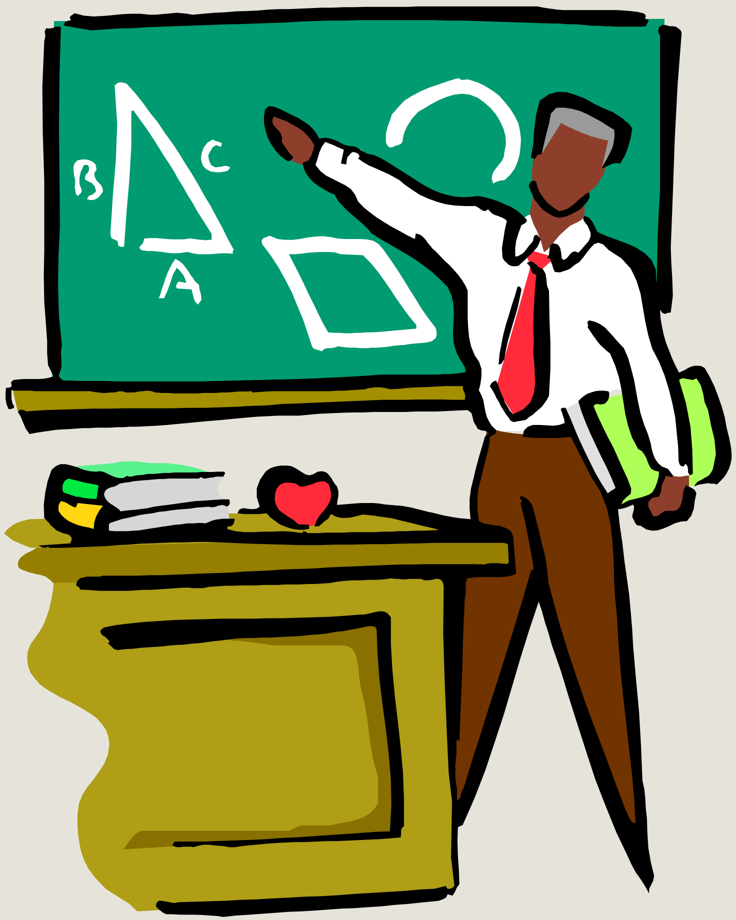
Signaling



- H.323
- SIP
- 3GPP Networks

H.323: Introduction

An umbrella ITU-T standard

- First set of standards for Internet Telephony
  - First version published 1996
  - Targets LAN / enterprise networks and subsequently “twisted” to work in telecommunications networks environment

- Includes:
  - signalling standards:
    - H.225.0
    - Q.931
    - H.245
  - Others (e.g. H.324 Terminal for low bit rate multimedia communications)

H.323: Introduction

Used in conjunction with other IETF protocols

- QOS related protocol (e.g. RSVP)
- Media transportation related protocol (e.g. RTP - RFC 1889)

Lost the battle for 3G cellular networks to SIP in 2000

- Decision initially made by 3GIP, then subsequently endorsed by:
  - 3GPP / 3GPP2
  - PacketCable

Remain important because of:

- Installed base, especially in enterprise environments

H.323: The functionality entities

Terminals

- End point
- Used for real time two way multimedia communications with another end point

Gatekeeper

- Control how terminal access networks
- Provide address translation

Gateway

- End point
- Used for communications between H.323 terminals and terminals in the PSTN

Multipoint control unit (MCU)

- Provides centralized conferencing functionality

H.323 signaling: Registration Admission and Status (RAS)

- Signaling between end-points
  - Terminal or gateway
    and
  - Gatekeeper
  - Use unreliable channels
    - Retries
    - Timeouts
  - Discovery of gatekeepers and registrations

H.323 signaling: Call Set Up (H.225)

Key features

- ISUP signaling (Q.931) based
- Signaling between end-points
  - Terminal or gateway

    and
  - Gatekeepers
  - Use reliable channels

Several alternatives

- All end points hidden behind gatekeepers (most common)
- End points connected directly
- Some end points hidden and others connected directly

H.323 signaling: Media signaling (H.245)

Key features

- Several modes
  - Request/reply
  - Commands
  - Indications
- Signaling between end-points
  - Terminal or gateway

    and
  - Gatekeeper
- Use reliable channels

H.323 signaling: An important feature - Fast connect

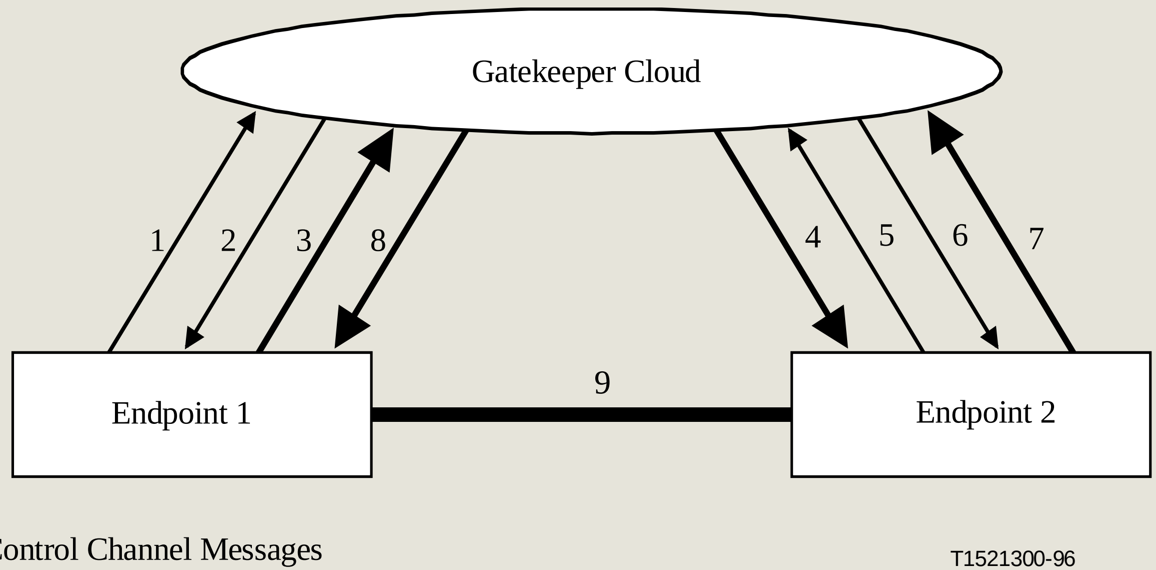
Introduced as an afterthought in H.323

Allow call set up and logical channel set up using a single message

- FASTCONNECT
  - Include as parameter fast start to indicate that logical channel should be opened
  - May be refused by the other end (Fast connect refused)

H.323 signaling : Putting it together …An example

1 ARQ
2 ACF/ARJ
3 Setup
4 Setup
5 ARQ
6 ACF/ARJ
7 Connect
8 Connect
9 H.245 Channel



H.245 Control Channel Messages

Call Signalling Channel Messages

RAS Channel Messages

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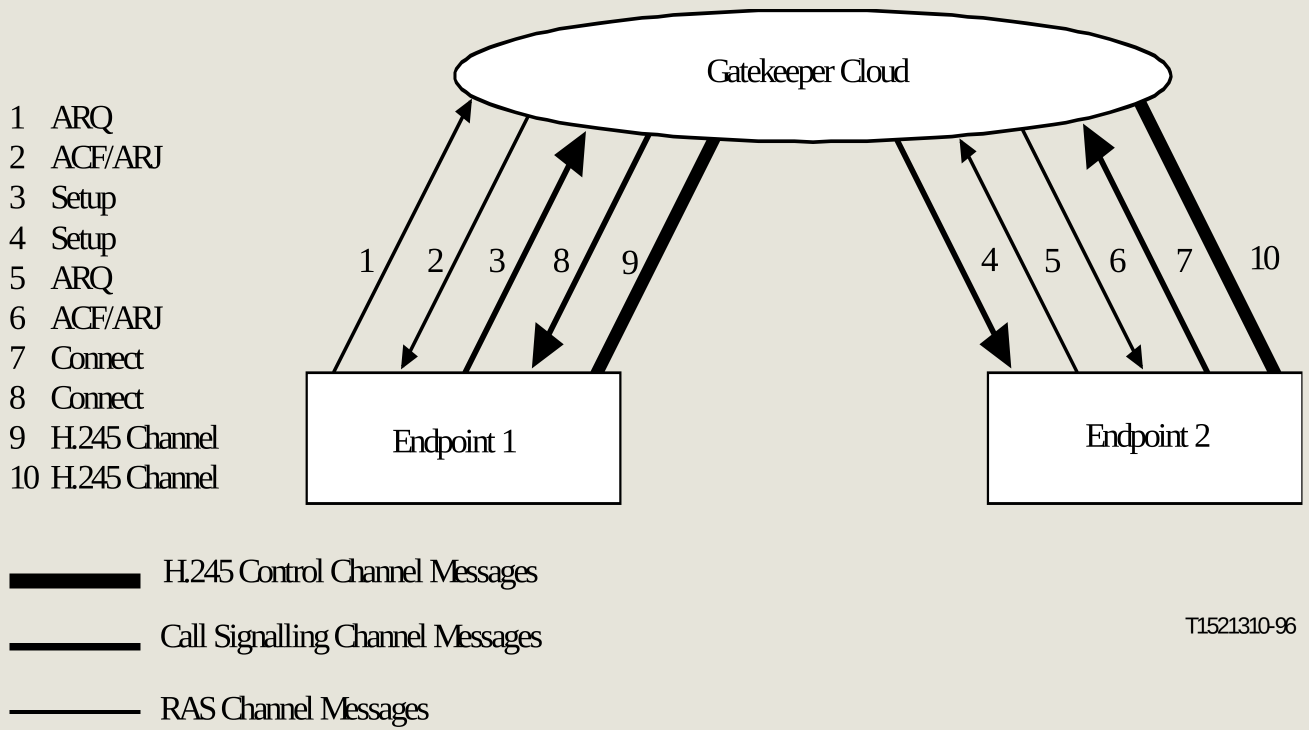
H.323 signaling: Putting it together - Another example



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SIP: Introduction

A set of IETF specifications including:

- **SIP core signalling:**
  - RFC 2543, March 1999
  - RFC 3261, June 2002 (Obsoletes RFC 2543)

- **SIP extensions (e.g. RFC 3265, June 2002 - Event notification)**
  - May have nothing to do with signalling

- **Used in conjunction with other IETF protocols**
  - QOS related protocol (e.g. RSVP)
  - Media transportation related protocol (e.g. RTP - RFC 1889)
  - Others (e.g. SDP - RFC 2327)

SIP: Introduction

- Prime signaling system because adopted by all key next generation networks:
  - 3GPP
  - 3GPP2
  - PacketCable:

SIP: The functional entities

User agents

- **End points, can act as both user agent client and as user agent server**
  - **User Agent Client: Create new SIP requests**
  - **User Agent Server: Generate responses to SIP requests**
- **Dialog: Peer to peer relationship between two user agents, established by specific methods**

Proxy servers

- Application level routers

Redirect servers

- Redirect clients to alternate servers

Registrars

- Keep tracks of users

SIP: The functional entities

State-full proxy

- Keep track of all transactions between the initiation and the end of a transaction
- Transactions:
  - Requests sent by a client along with all the responses sent back by the server to the client

Stateless proxy

- Fire and forget

SIP: The messages

Request messages

- Methods for setting up sessions
  - . INVITE
  - . ACK
  - . CANCEL
  - . BYE

- Others
  - . REGISTER (Registration of contact information)
  - . OPTIONS (Querying servers about their capabilities)

SIP: The messages

Response message

- Provisional
- Final

Examples of status code

1xx: Provisional

2xx: Success

6xx: Global failure

A digression on SDP …

Session Description Protocol

- Convey the information necessary to allow a party to join a multimedia session
  - Session related information
  - Media related information
- Text based protocol

- No specified transport
  - Messages are embedded in the messages of the protocol used for the session
    - Session Announcement Protocol (SAP)
    - Session Initiation Protocol (SIP)

A digression on SDP …

Session Description Protocol

- <Type> = <Value>
- Some examples

    Session related

        v= (protocol version)

        s= (Session name)

    Media related

        m= (media name and transport address)

        b= (bandwidth information)

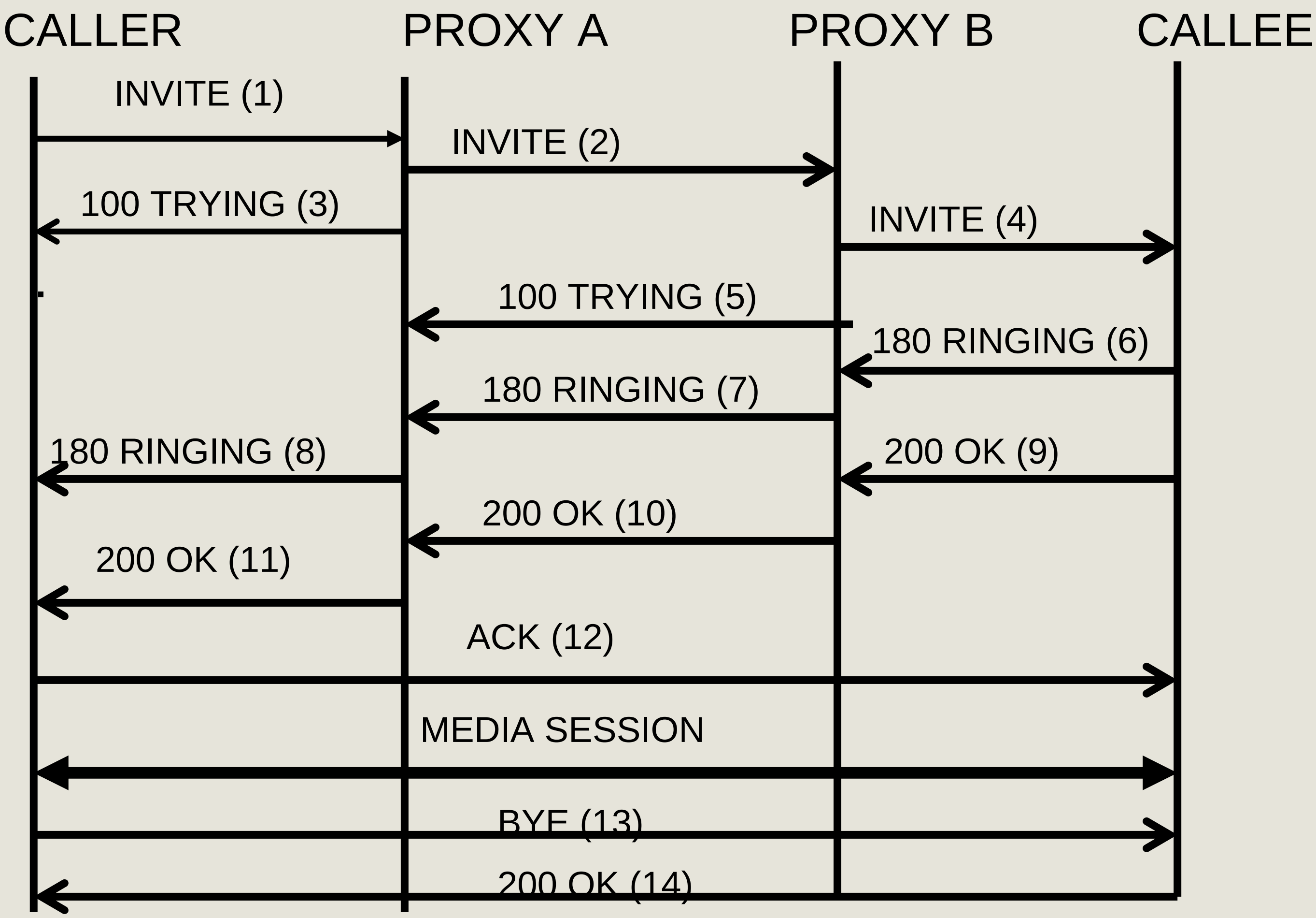
A digression on SDP …

Session Description Protocol

Use with SIP

- Negotiation follows offer / response model
- Message put in the body of pertinent SIP messages
  - INVITE Request / response
  - OPTIONS Request / response

SIP: A simplified call case



3GPP networks: Essentials

Made of:

- Legacy
  - Circuit switched part (GSM)
  - Packet switched (GPRS)
- Next generation part (IP multimedia (IM))
- Inter-working
- Some of the functional entities are common to both legacy and NGN (e.g. Home Subscriber Server)

- Adoption/extension of existing NGN specifications:
  - SIP instead of H.323
  - H.248/Megaco

3GPP Networks: Functional entities

Some of the functional entities

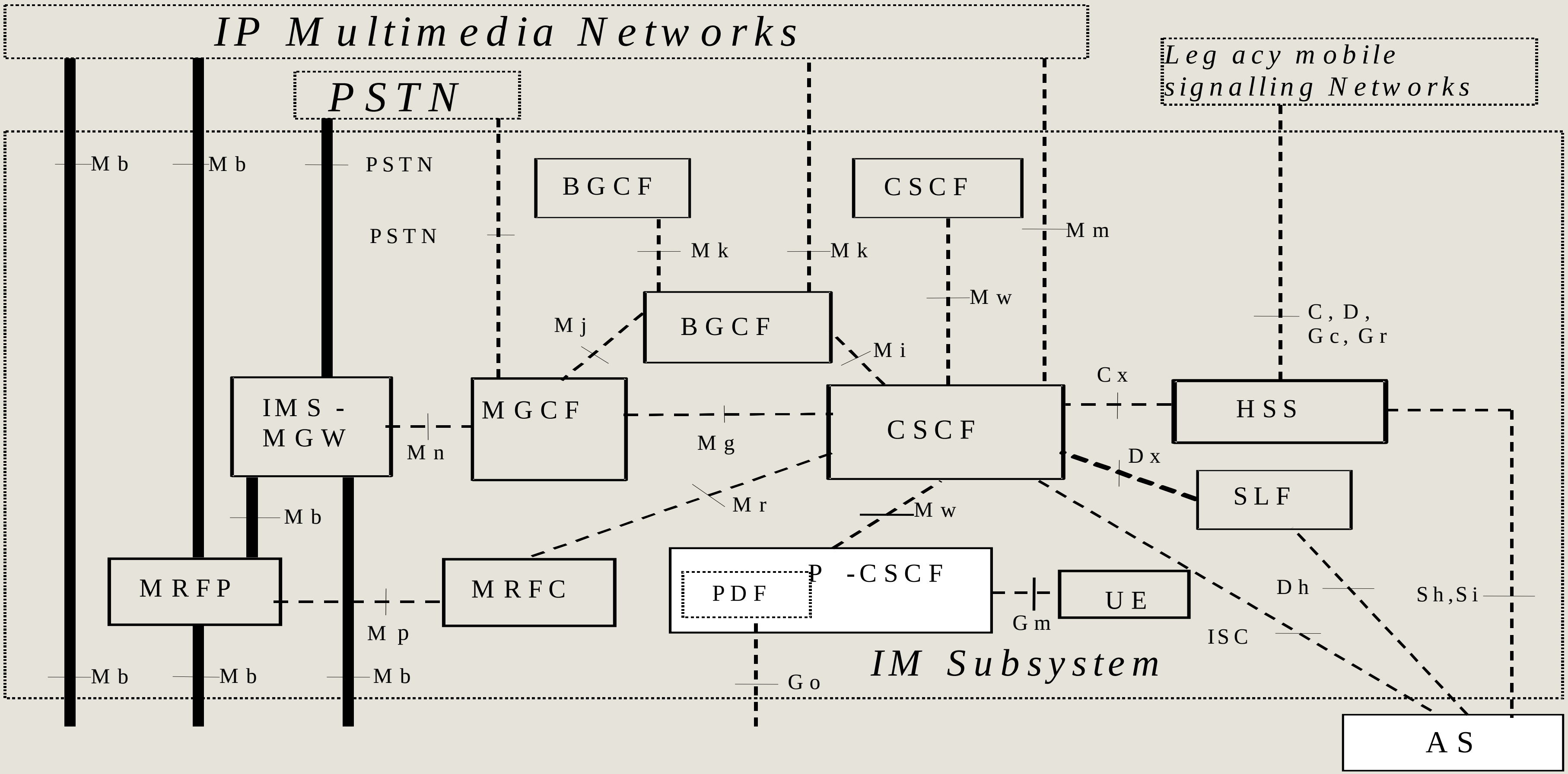
Call Session Control Function (CSCF)

- Proxy-CSCF: First contact point in the IM network – Accepts requests and proxies them
- Serving-CSCF: Perform session control for all user entities in the networks including visitors
- Interrogating CSCF: Contact point in an operator domain for all users (home users, and visiting users)
  -

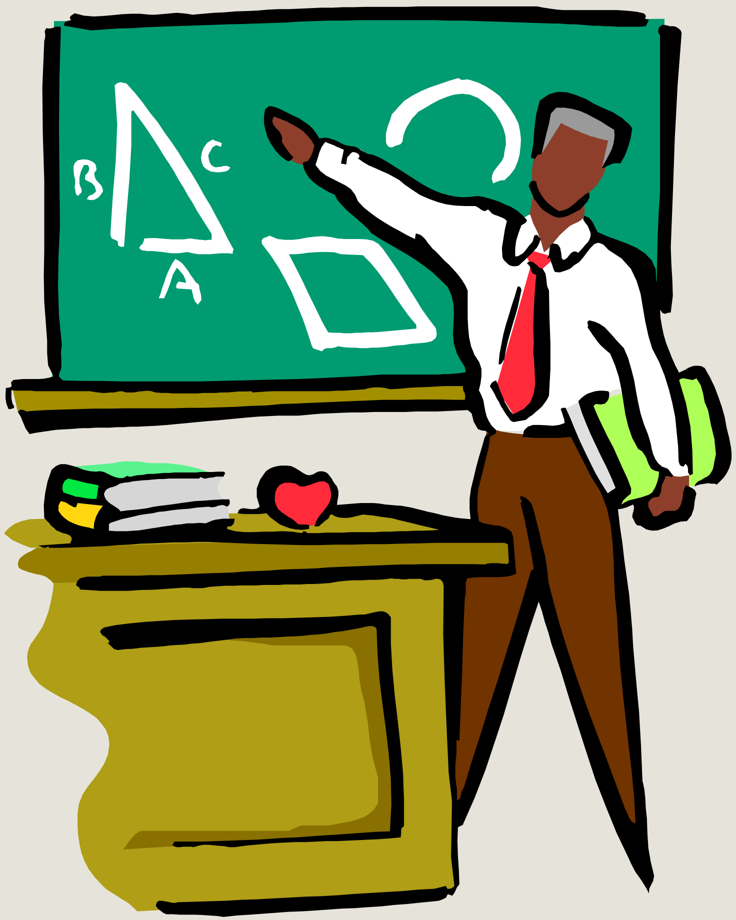
Home Subscriber Server (HSS)

- Master data base – subscription / location information

3GPP Networks: IP multimedia portion



Media handling



- Basics
- RTP
- RTCP

Basics …

Media handling …

- Transportation …
- Conversion
- Mixing

Related concepts

- Media stream
  - Simple streams (e.g. voice)
  - Multiplexed streams (e.g. voice + video)
  - Ports or transport selectors
- Content type (I.e. format)
  - Examples: MPEG

Key issue: Real time delivery and processing

Basics …

Two complementary protocols

- Actual transportation:
  Real-time Transport Protocol (RTP)
- Control of transportation:
  RTP Control Protocol (RTCP)

Basics

Main characteristics

RTP:

No provision for Quality of service

No guarantee for out of sequence delivery

Typically runs on top of UDP but may run on top of other protocols

RTCP:

Help in providing control

RTP concepts …

Session

- Logical association between parties communicating with RTP
  - Identified for each participant by:
    - IP address (may be common for all participants)
    - RTP port
    - RTCP port

End system

- Application that generates the content to be sent and/or
- receive the content to be consumed
- Examples: IP phones, PCs, microphones …

RTP concepts …

Mixers / translators

- Intermediate systems
- Connect 2 or more transport level clouds
  - End systems
  - Mixers / translators
- Use cases
  - Centralized conference bridges
  - Heterogeneous conferences
    - Low speed connection
    - High speed connection
    - Different encoding schemes
  - Some participants behind firewalls

RTP packets: Structure

Header

- Fixed
- Maybe followed by one header extension if extension bit is set

Body

- Contains the actual data

RTP header – Selected fields

- Version :
- Extension:
- Payload type: Format of payload (e.g. encoding scheme)
  - Profile for audio and video conference
  - Other types
- Sequence number
- Time stamp

RTCP concepts …

Monitor:

- Application that receives RTCP packets sent by participants in an RTP session

Reports

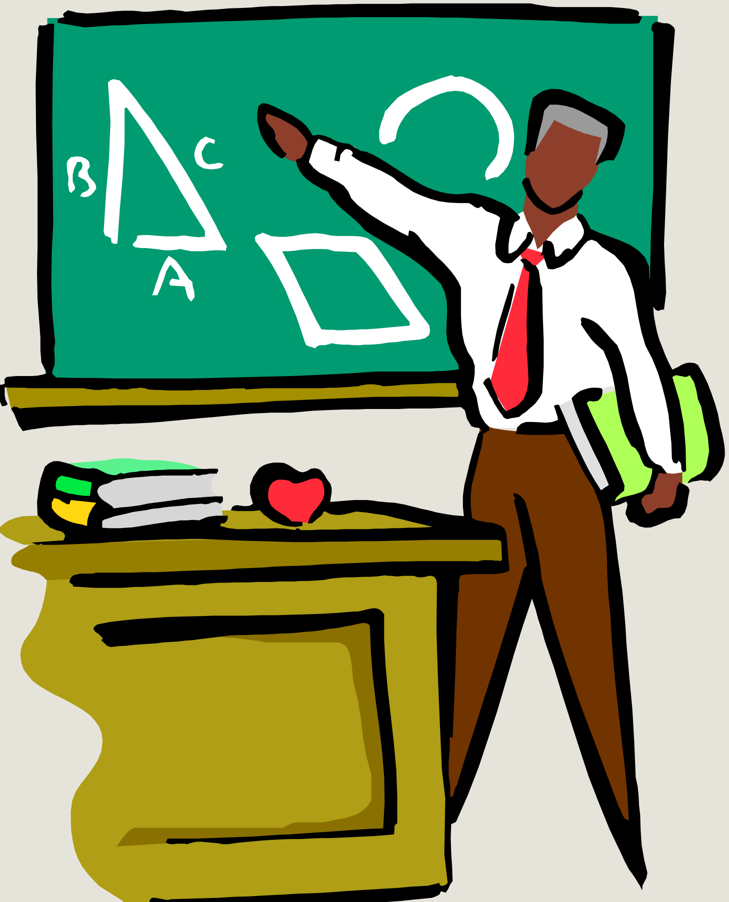
- Reception quality feedback
- Sent by RTP packets receivers (which may also be senders)

RTCP packets ...

- Packet types
  - Simple
  - Compound

- Examples of packets
  - Sender reports
  - Receiver reports
  - Bye

Quality of Services



1. Introduction
2. Early attempts
3. Differentiated services

Introduction …

1. Circuit switched telephony

- Reserved path
- Single grade of service (The highest)

2. Classical Internet

- No reserved path
- Single grade of service (I.e best effort)
- Highly unsuitable for telephony

Early attempts: Integrated Service Architecture - IntServ …

Provide end to end QoS guarantees

Service classes

1. Guaranteed service

- Hard guarantee on delay and bandwidth
- Parameters provided by application
  - Peak rate
  - Packet size
  - Burst size

2. Controlled load

- Softer version of guaranteed service
- Guarantee that the QoS is equivalent to what it would have been if the network is not overloaded
- May not meet some of the hard requirements (e.g. delay)

Integrated Service Architecture - IntServ …

Requirements on each router in the path:

1. Policing
2. Admission control
3. Classification
4. Queuing and scheduling

Resource Reservation Protocol - RSVP …

- Soft state signaling protocol used in InServ for uni-directional resource reservation

- Rely on two messages:

PATH

- Propagated from sender to receiver

RESV

- Propagated in the opposite direction

Integrated Services …

Disadvantages

- Require major new software and firmware in routers
- Major overhead due to flows management
  - Flows are quite similar to telephone calls
    - Set up
    - Tear down

Differentiated services - DiffServ …

Aim at addressing IntServ drawbacks by focusing on traffic aggregates instead of individual flows:

Scalability

- No need for router to maintain flow states
- No for refreshment messages due soft-state

Lack of general applicability

- Work even if every router in the path does not support it

No need for applications to support new APIs

Differentiated services - DiffServ …

Fundamental principle: A code point – Differentiated service code point (DSCP) to tell routers how to treat a packet relatively to other packets

Per hop behaviour (PHB)

- Default
- Expedited forwarding
- Assured forwarding

Routers use PHB to drop/ prioritize packets on their output queue

Differentiated services - DiffServ …

The two approaches:

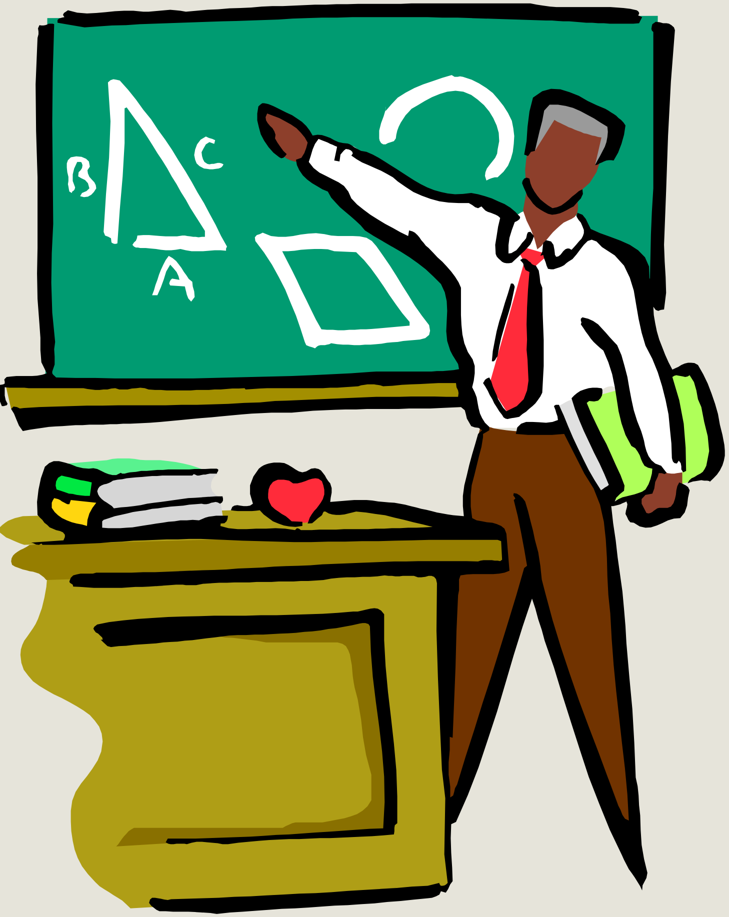
Absolute service differentiation

- Try to meet IntServ goals, but:
  - **Without per-flow state**
  - **With static / semi-static resource reservation**

Relative service differentiation

- **Lower level of ambition**
- **Just ensure that relative priorities are respected**

Inter-working with legacy: Megaco / H.248



1. Introduction
2. Concepts
3. Protocol
4. Call cases

Introduction

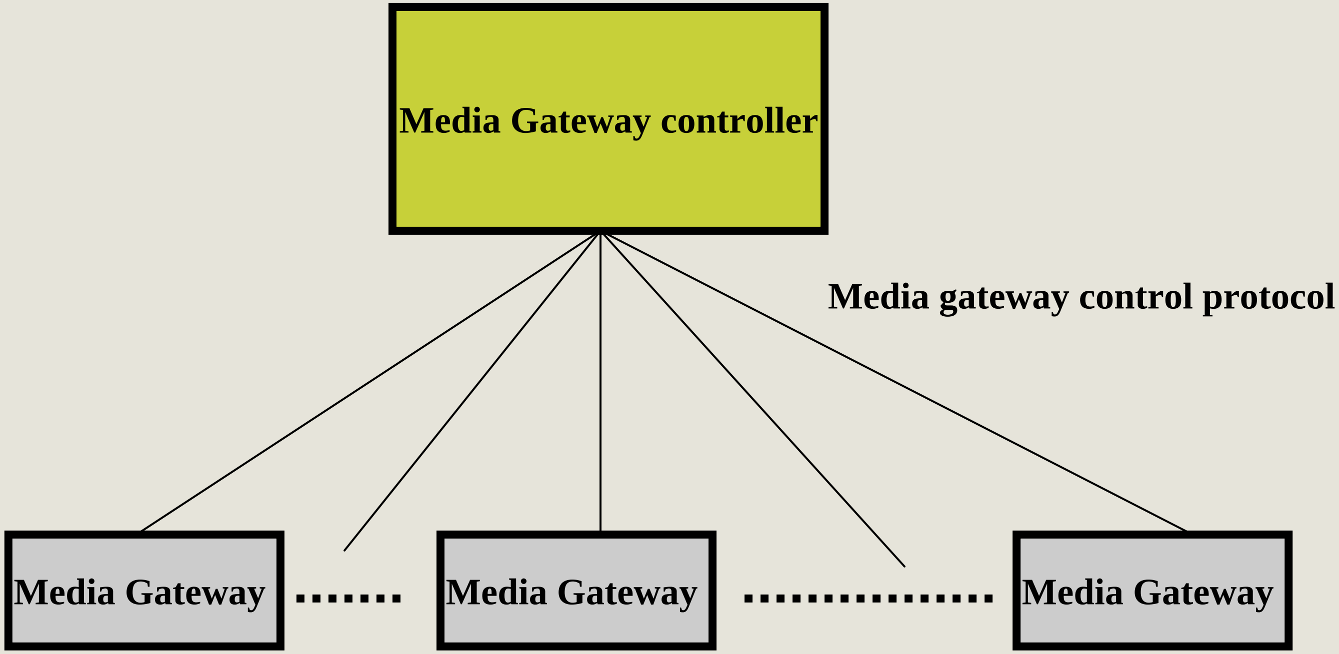
Primary motives for decomposing gateways between PSTN and next generation networks:

- Scalability
- Specialization
- Opening up of market to new players

Side-effect

- Possibility of using the part of the decomposed gateway for call control
  - Soft-switches

Introduction



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Megaco/H.248: Genesis

A long history starting in 1998

- Simple Gateway Control Protocol (SGCP)
  - Text based encoding, limited command set
- IP Device Control Protocol (IPDCP)
  - A few more features to SGCP
- Media Gateway Control Protocol (MGCP)
  - Merge of SGCP and IPDC
- Media gateway Decomposition Control Protocol (MDCP)
  - Binary encoded
- Megaco / H.248 (Joint IETF / ITU-T specifications)
  - A compromise
    - Both text based and binary encoding
    - A wide range of transport protocols(e.g. UDP, TCP, SCTP)

Concepts

Termination

Source or sink of media

- Persistent (circuit switched) or ephemeral (e.g. RTP)

Context (mixing bridge)

- Who can hear/see/talk to whom
- Association between terminations
- May imply
  - Conversion (RTP stream to PSTN PCM and vice versa)
  - Mixing (audio or video)

Protocol - Commands

Add termination to a context

Modify the properties of a termination

Subtract a termination from a context

Move a termination from a context A to context B

Audit (values or capabilities)

Notify

ServiceChange (specific type of notify – terminations about to be taken out of service)

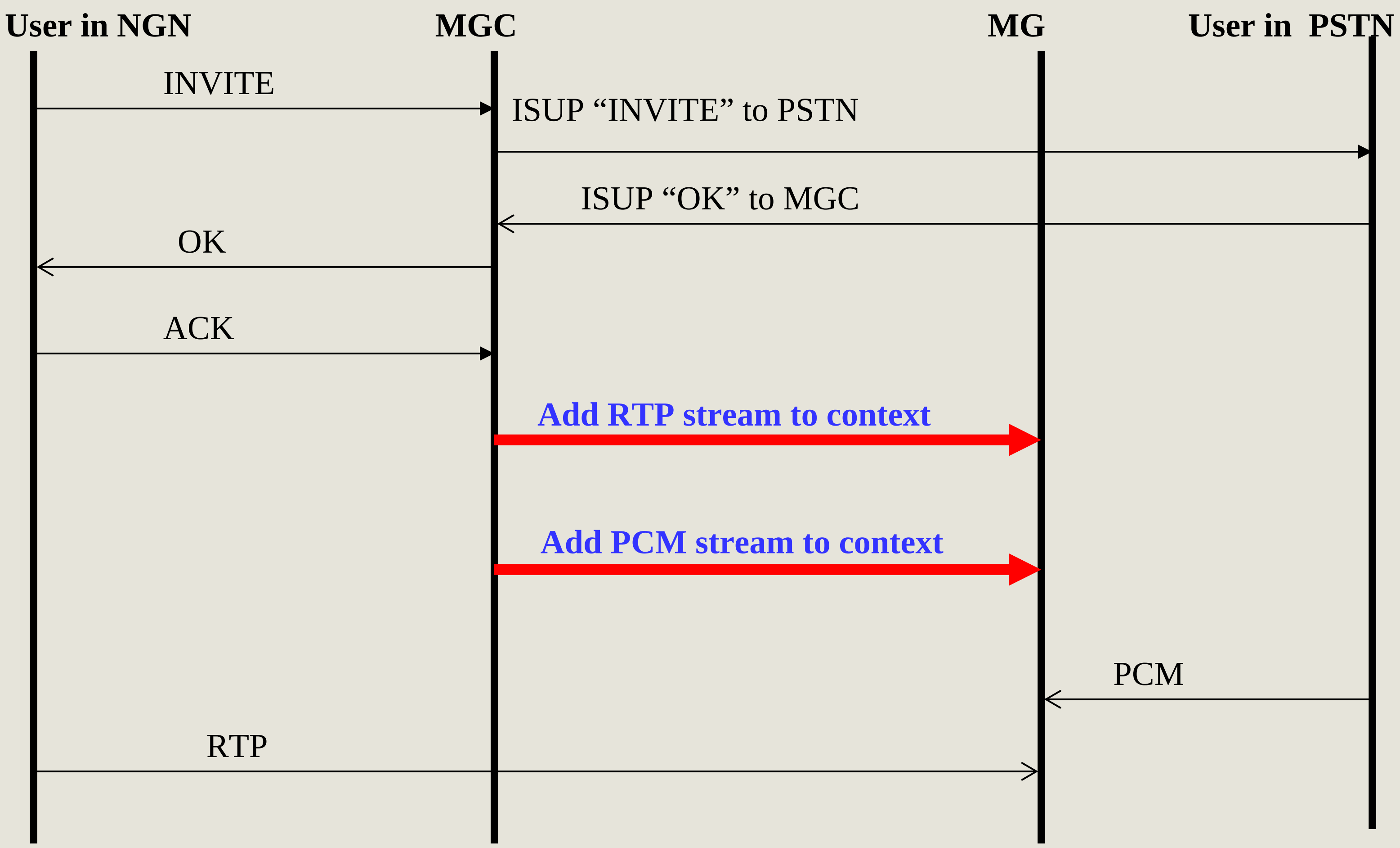
Protocol - Transportation

Several alternatives

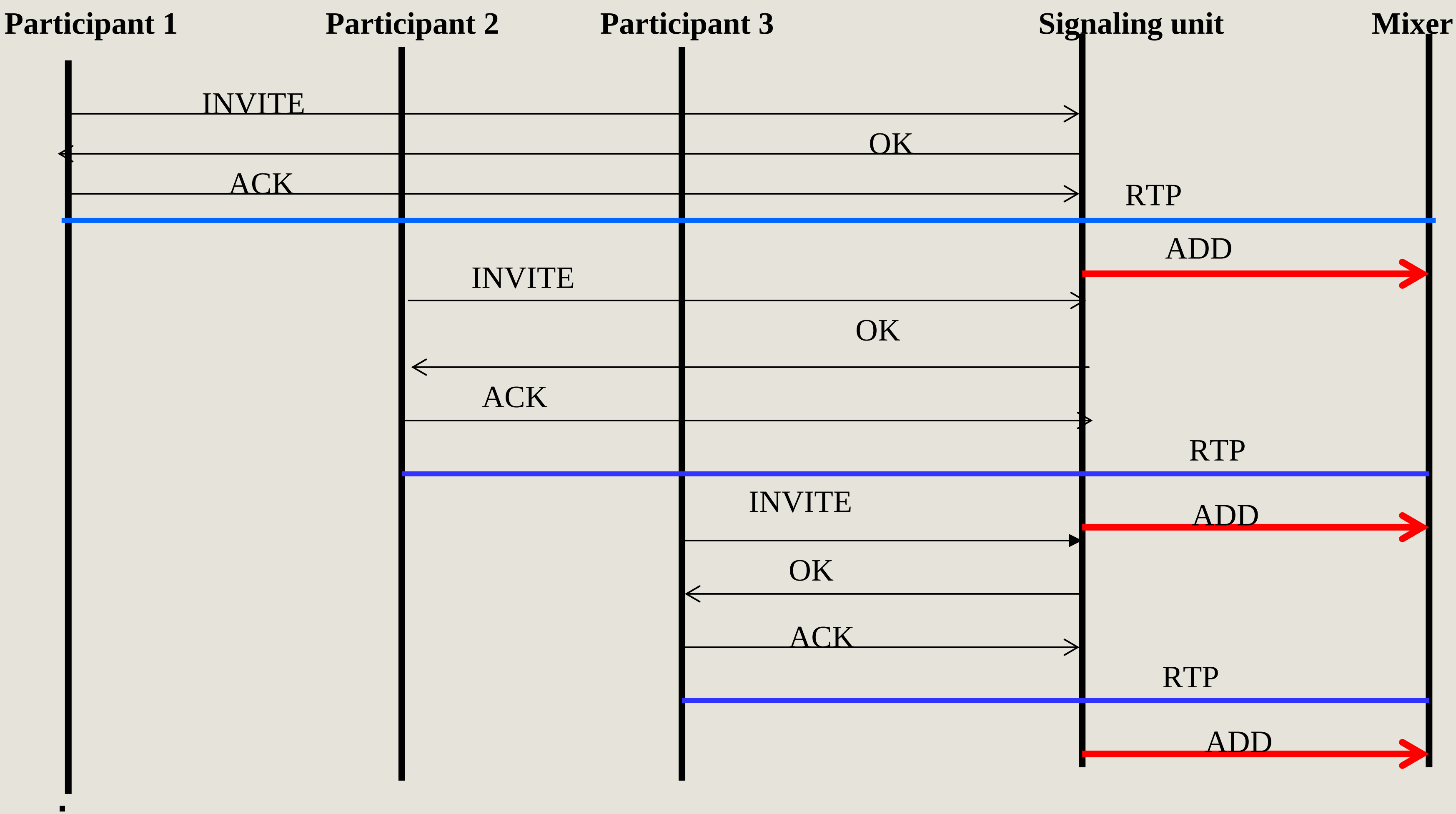
An example

- UDP/IP
  - Unreliable, timeouts / resends
  - At most once functionality required (Receivers should keep track of received commands)

Megaco/H.248: PSTN / NGN Interconnection …



Megaco/H.248: Conferencing ...

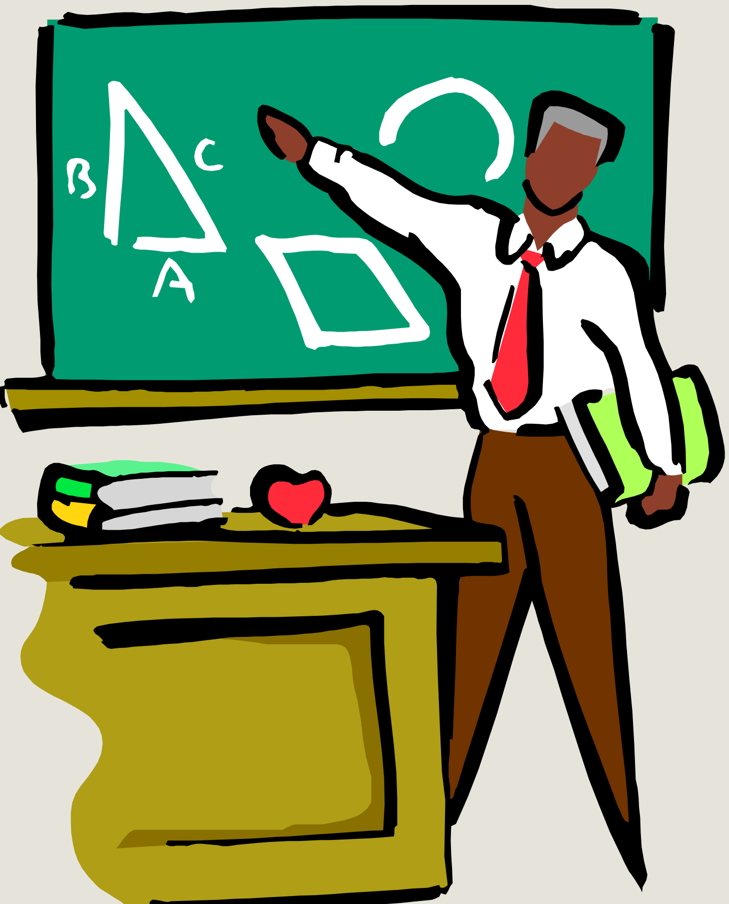


Megaco/H.248: Megaco IP phones

Phone considered as a media gateway …

- Terminations
  - User interface
  - Audio transducers
    - Hands free
    - Headset
    - Microphone
- Interactions
  - Add
  - Move
  - Subtract
  - Modify

Soft-switches



1. Introduction

2. Overall Architecture

3. Use Cases

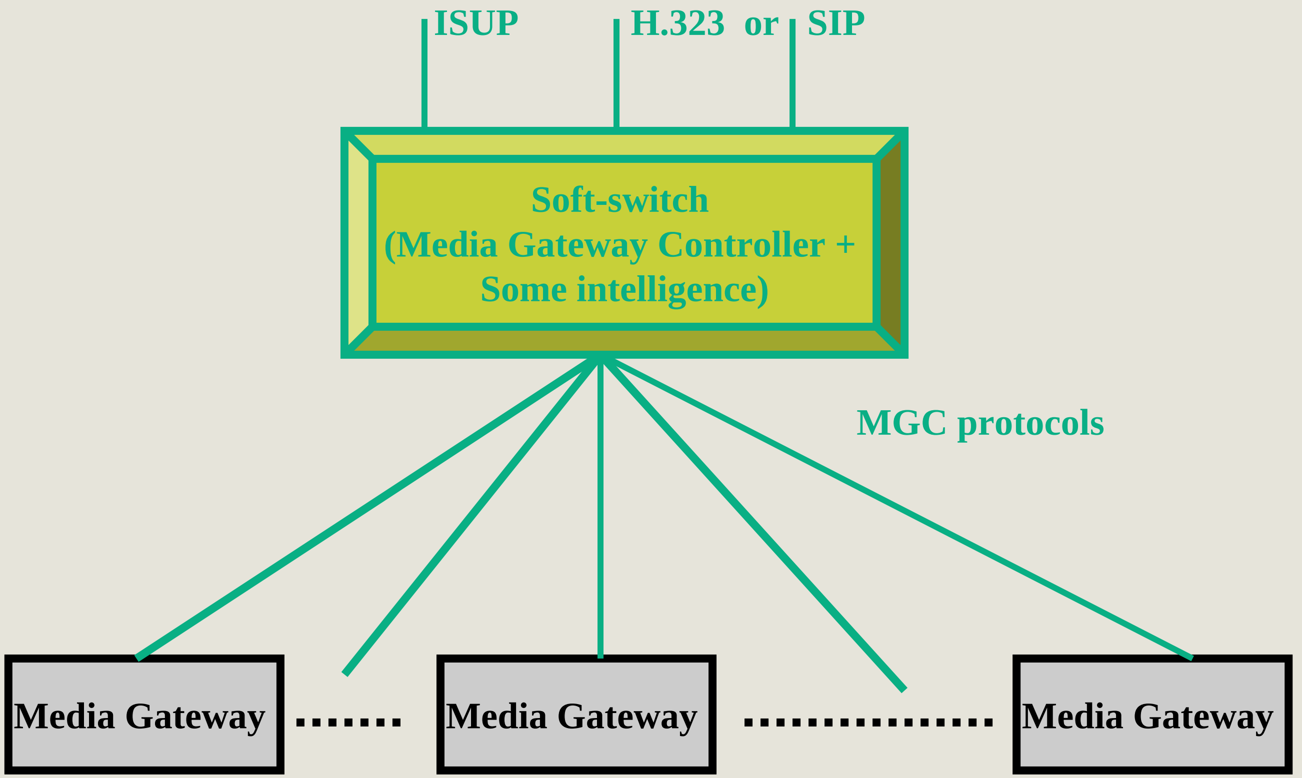
3. A call case

Introduction

A "side effect" of media gateway decomposition

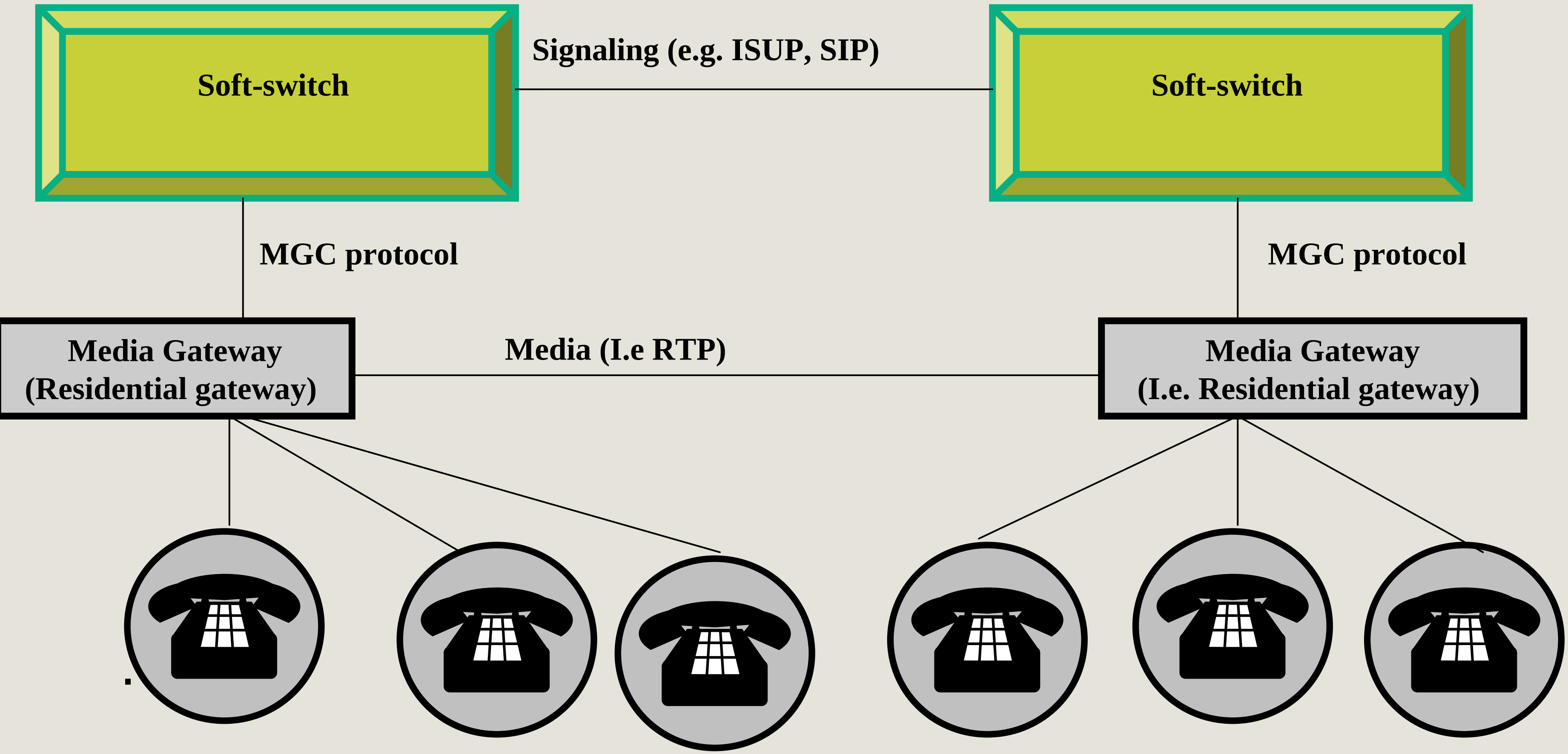
- Aggressively promoted by the soft-switch consortium, now known as the International Packet Communication Consortium (IPCC)
  - Adoption of existing standards (e.g. SIP, H.323, MGCP, Megaco)
- Gateway controller (plus some additional features) acts as a switch
  - Switching in software instead of hardware
- Can act as local exchange (class 5) or toll centre (class 4)
  - Lower entry costs for new incumbents
  - New local telephony networks and "by pass" for long distance call providers
- Soft-switches vs. classical switches debate
  - Scalability
  - Reliability
  - QoS

Overall Architecture



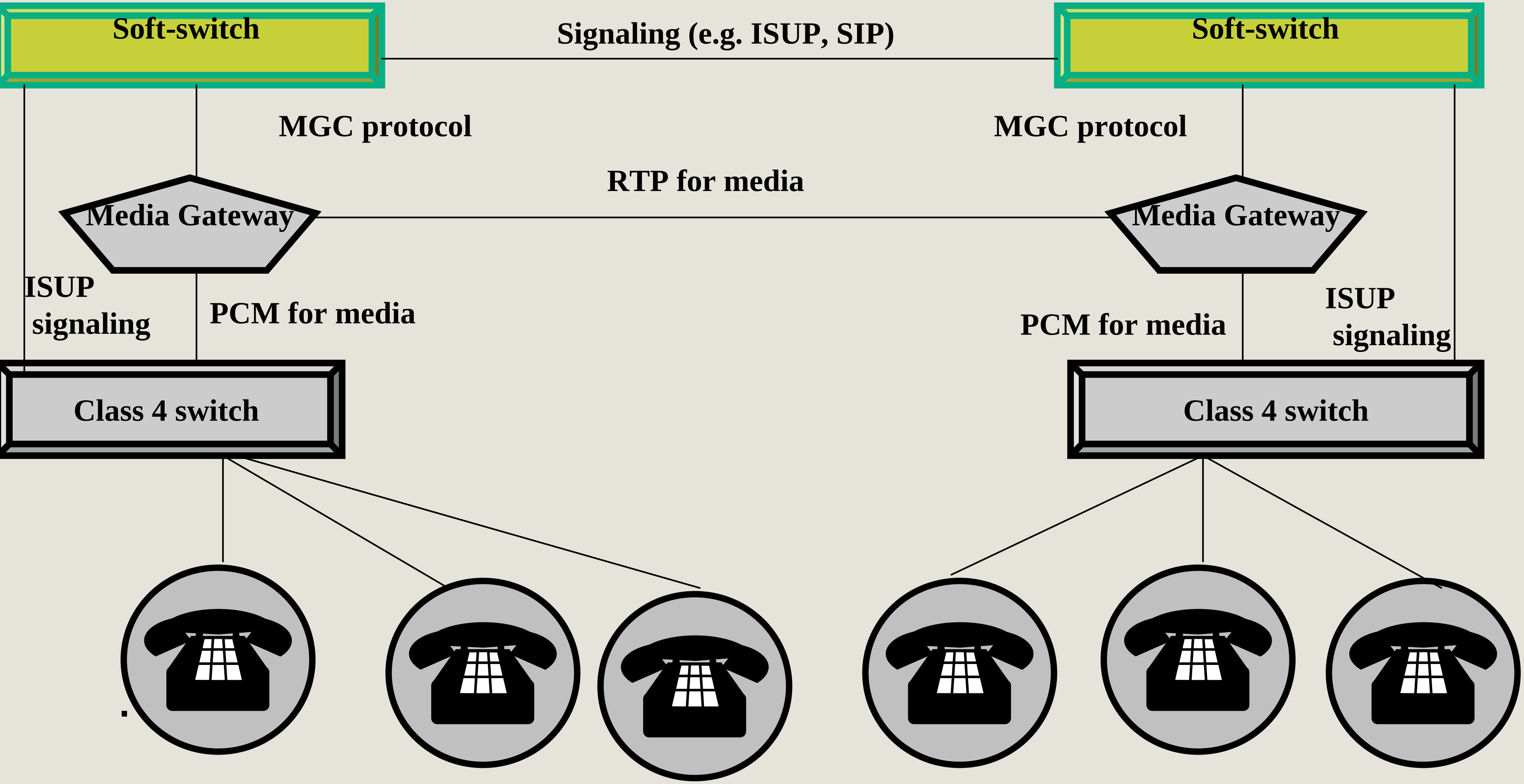
Use cases

An example of soft-switch as class 5 replacement …

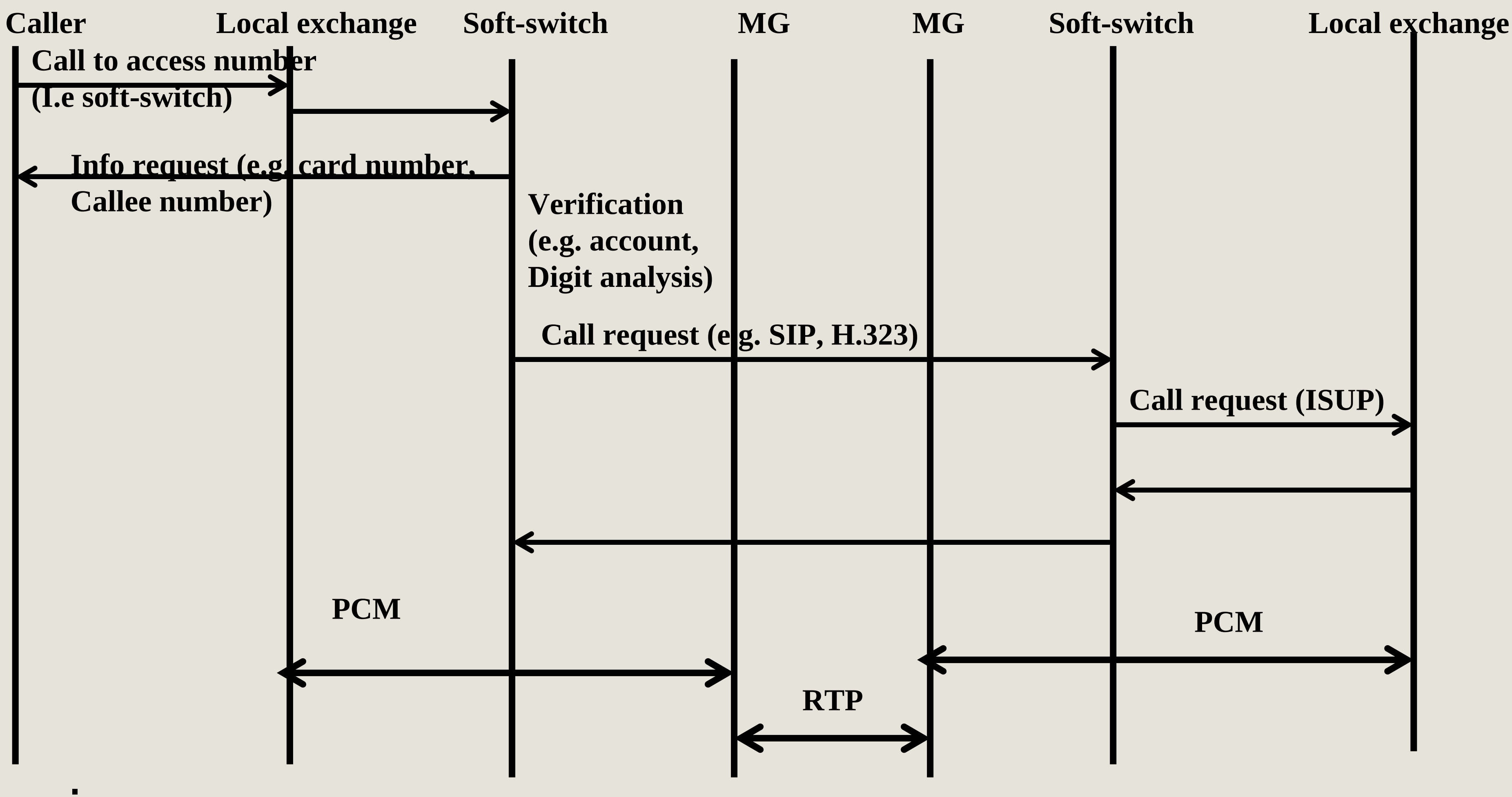


Use cases

An example of soft-switch as class 4 replacement …



Soft-switch: A simplified call case (Calling card)



To probe further ...

The network layer of NGN

- H. Schulzrinne, Converging on Internet Telephony, Special issue, IEEE Network / IEEE Internet Computing, May/June 1999
- A. Moderassi and S. Mohan, Advanced Signaling and Control in Next Generation Networks, Special issue IEEE Communications Magazine, October 2000

SIP

http://www.cs.columbia.edu/sip/

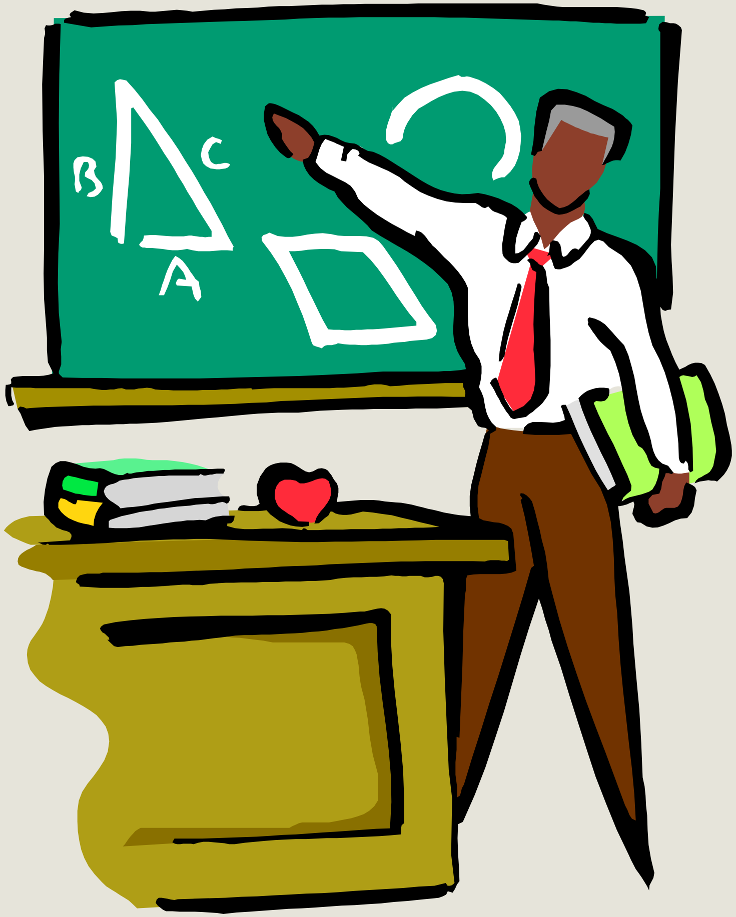
3GPP / 3GPP2 networks

- 3GPP / 3GPP2 specifications
- Lieve Bos, Suresh Leroy, "Toward an all-IP-based UMTS system
- architecture", IEEE Network, no. 1, January/February 2001

Part I: The service layer

- **Basics**
- **Signaling protocol specific approaches**
- **Signaling protocol neutral approaches**

Basics



1. Services
2. Service Life Cycle
3. Service Engineering

Services ...

What are value added services (or simply services)?

- **Anything that goes beyond two party voice call**
  - Telephony services (interact will call control)
    - Call diversion, screening, split charging
    - dial in/dial out conferencing, multiparty gaming …
  - Non Telephony services (Do not interact with call control)
    - Web access
      - » Customised stock quotes
      - » Surfing from a cell phone …
  - Combination
    - Call centers

ERICSSON

Service life cycle ...

Four phases

- Creation (also known as construction)
  - Specification, design/coding, and testing
- Deployment
  - Service logic (or executable) resides on specific node(s) and needs to be deployed there
- Usage
  - Subscription/billing, triggering, features interactions
- Withdrawal
  - Removal from network

Service Engineering ...

Key issue: How to engineer "cool" services

- **In more academic terms**
  - Issues related to the support of all the phases of the life cycle.
    - Creation
    - Deployment
    - Usage
    - Withdrawal
  - These issues are architectural issues
    - Concepts, principles, rules
    - Functional entities, interfaces and algorithms

Service Engineering ...

Why is it an important discipline?

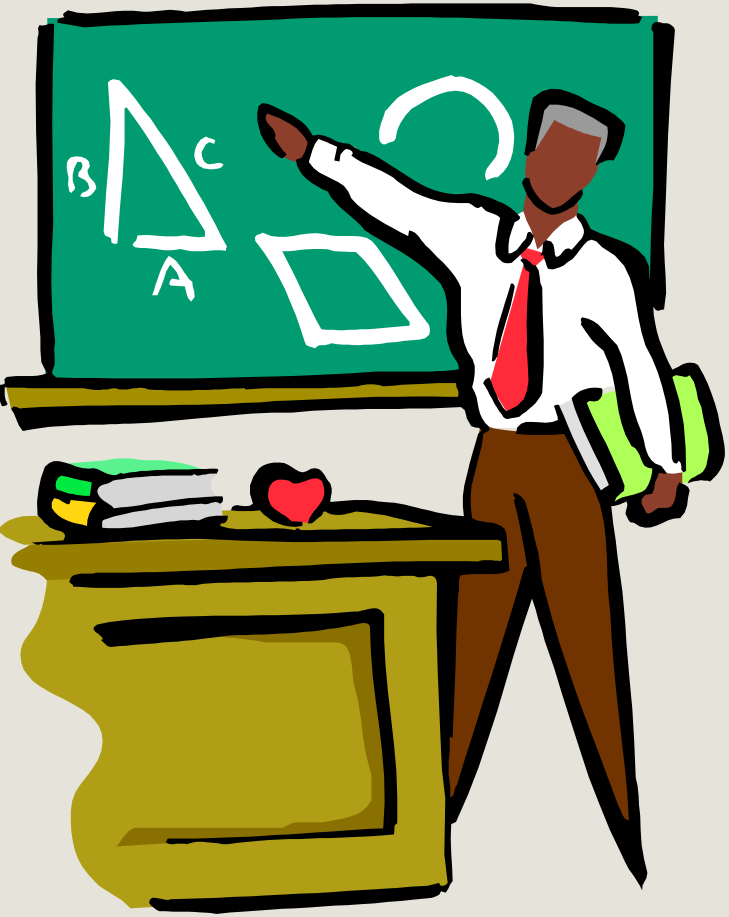
- **Business standpoint**
  - High quality two party voice call is now a commodity
  - Value added services are needed to attract subscribers and generate revenues.
- **Engineering standpoint**
  - It is less than trivial
  - Example: Service creation
    - Secure and selective access to network resources is required
    - Related issues: Level of abstraction, security framework, service creation tools …etc.

Service Engineering ...

How is it done in today's circuit switched telephony networks?

- **Intelligent Networks (IN)**
  - Focus on telephony services
  - Separate switching from service logic
    - Service switching point (SSP)
    - Service control point (SCP).
- **Wireless Access Protocol (WAP)**
  - Focus on non telephony services (wireless access to Internet)
    - WAP 1.0: Rely on a set of home designed protocols
    - WAP 2.0: Rely on a wireless profile of Internet protocols (e.g. WP TCP, WP HTTP)

SIP Servlet API / SIP AS



1. Introduction
2. SIP servlet API
3. Examples
4. Pros and cons

Introduction ...

Key features

- Signalling protocol specific (I.e. applicable to SIP only)
- Prime target: trusted parties
  - Service providers
  - Third party developers
- Very few constraints on what can be done
- Reliance on SIP servlet API
  - SIP servlet API is widely used in the Internet world
    - A tool which relies on it should attract many users including Web masters.
    - A wide range of developers should favour the development of cool and brand new services

.

SIP servlet API...

Adjustments made to HTTP servlet:

- Initiate requests
  - Needed for some services
    - wake up call
- Receive both requests and responses
  - Needed for some services
    - Terminating services (e.g. call forward on busy)
- Possibility to generate multiple responses
    - Intermediary responses, then final response
- Proxying requests, possibly to multiple destinations
  - Needed for applications such as intelligent routing

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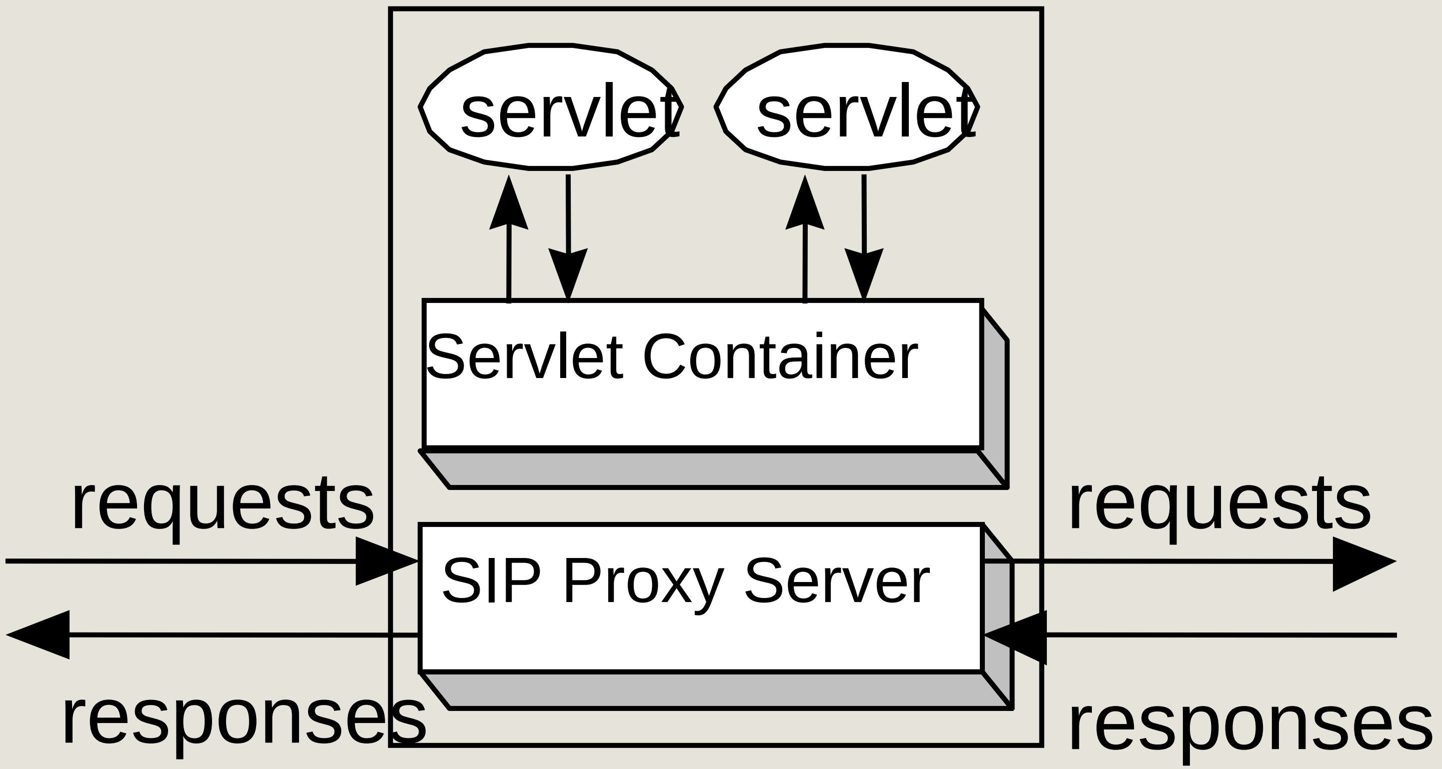
SIP Servlet container ...

A container collocated with a proxy server



SIP servlet Request/response hierarchy...

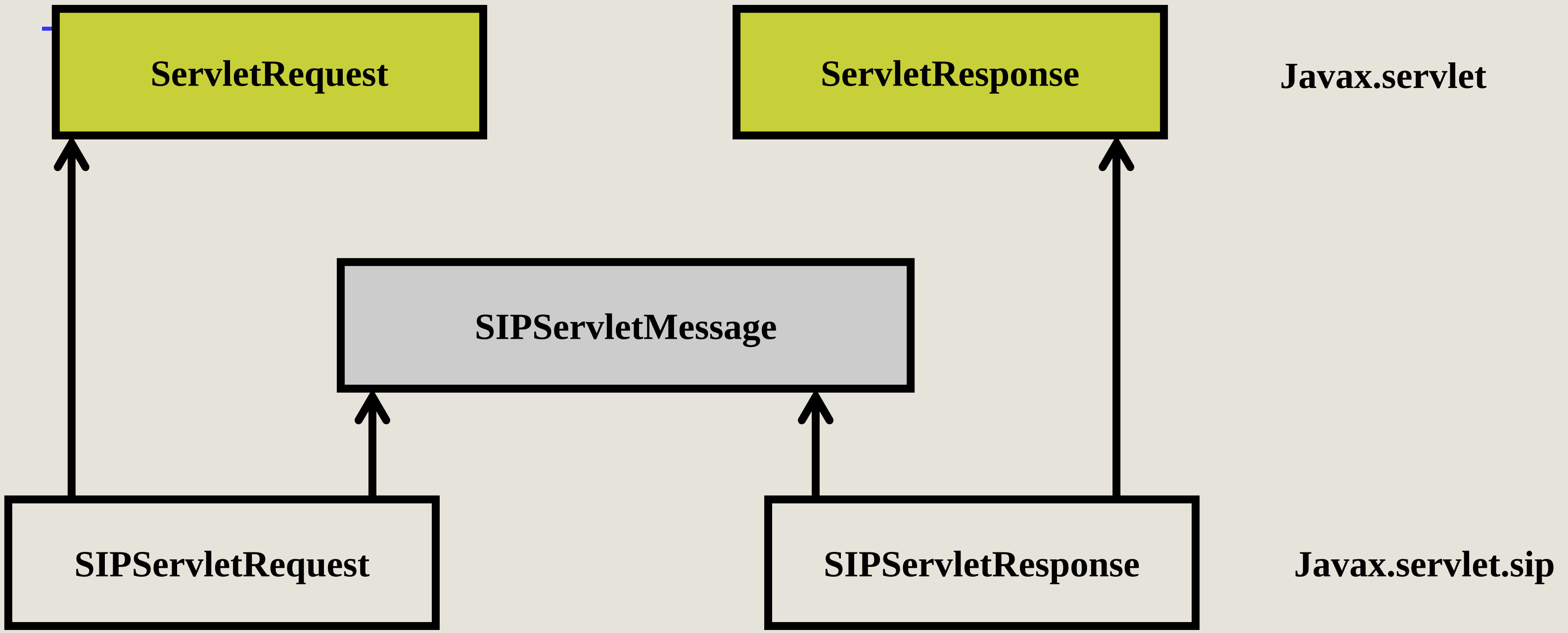
Build on the generic servlet API, like HTTP servlet

- javax.servlet.sip (just like javax.servlet. http)
- Container must support
  . javax.servlet
  . Javax.servlet.sip

.

SIP servlet request response hierarchy...

Request-response Hierarchy



SIP servlet Request interface ...

SIP specific Request handling methods (Based on both core SIP and SIP extensions):

- doInvite
- doAck
- doOptions
- doBye
- doCancel
- doRegister
- doSubscribe
- doNotify
- doMessage
- doInfo

SIP servlet Response interface ...

SIP specific Response handling methods (Based on both core SIP and SIP extensions):

- doProvisionalResponse
- doSuccessResponse
- doRedirectResponse
- doErrorResponse

SIP servlet Message interface ...

SIP specific message handling methods (Access to message header):

- getHeader
- getHeaders (Used when there are several headers)
- setHeader
- addHeader

Note: system headers cannot be manipulated by servlets (Call-ID, From, To)

SIP servlet Message interface ...

SIP specific message handling methods (Access to message content):

- getContentLength
- setContentLength
- getContentType
- getContent
- getRawContent
- setContent

An example of service:

Algorithm for call forward

- Get the destination from the SIP request
  - Done by retrieving the To_Field by using the GetHeaders
- Obtain the forwarding address from a data base
- Forward the call
  - Done by setting the Request_URI (and not the To_field) using the setHeader

.

Another example:

Algorithm for a centralized dial-out conference

Assumptions

- INVITE is used
- URIs of participants are put in the INVITE body

Agorithm used in servlet:

- Use GetContent to get the participant's URIs from INVITE Request
- Use doINVITE to generate and send an INVITE to each participant.

Pros and cons ...

Pros

- Possibility of creating a wide range of services due to the full access to all the fields from the SIP Request
- More performance and more scalability
- Possibility to create services that combine both HTTP and SIP

Cons:

- Require detailed knowledge of SIP
  - Some application developers may not have that expertise
    - Issue faced by all signalling protocol specific approaches
- SIP Servlet is not exactly the same thing as HTTP Servlet
  - There will be less developers from the Internet world than expected
- Language dependence

.

To probe further ...

H.323 specific approaches

ITU-T specifications: H.450.x series, recommendation H.323

- H. Liu and P. Mouchtaris, Voice over IP Signalling: H.323 and Beyond, IEEE Communications Magazine, October 2000, Vol. 38 No 10

- R. H. Glitho, Advanced Services Architectures for Internet Telephony: A Critical Overview, IEEE Network, July 2000, pp. 38-44

SIP specific approaches

IETF RFCs: SIP CGI - RFC 3050

- J. Rosenberg, J. Lennox and H. Schulzrinne, Programming Internet Telephony Services, IEEE Network, May/June 1999, Vol.13, No3, pp. 42-49

- R. H. Glitho, Advanced Services Architectures for Internet Telephony: A Critical Overview, IEEE Network, July 2000, pp. 38-44

Java Developers Community draft JSR 116

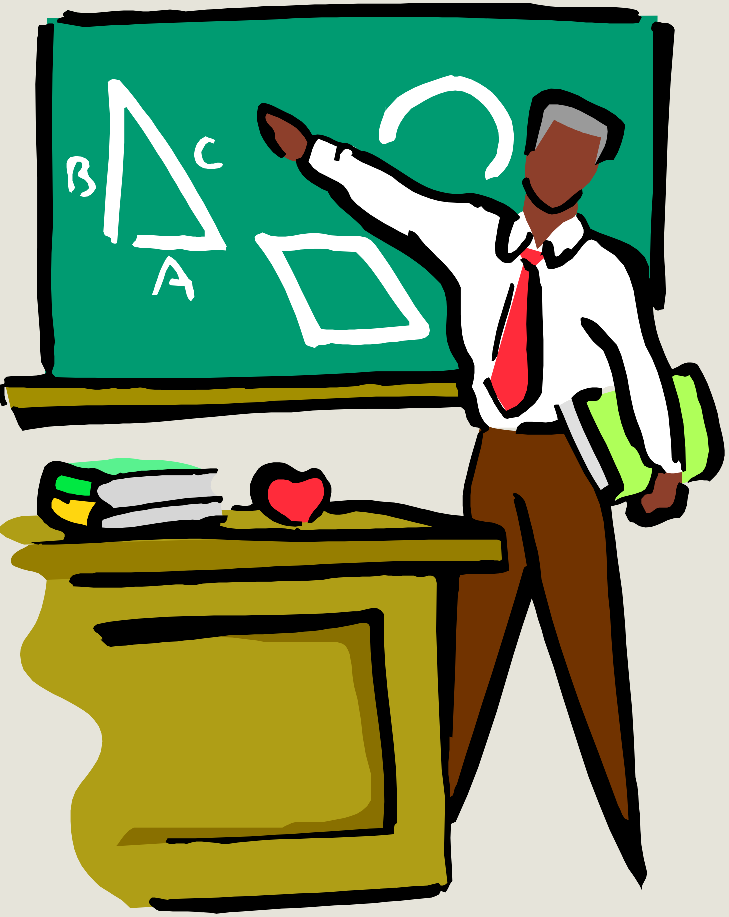
http://www.jcp.org/aboutJava/communityprocess/review/jsr116/

W. Leekwijck and D. Brouns, SIPlets: Java Based Service Programming for IP Telephony

R. Glitho, R. Hamadi and R. Huie, Architectural Framework for Using Java Servlets in a SIP environment, ICN2001, July 2001

3GPP TS 23.228, 23.002, http://www.3gpp.org/

OSA/PARLAY



1. Introduction
2. Business model
3. APIs
4. Pros and cons

Introduction

PARLAY forum

- Created in 1998 as close forum
- Open since 2000
- Include most major players from telecommunications and computer industries (e.g. Ericsson, Lucent, Siemens, IBM)
- Fourth release of specifications recently released

Relationship of Parlay specifications to 3GPP specifications

- API called Open Service Access (OSA) in 3GPP
  - Thus Parlay/OSA
  - Joint development

Introduction

OSA a tool kit of Virtual Home Environment (VHE)

- VHE
  - 3GPP concept for service mobility
    - **Access to services from any location and with any terminal (within the limit of the terminal capabilities)**
    - **Include several tool kits:**

OSA allow third party to access 3GPP next generation networks

- OSA application servers
  - reside in third party domains
  - Access 3GPP network functionality via service capability servers (SCS) (I.e. gateways)

Introduction

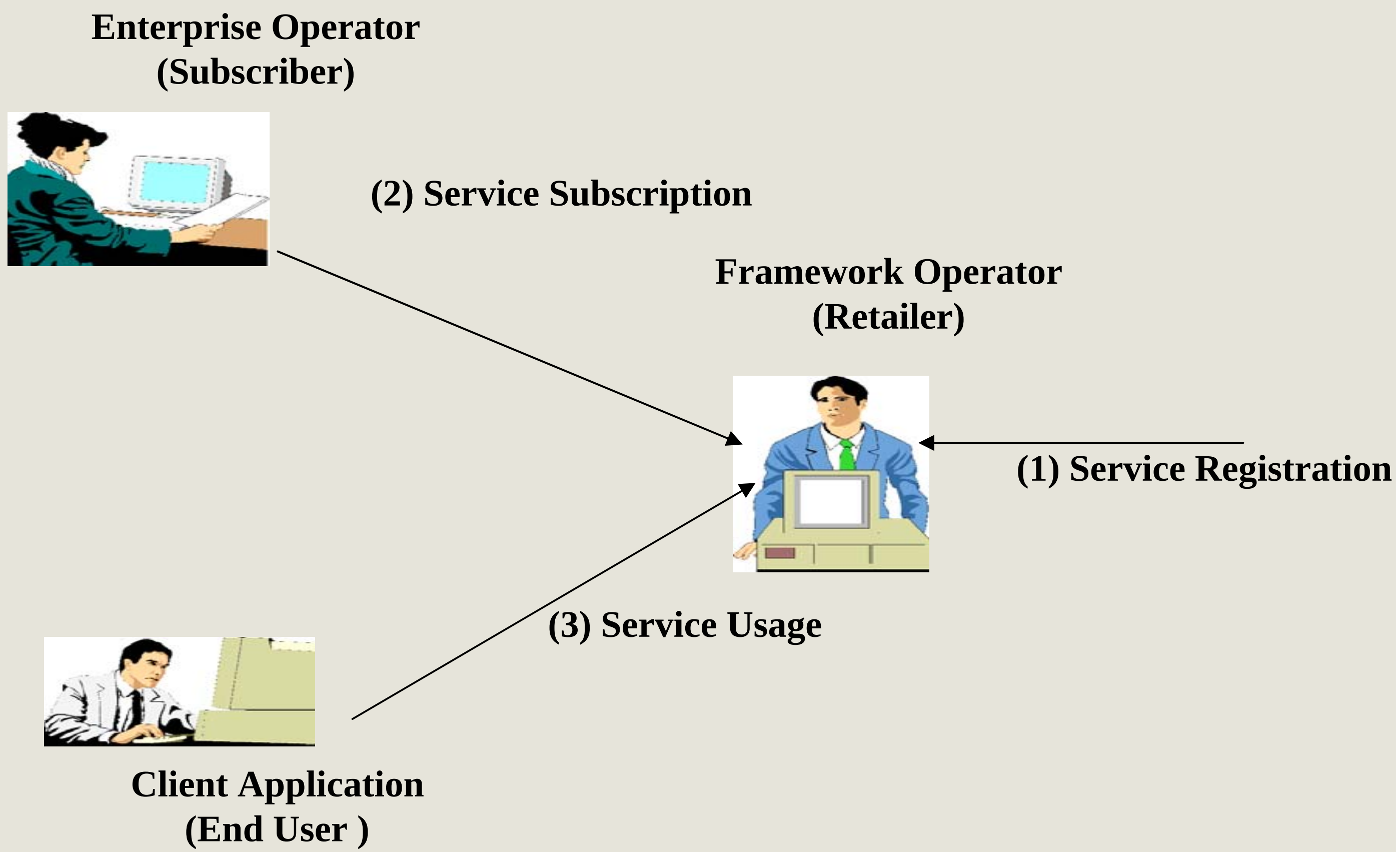
PARLAY main goal: Open up telecommunication networks

- Enable new business models
- Use open information technology middleware
- Make telecommunication network capabilities available for application development
  - Two types of APIs
  - Services APIs
    - Expose the network capabilities (e.g. call control, presence)
  - Framework APIs
    - Make the use of the service APIs secure, accountable and resilient (e.g. security, registration, authentication)

The business model

- Introduction
  - TINA-C inspired business model
  - Terminology: Services mean network capabilities
- Roles
  - Client application
    - Consume/use the services (e.g. network capabilities)
    - Equivalent to end users in TINA-C.
  - Enterprise operator
    - The entity that subscribes to the services
    - Subscriber in TINA-C
  - Framework operator
    - Entity that handles the subscriptions
    - Equivalent to the retailer in TINA-C

The business model



The APIs

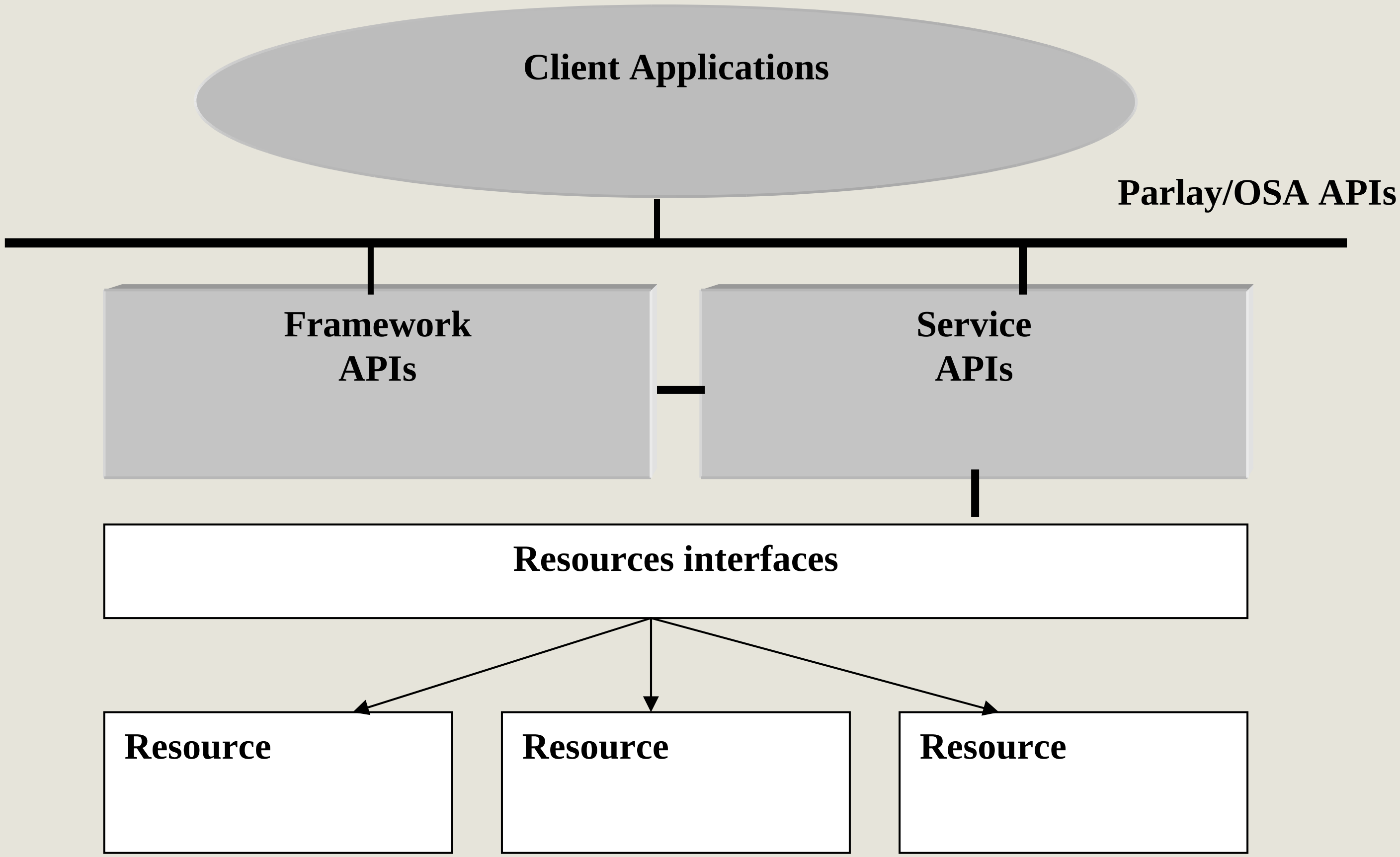


Figure 2 Parlay APIs interfaces

The APIs

Some common characteristics

Specifications include

- High level specification in UML (Universal Modelling Language)
- API specifications for several IT technologies
  - CORBA IDL
  - WSDL
  - Java

Two modes of communications

- Synchronous
- Asynchronous

Service API: Give access to network capabilities

- Call control
- User interactions
- Generic messaging
- Mobility
- Terminal capabilities
- Connectivity management
- Account management
- Charging service
- Data session control
- Presence and availability management
- Integrity management

Framework API: Make the use of the service APIs secure and resilient

Trust and security management

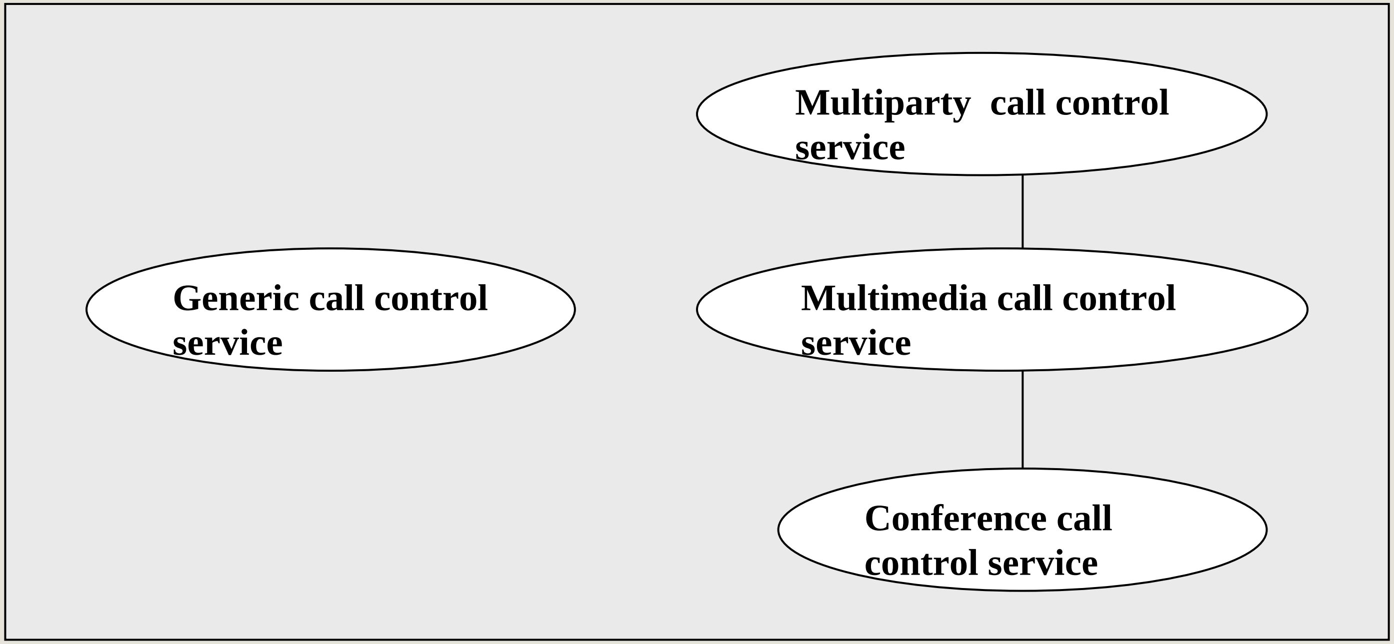
Event notification

Service discovery

Service registration

Integrity management

An example of Service API: Call control



The call control API

Call model

- Terminal
  - End point (Not covered in the current specifications)
- Address
  - Represents a party in a call (E.164 number, IP address)
- Call
  - Abstraction of the physical call that occurs in a network
- Call leg
  - Logical association between a call and a party involved in a call

The call control API

Generic call control

- Two party voice call only
- Remain in Parlay for historical reasons

Multiparty call control

- Establishment of calls with any given number of users
- Root of the inheritance tree

Multimedia call control

- Add multimedia (e.g. media negotiation) capabilities

Conference call control

- Add conferencing capabilities
  - Creation and manipulation of sub-conferences

Pros and cons ...

Pros

- PARLAY/OSA allows the creation of a wide range of services including services that combine different types of network capabilities (e.g. call control, mobility, presence)
- Parlay allow the creation of services that span several network technologies (e.g. Sip, H.323, 3GPP, soft-switches)

Cons

- The level of abstraction is still low
  - 3N+1 calls were required to create a conference call in older versions of Parlay – The number is now N+1
- Parlay is not easy to grasp by people with no circuit switched telephony/IN background
  - Call leg concept

To probe further on Parlay ...

PARLAY:

PARLAY specifications, http: //www.parlay.org/

R. Glitho and K. Sylla, Developing applications for Internet Telephony: A case study on the use of Parlay call control APIs in SIP networks, IEEE Network Magazine, May/June 2004

A.J. Moerdijk and L. Klostermanns, Opening the networks with Parlay/OSA: Standards and Aspects behind the APIs, IEEE Network Magazine, May/June 2003

S. Bessler, A.V. Nisanyan, K. Peterbauer, R. Pailer and J. Stadler, A Service Platform for Internet-Telecom Services Using SIP, SmartNet 2000.

S. Desrochers, R. Glitho, K. Sylla, Experimenting with PARLAY in SIP Environment: Early Results, IPTS, Atlanta, July 2000 -

N. Edwards, SIP Support in PARLAY, Btexact, www.btexact.com

N. Edwards, PARLAY and SIP, Issue 1, British Telecoms plc (2000)

R. Glitho, A. Poulin, K. Sylla, and O. Cherkaoui, Using PARLAY for Centralized Conferences in SIP Environment, ICIN 2001, October 2001

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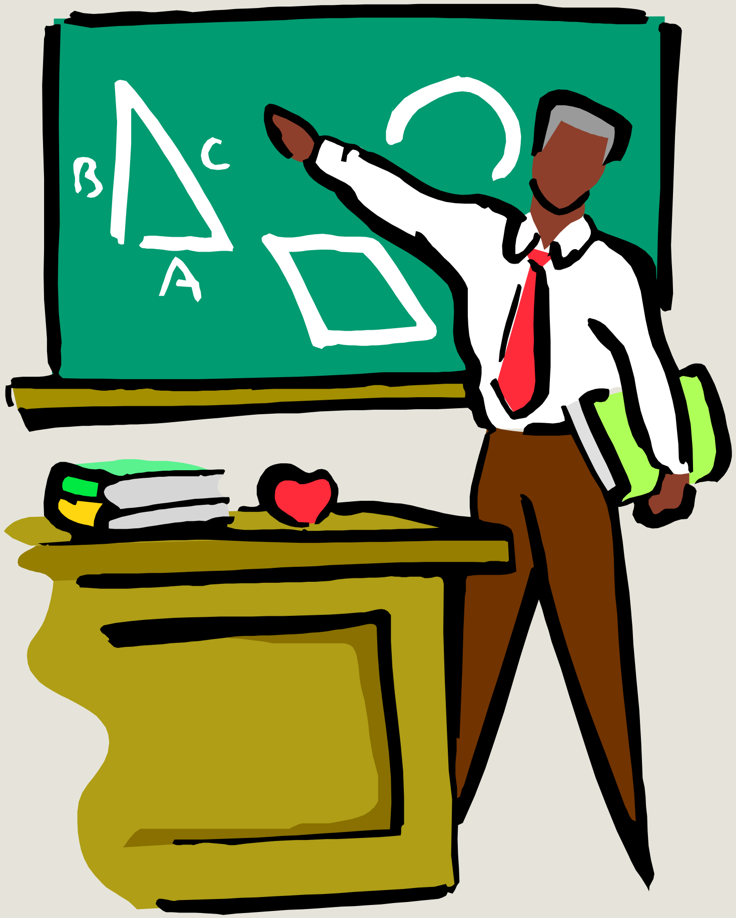
ERICSSON

Web services …



1. Basics
2. Architecture
3. Application Areas

Basics

Today		Tomorrow
• Publication of documents	XML Technology →	• Publication of "reusable business logic"
• Human interaction	→	• Automated P2P interaction
• Proprietary ad-hoc interfaces	→	• Industry standard interfaces

Basics

"The term Web Services refers to an architecture that allows applications (on the Web) to talk to each other. Period. End of statement"

Adam Bobsworth in ACM Queue, Vol1, No1

The three fundamental principles, still according to Adam Bobsworth:

1. Coarse grained approach (I.e. high level interface)
2. Loose coupling (e.g. application A which talks to application B should not necessarily be re-written if application B is modified)
3. Synchronous mode of communication, but also asynchronous mode

Basics

Some of the involved standards bodies / Consortia

- **Architectures and Technologies**
  - World Wide Web Consortium (W3C)
    - Interoperable technologies for the Web
  - OASIS
    - Structured information standards that permit interoperability
  - Liberty Alliance
    - Open standards for federated network identities (pertinent to Web service security)

Basics

Some of the involved standards bodies / Consortia

Application to specific areas

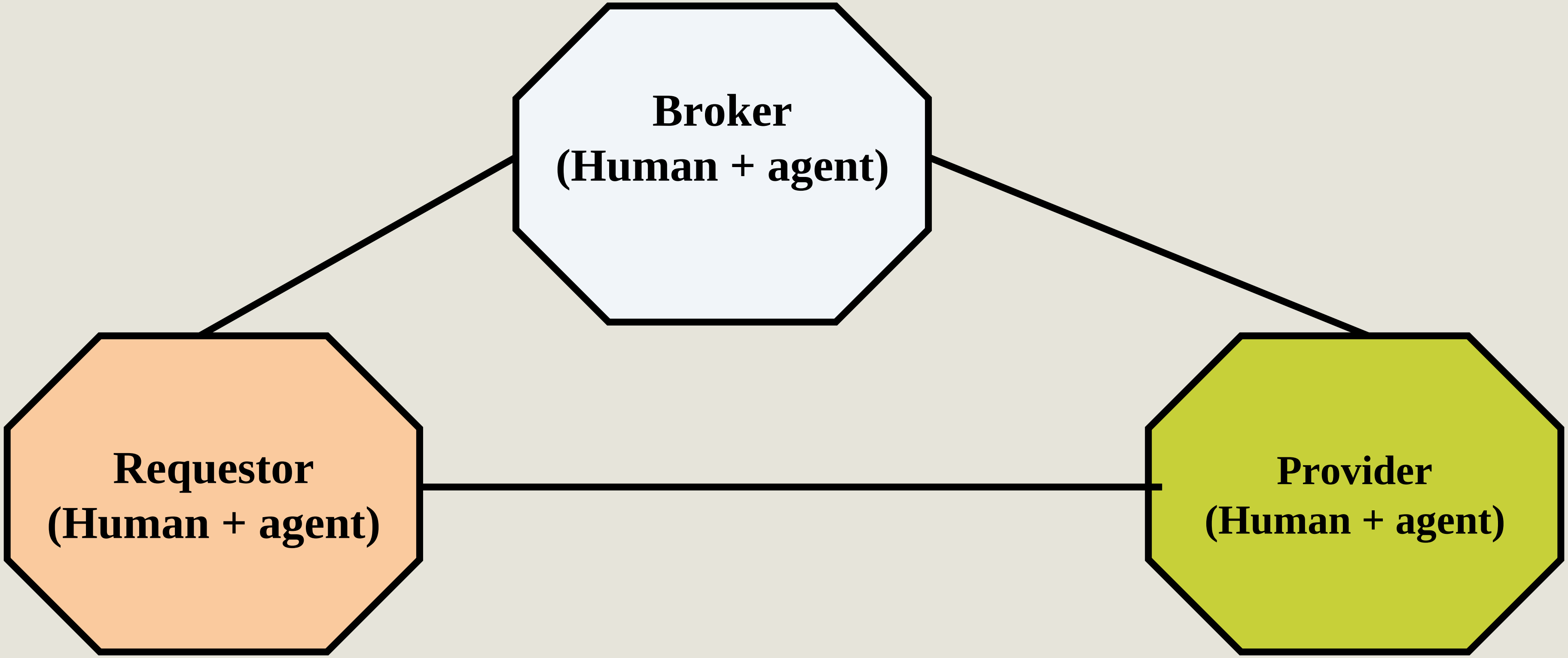
Telecom

- Parlay-X
- Open Mobile Alliance (OMA)

Digital images

- International Imagery Association

Architecture



Entities

Requestor

- Person or organization that wishes to make use of a Web service.
- Uses an agent (I.e requestor agent) to exchange messages with both broker agent and provider agent.

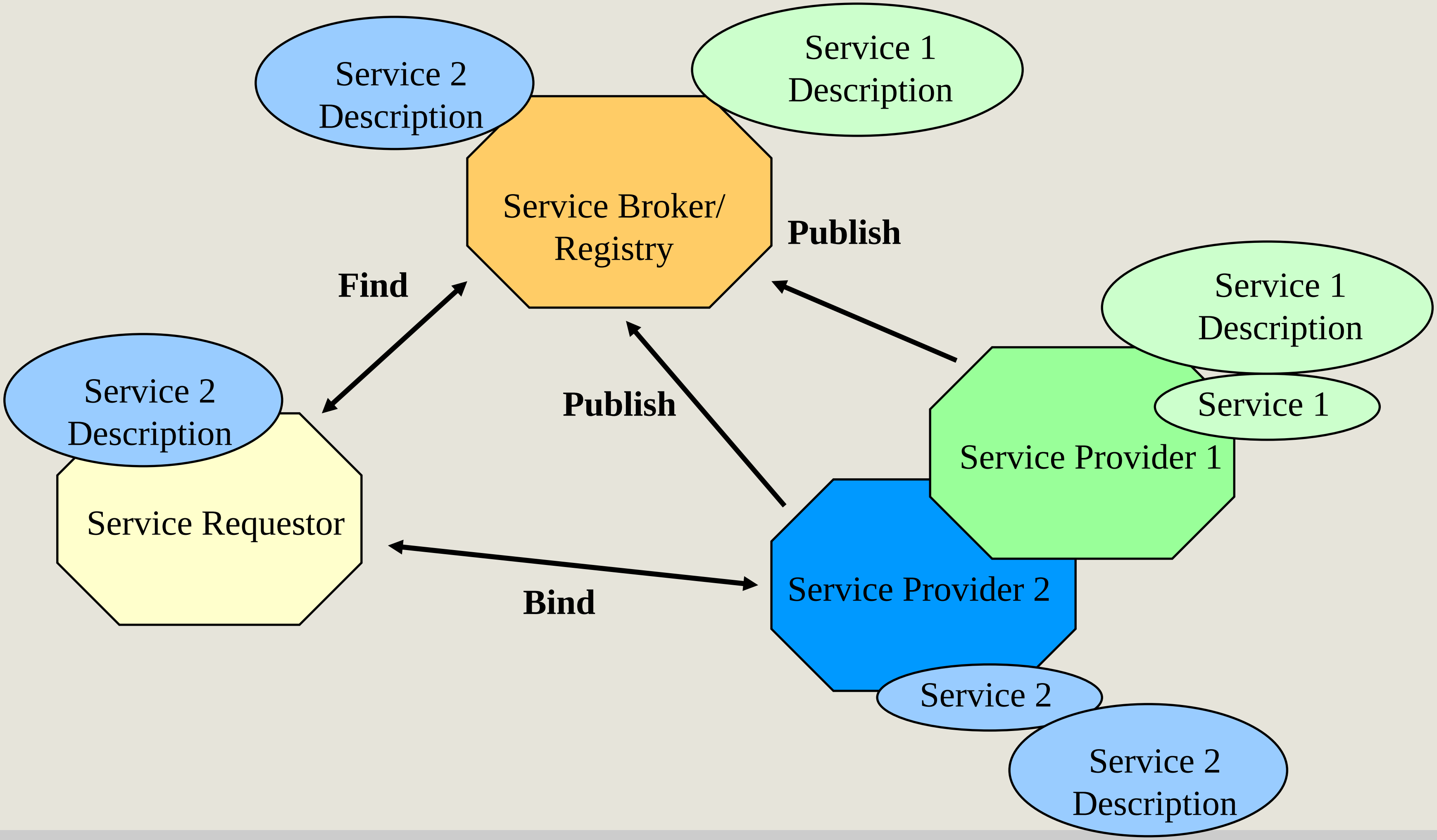
Provider

- Person or organization that owns a Web service it wants to make available for usage
- Use an agent (I.e provider agent) to exchange messages with broker agent and requestor agent.
- The provider agent is also the software piece which implements the Web service (e.g. mapping towards legacy)

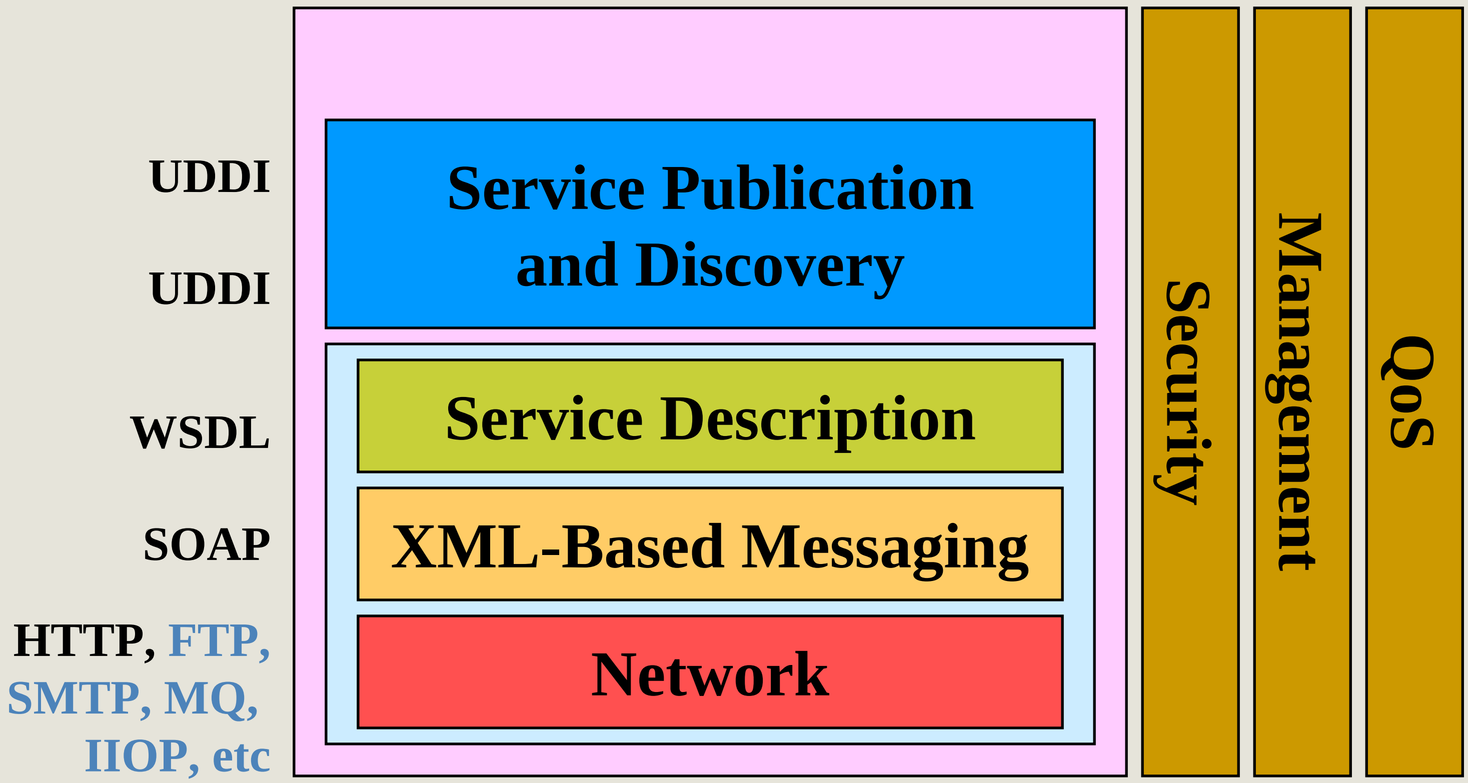
Broker

- Person or organization that puts requestors and providers in contact
  - Providers use brokers to publish Web services
  - Requestors use brokers to discover Web services
- Use an agent (I.e broker agent) to exchange messages with requestor agent and provider agent

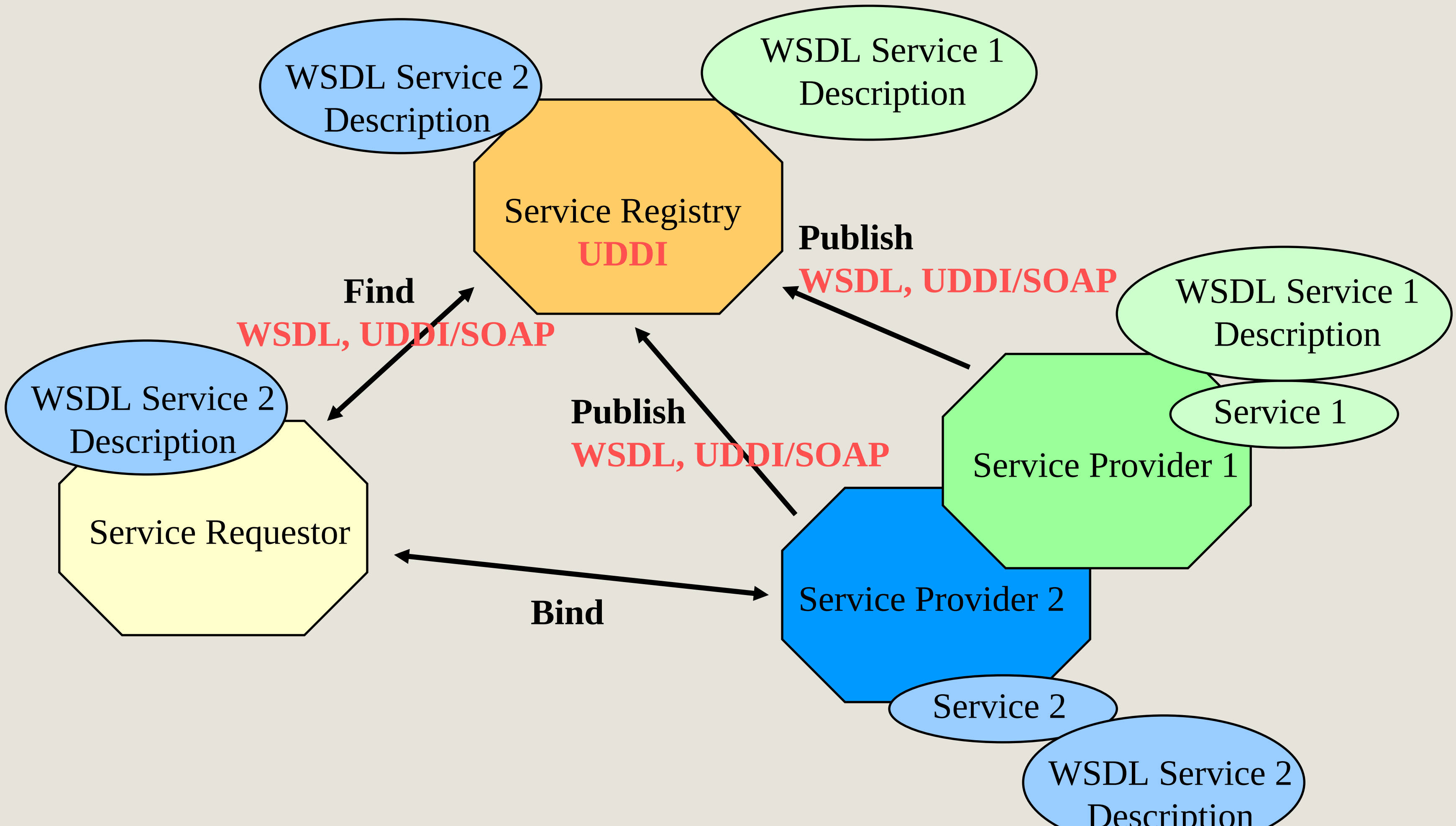
Interactions ...



Interactions



Interactions ...



Use in NGN service layer …

Motivation: Tackle the fundamental shortcomings of today's service architectures with a single technology

- Raise the level of abstraction
  - This level is inherently low for signalling protocol specific architectures
  - It is low for existing signalling protocol neutral architectures
- Use a technology widely known in Internet community
  - Parlay relied initially on CORBA/DCOM

Use in NGN Service Layer …

1. First issue: Define Web services for making telecommunications capabilities available to applications in same or foreign domain

- Call control
- Presence
- Location
- Messaging

Use in NGN Service Layer …

Second issue: Enable the use of Web services in telecommunications by providing

common / supporting functions such as:

Billing

Security -

- Authentication
- Authorization
- Non repudiation
- Others

Service management

- registration
- Discovery
- Others

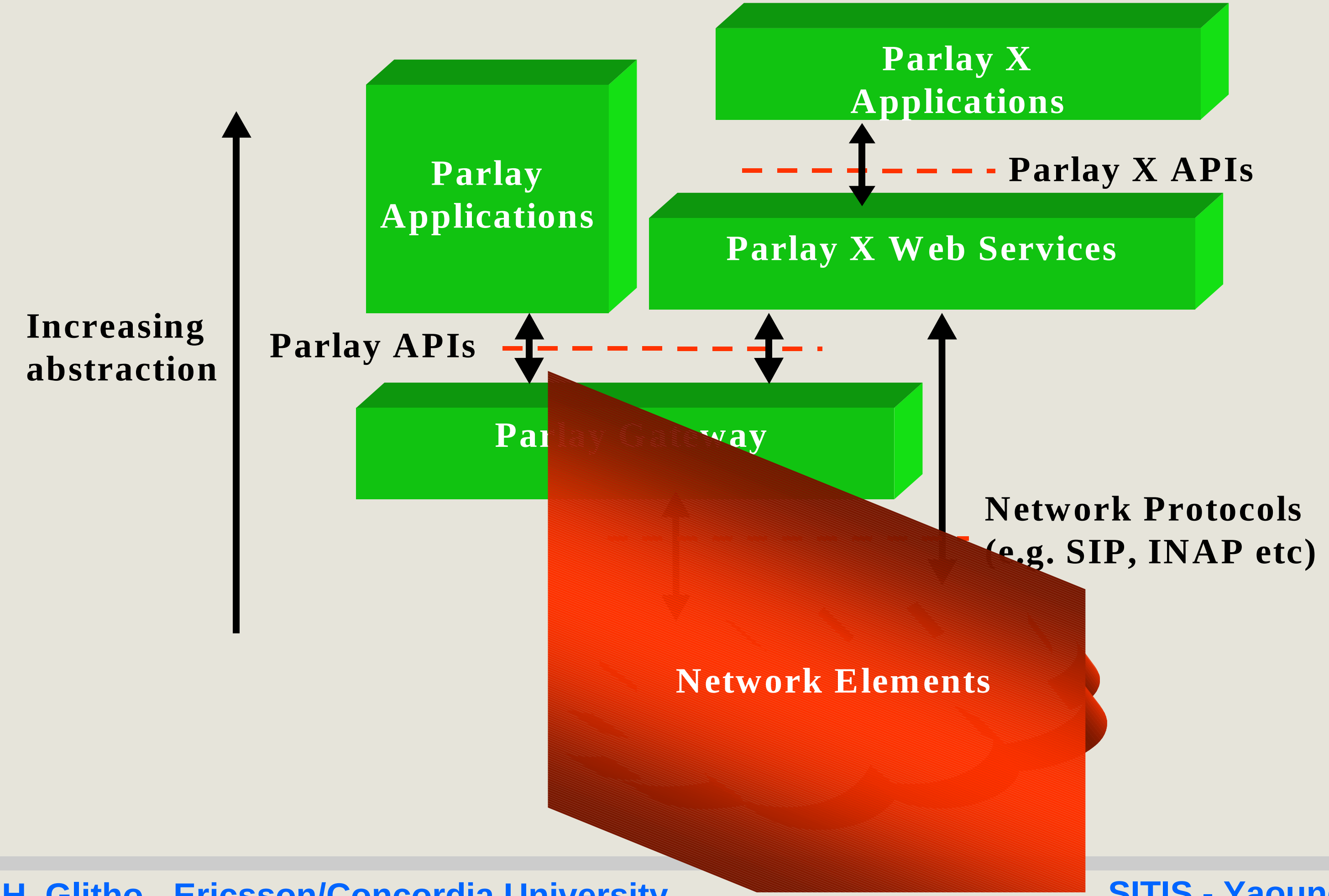
Use in NGN Service Layer: Parlay-X

1. **Specifications available in their first version**
   - White paper + actual specifications
   - Released as part of Parlay 4.1 specifications

2. **Application interfaces**
   - Focus: First issue
   - Aim at covering all telecommunication capabilities
     - Stand alone capabilities (e.g. presence, call control)
     - Combined capabilities (presence + call control)

3. **Use the reference Web service principles (e.g. coarse grained) technologies (e.g. WSDL)**

Parlay-X Architecture



Parlay-X Web Services

1. Call control
2. Messaging
   - SMS
   - MMS
3. Payment (e.g. volume charging)
4. Account management (e.g. account credit expiration date query)
5. User status (online / offline)
6. Terminal location

Parlay-X Call Control …

Make a call

Get call information

End call

Cancel call request

.

Parlay-X Call Control …

Handle busy

Handle Not reachable

Handle No answer

Handle off Hook

.

Parlay-X MMS …

- Send Message
- Get Message Delivery Status
- Get Received messages
- Get messages URIs
- Notify message reception

.

Use in NGN Service Layer: OMA

OMA

- Industry association created in 2002
- Focus on mobile services
- Aims at:
  - Consolidating standards for wireless services (e.g. 3GPP/PP2, IETF, W3C)
  - Producing new standards if needed-
  - Tackling the two issues

OMA

Aim at providing a general architecture for mobile services

- Requirements
- Principles
- Functional entities
- Common framework
- Service adaptability
- Consistency with Internet models

OMA principles

- Signalling protocol neutrality and independence from programming languages, operating systems and so on
- Leverage existing standards
- Interoperability, scalability
- Service adaptability
- Consistency with Internet models

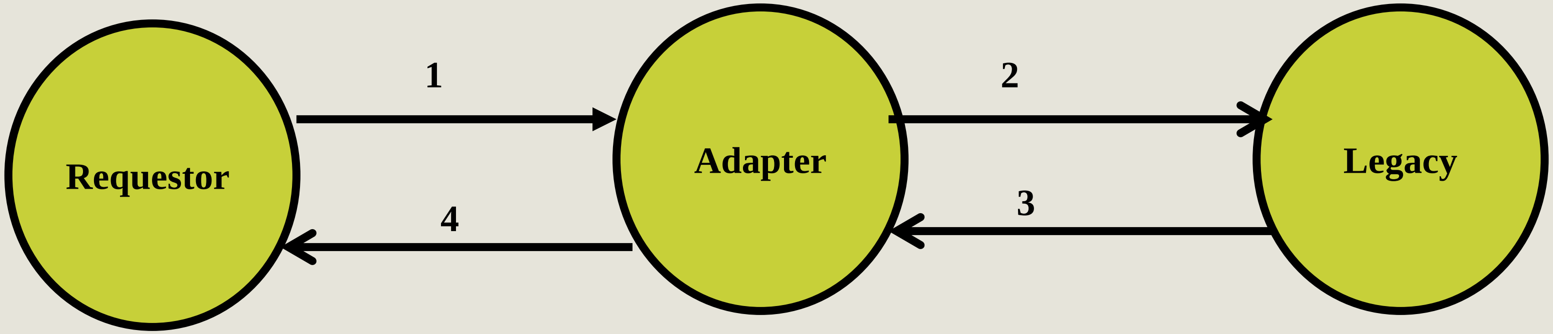
OMA Web service enabler (OWSER)

Aim at providing solutions to common problems faced by designers when using Web services in an OMA environment

- Practical deployment patterns
- Common functions (e.g. charging, security)
- Network Identity specifications (I.e. specific aspects of security – Based on Liberty alliance specifications)
- WSDL Style guidelines
- Test requirements

Examples of deployment patterns

The adapter pattern



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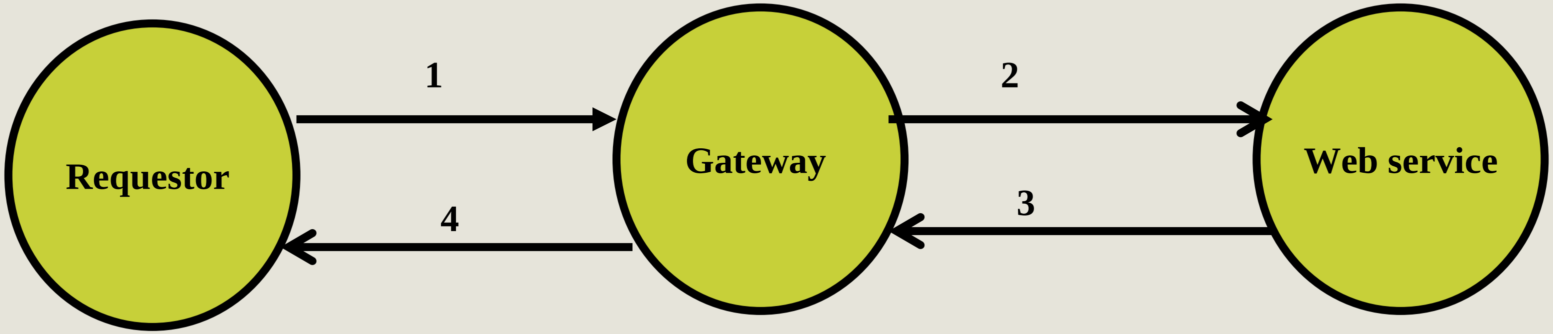
Examples of deployment patterns

The gateway pattern



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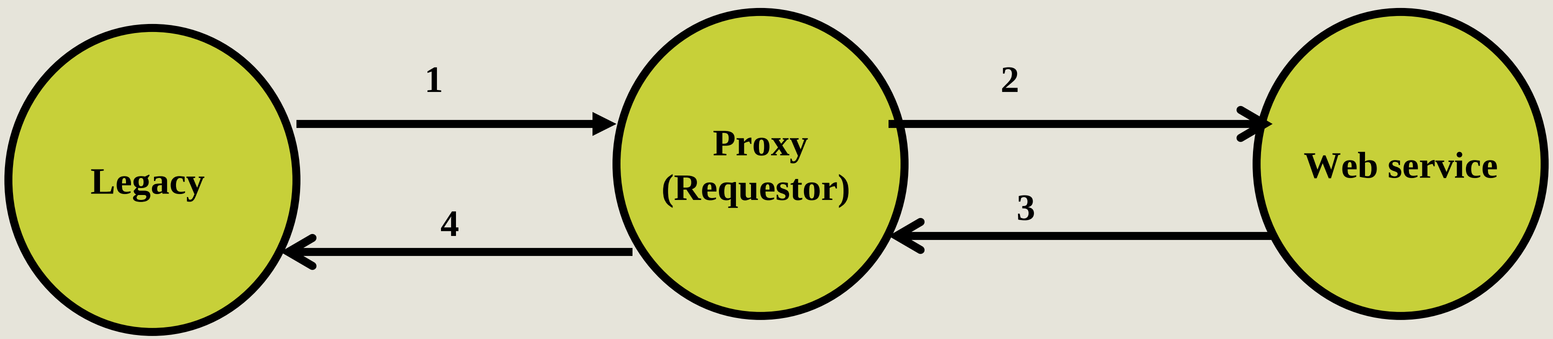
Examples of deployment patterns

The proxy pattern



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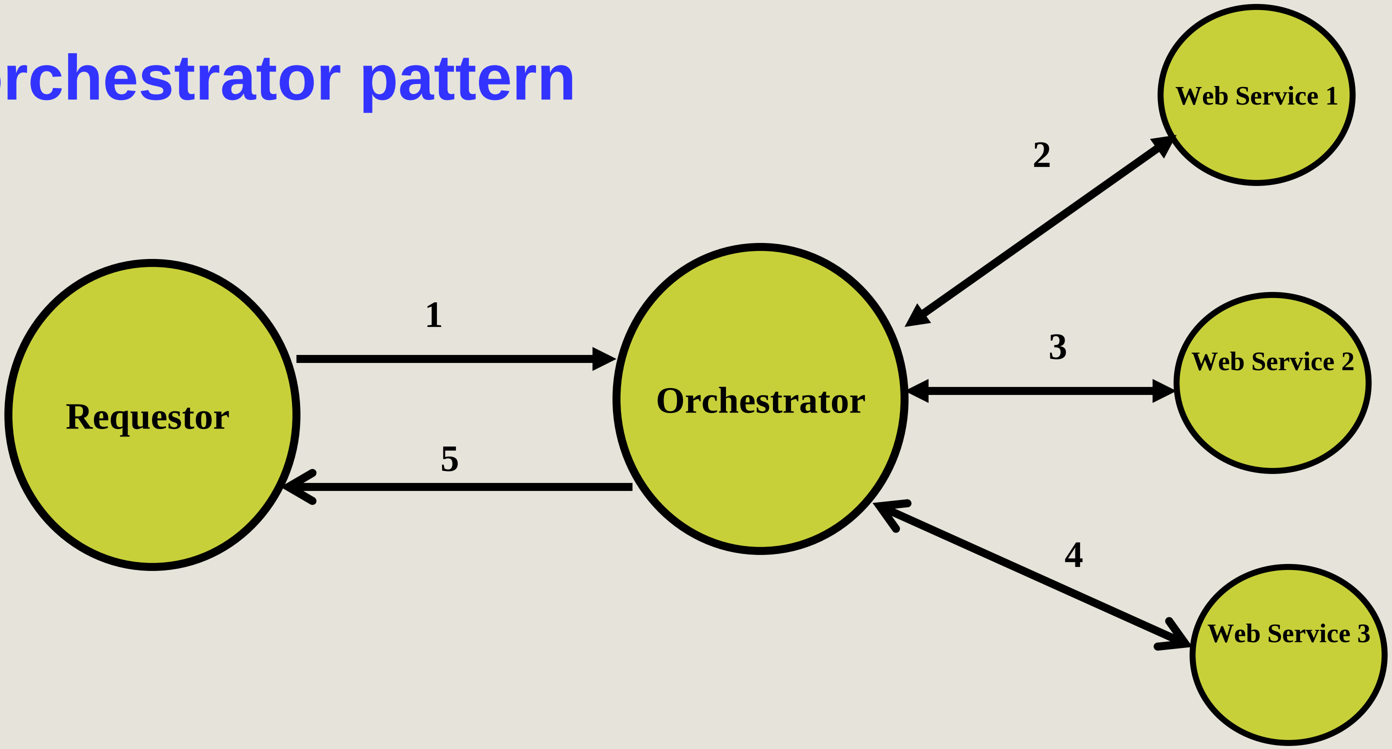
Examples of deployment patterns

The orchestrator pattern



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Common functions

Common functions are key to interoperability

Common supporting technologies

- XML 1.0
- SOAP 1.0
- WSDL 1.1
- HTTP 1.1
- UDDI 2.0X
- Use of WS-I profile

.

Common functions

Common functions are key to interoperability

Security (Identification of relevant standards and normative security technologies)

- Authentication
- Data integrity
- Confidentiality
- Key management
- Access control / authorization
- Non repudiation

.

Common functions

Common functions are key to interoperability

Service management (Identification of specific versions of UDDI)

- Registration
- Publication
- Discovery

.

A quick assessment

1. **Parlay-X Web services**
   - True Web services
     - Coarse grained approach (unlike WSDL version of Parlay specifications)
   - Work done “independently” of OMA
     - Situation is evolving

2. **OMA**
   - Tackle critical issues such as common functions
   - Integration of existing standards may take longer than planned

To probe further ...

On Web services

W3C specifications

Parlay-X specifications

OMA specifications

http://www.webservices.org/

- F. Curbera et al., Unraveling the Web services Web: An Introduction to SOAP, WSDL and UDDI, IEEE Internet Computing, Vol. 6, No2, March-April 2002, pp. 86-93

IEEE Internet Computing Special Issue, Middleware for Web services, January/February 2003

IEEE Computer Magazine, Special issue, Web services computing, October 2003

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Part III: Case Studies

- **Gaming with Parlay**
- **Web Service based application development**

A case study on PARLAY/OSA and SIP: Run For Your Life game

- 1 - Introduction
- 2 - Game
- 3 - Architecture
- 4 - Mapping

Introduction ...

Run-For-Your-Life

- Built from scratch in Ericsson Research lab in Montreal Canada
- Demonstrated at several trade shows (e.g. ICIN 2001, Parlay Munich meeting, Parlay Hong Kong meeting)
- Objectives assigned to the game design
  - Extensive usage of call control capabilities
  - Have fun …

.

Introduction ...

Objective of the case study …

Aim at helping in tackling two issues:

1. **PARLAY Call Control APIs that cannot be mapped onto SIP**
   - What are they?
   - What is the impact on service creation?

1. **SIP semantics that are not visible in PARLAY APIs as per today's specification**
   - What are they?
   - What is the impact on service creation?

The game ...

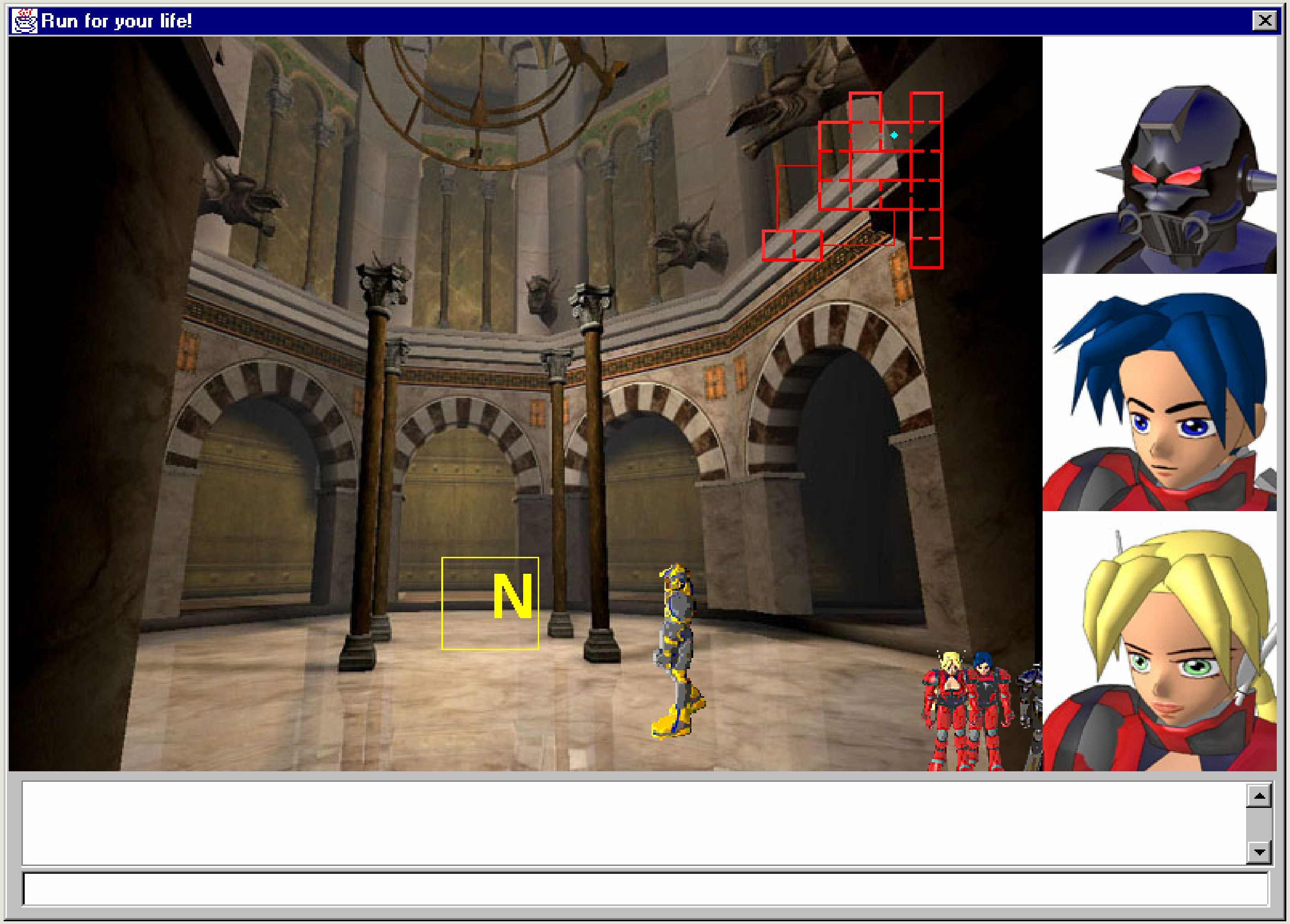
A multiparty cooperative game

- Group of people trapped in a house with several rooms set to burn/explode in a given time
- Can escape only if password is found
- Letters making the password scattered in selected rooms of the house
- People ending up in the same room can exchange hints about the password via audio and chat
- Game can be assimilated to a conference with as sub-conference people ending up in a same room

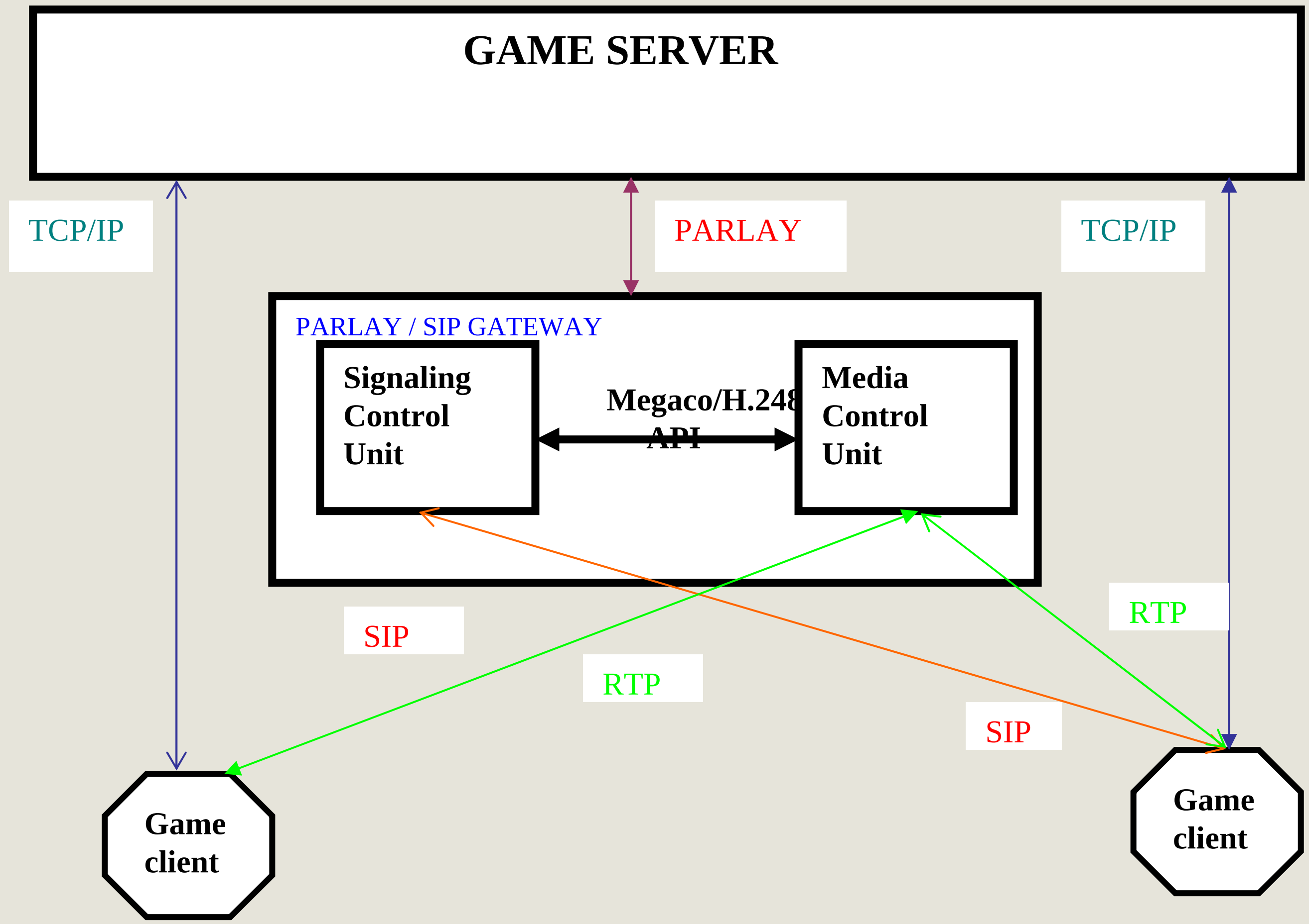
Requiring a set of well defined conferencing functionality

- Conferencing
- Sub-conferencing

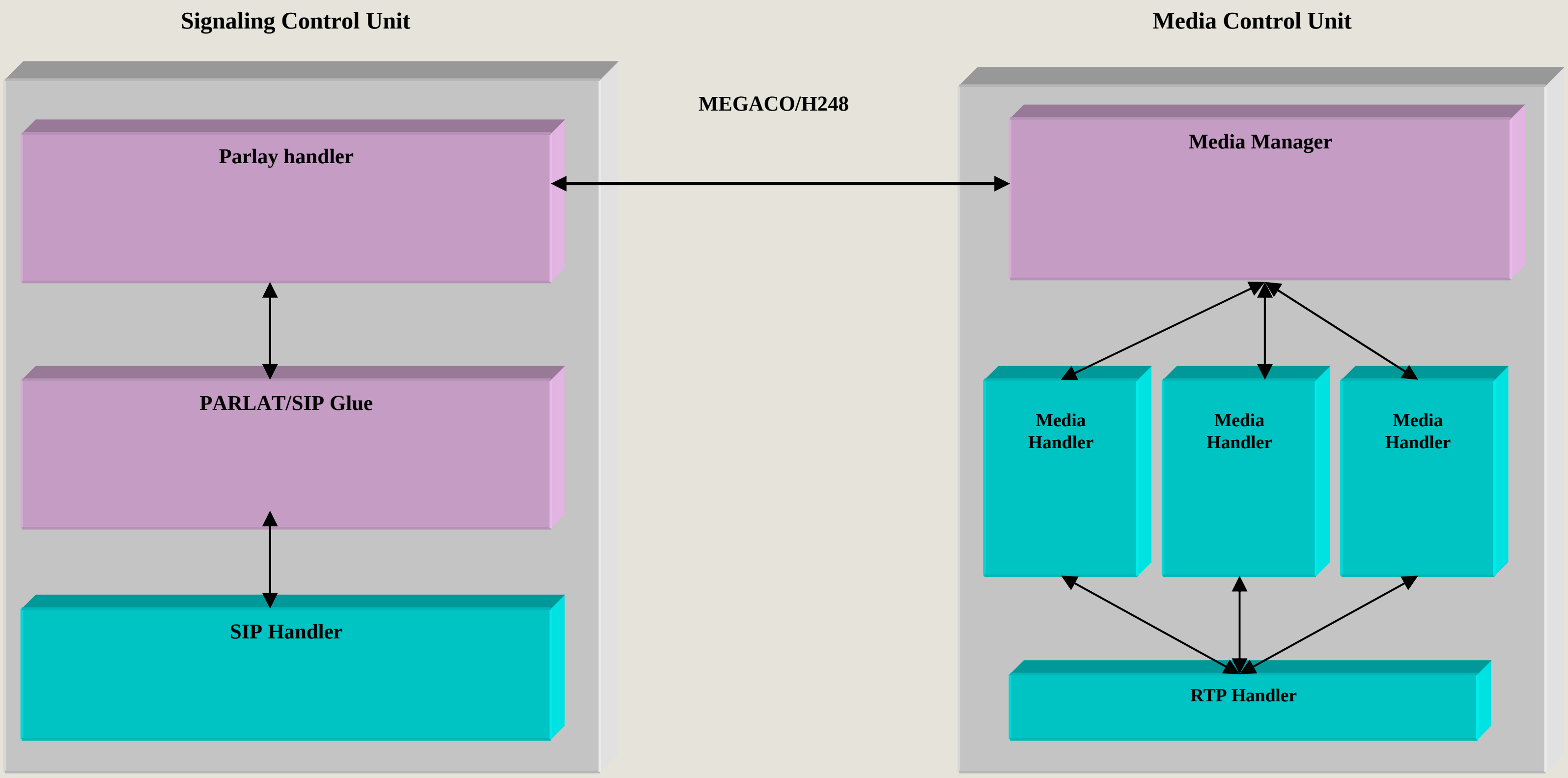
The game ...



Architecture ...



Architecture ...



Dial in ..

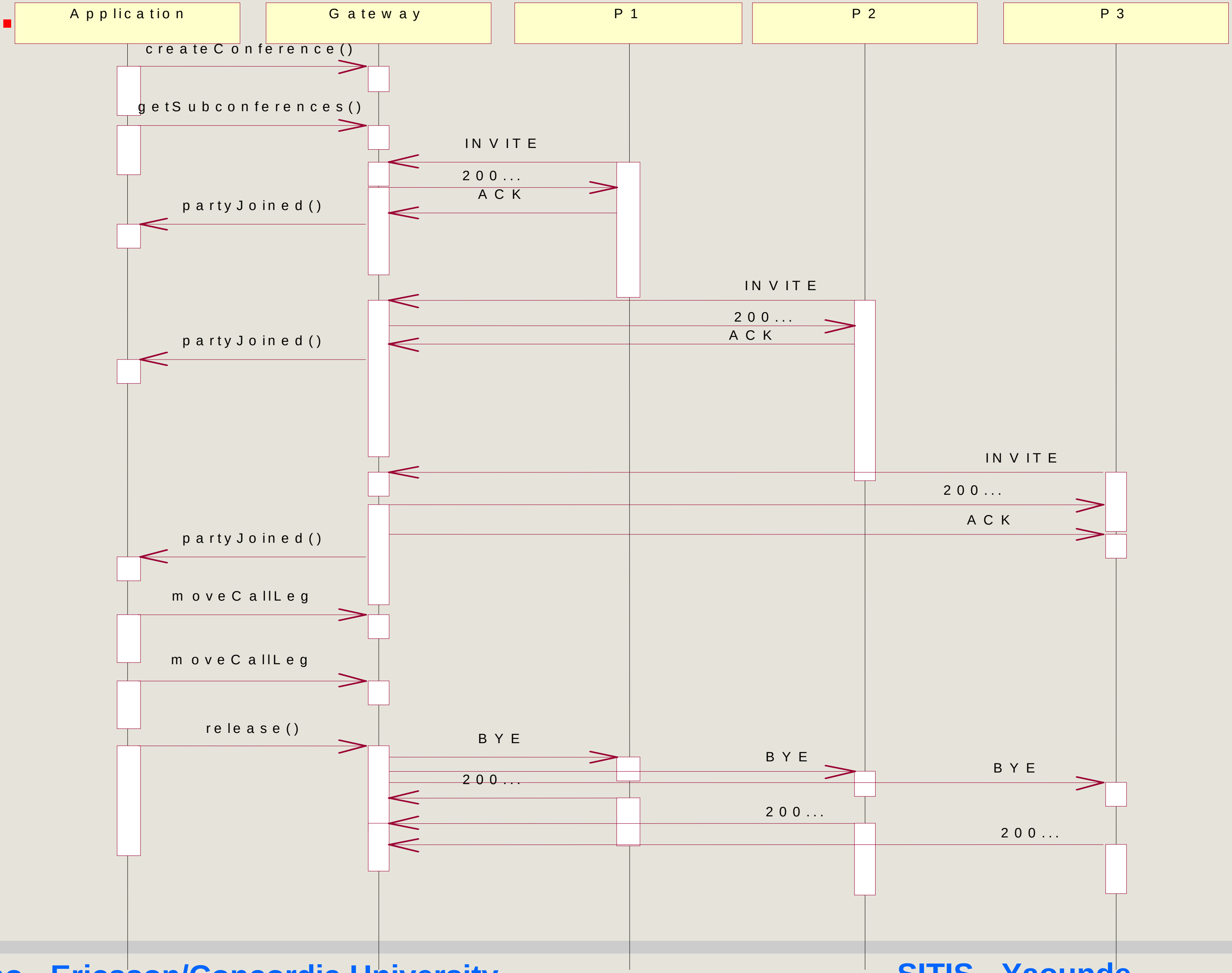
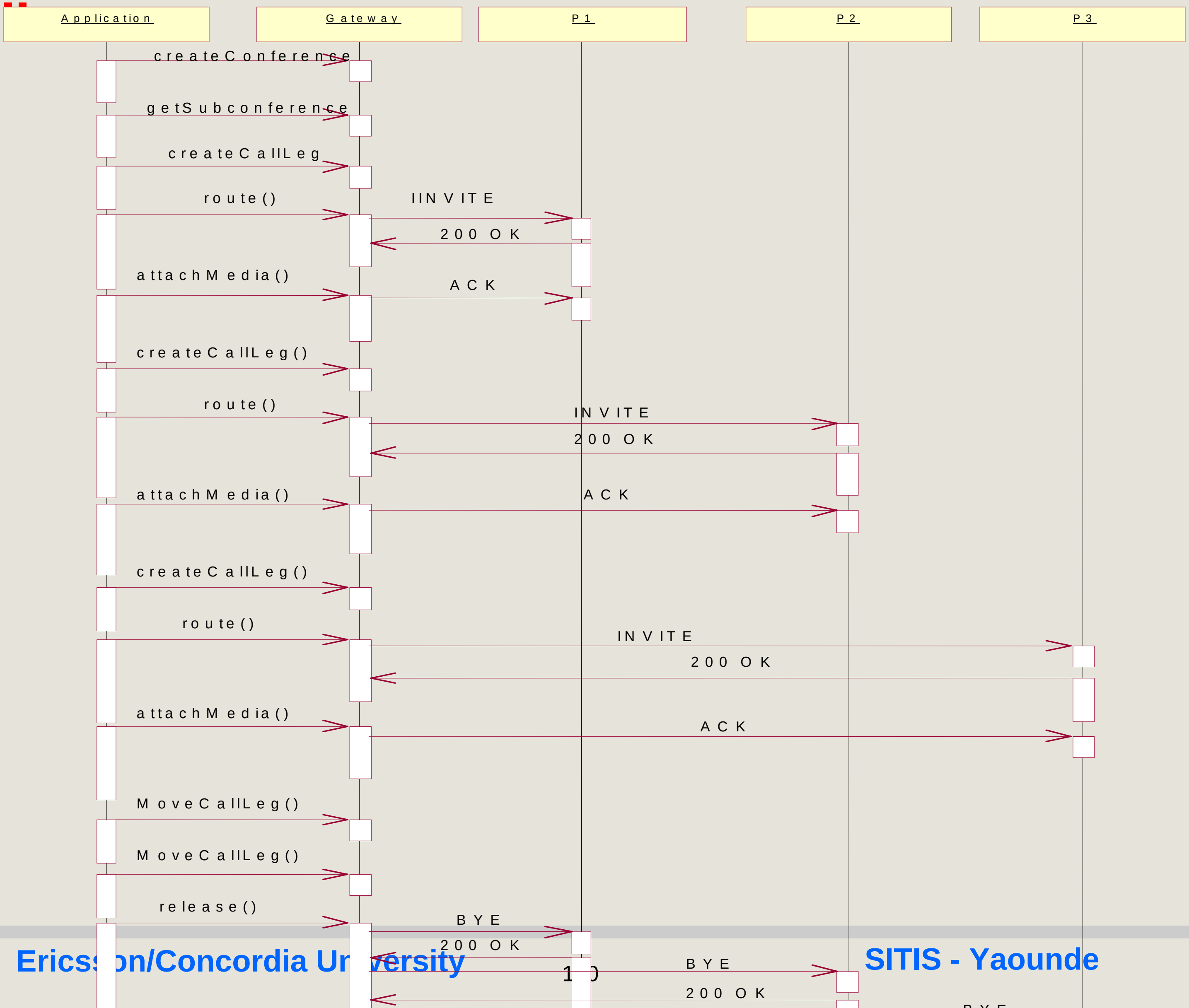


Figure 5 - Mapping for dial-in

Dial out ...


Application
Gateway
P1
P2
P3
createConference
getSubconference
createCallLeg
route()
IINVITE
200 OK
attachMedia()
ACK
createCallLeg()
route()
INVITE
200 OK
attachMedia()
ACK
createCallLeg()
route()
INVITE
200 OK
attachMedia()
ACK
MoveCallLeg()
MoveCallLeg()
release()
BYE
200 OK
BYE
200 OK

The mapping ...

PARLAY Call Control Services that cannot be mapped onto SIP

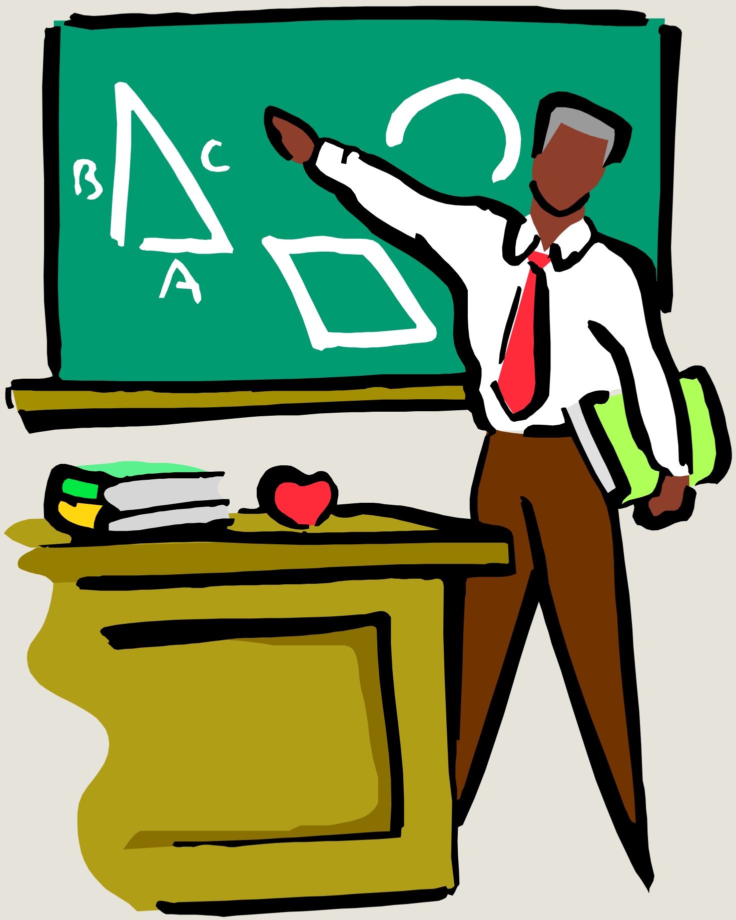
- There seems to be none
- However the mapping can be done in several ways in some cases

SIP semantics that are not visible in PARLAY APIs as per today's specification

- There exist a few (e.g. Possibility of a caller to state for instance that the call should not forwarded)
- PARLAY may be extended to cater to these features

.

Web service based application development …



1. Aims
2. Some of the applications
3. Some of the Web services
4. Implementation issues

Aims

1. Define telecommunication Web services that go beyond what is currently defined in the standards

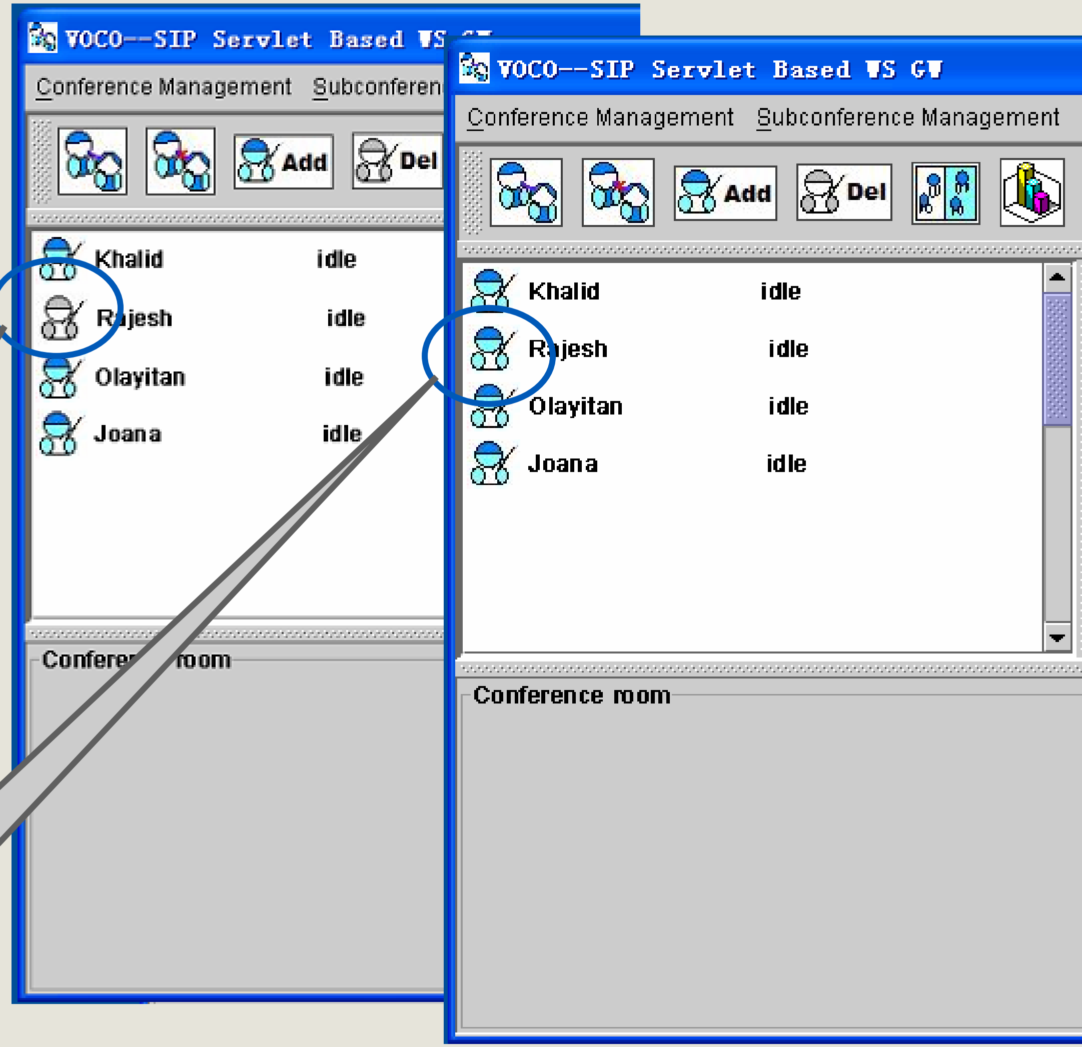
2. Study the implementation architectures and the related mapping issues (e.g. Parlay vs. SIP servlet)

3. Evaluate performance

4. Build demo applications

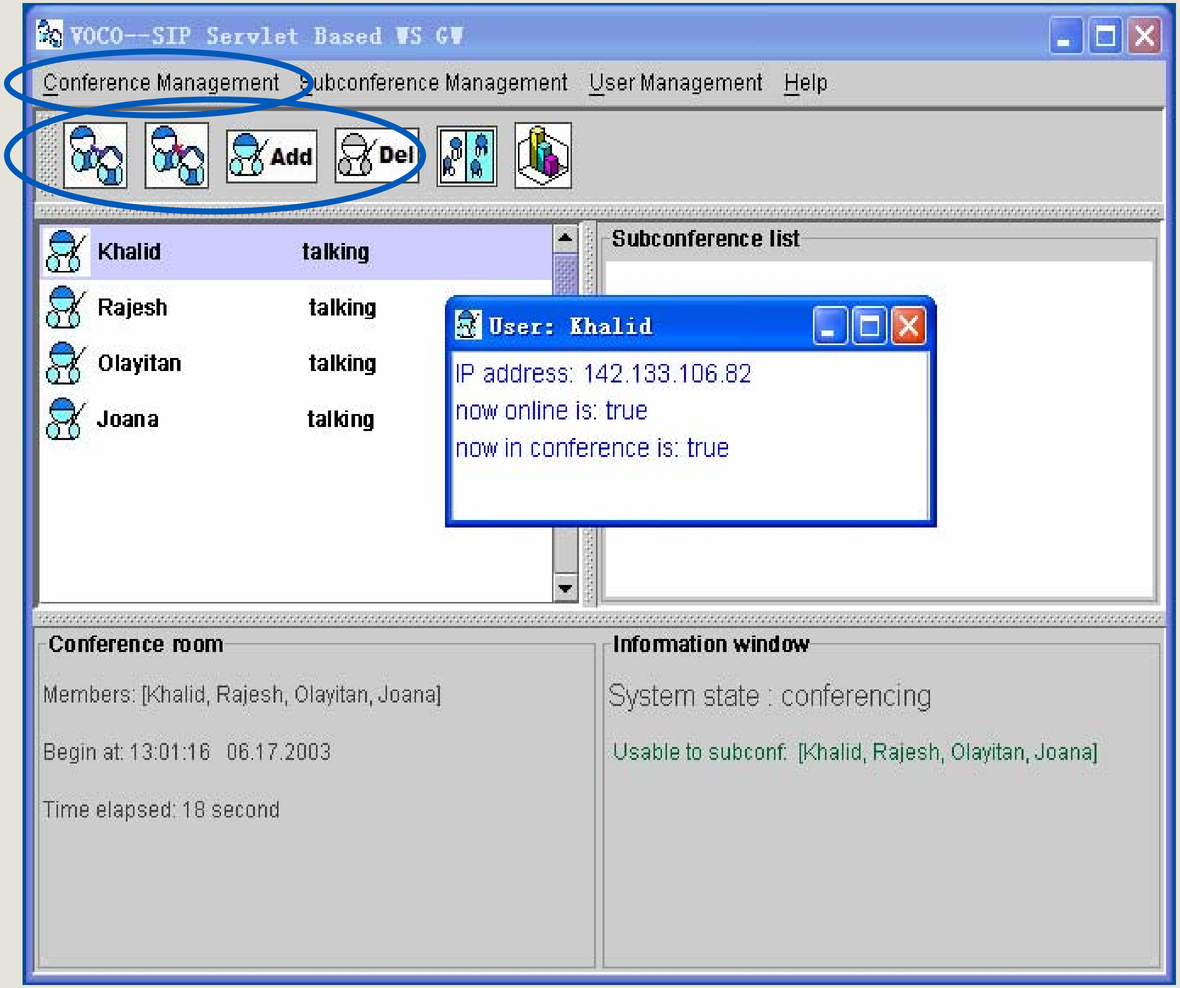
Some of the applications – Presence enabled conferencing

- User may be:
  - Online
  - Offline
- Update user’s presence in every ten seconds



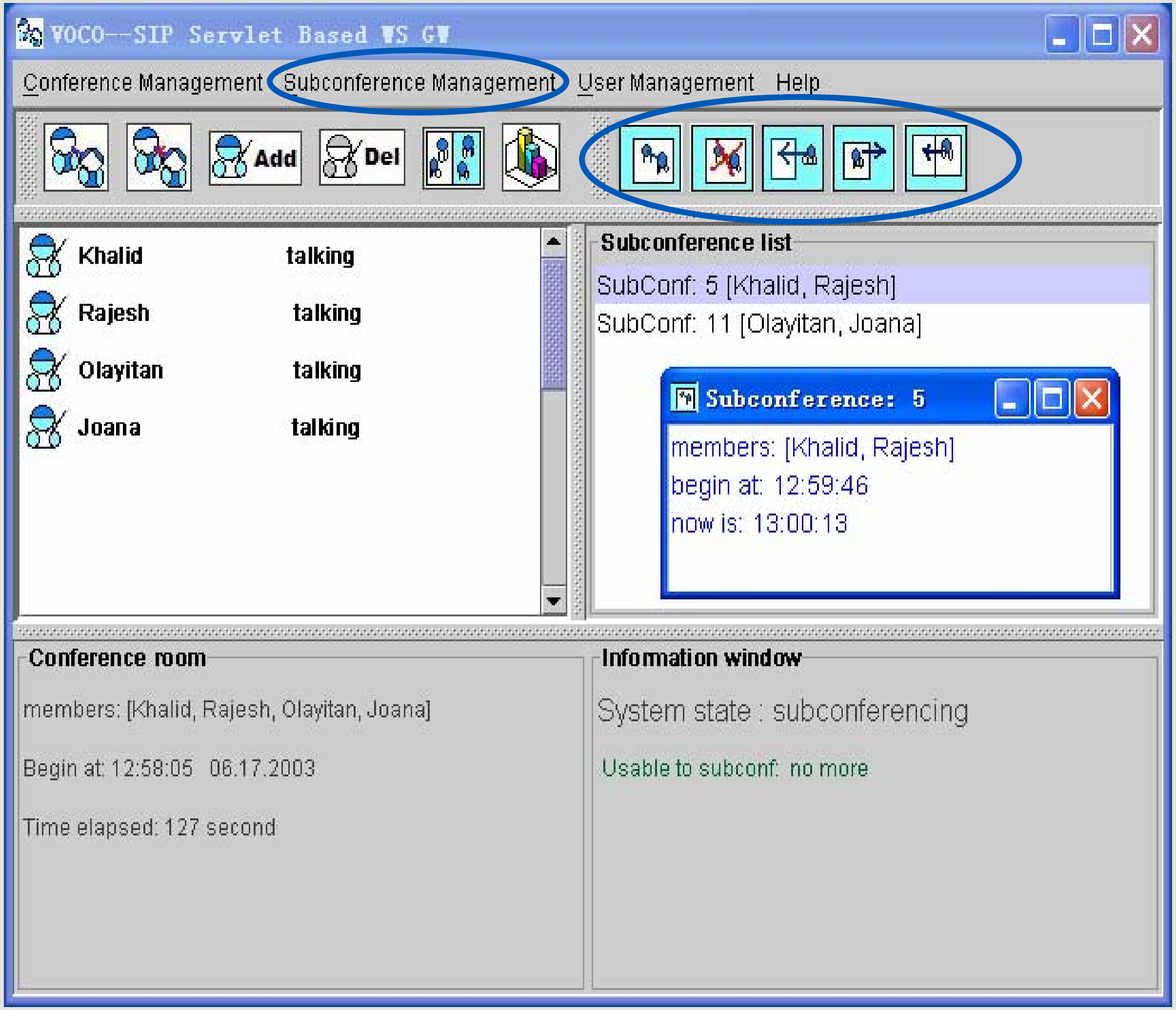
Some of the applications – Presence enabled conferencing

- Four operations provided:
  - Initiate a conference
  - Terminate a conference
  - Add a user to conference
  - Remove a user from conference

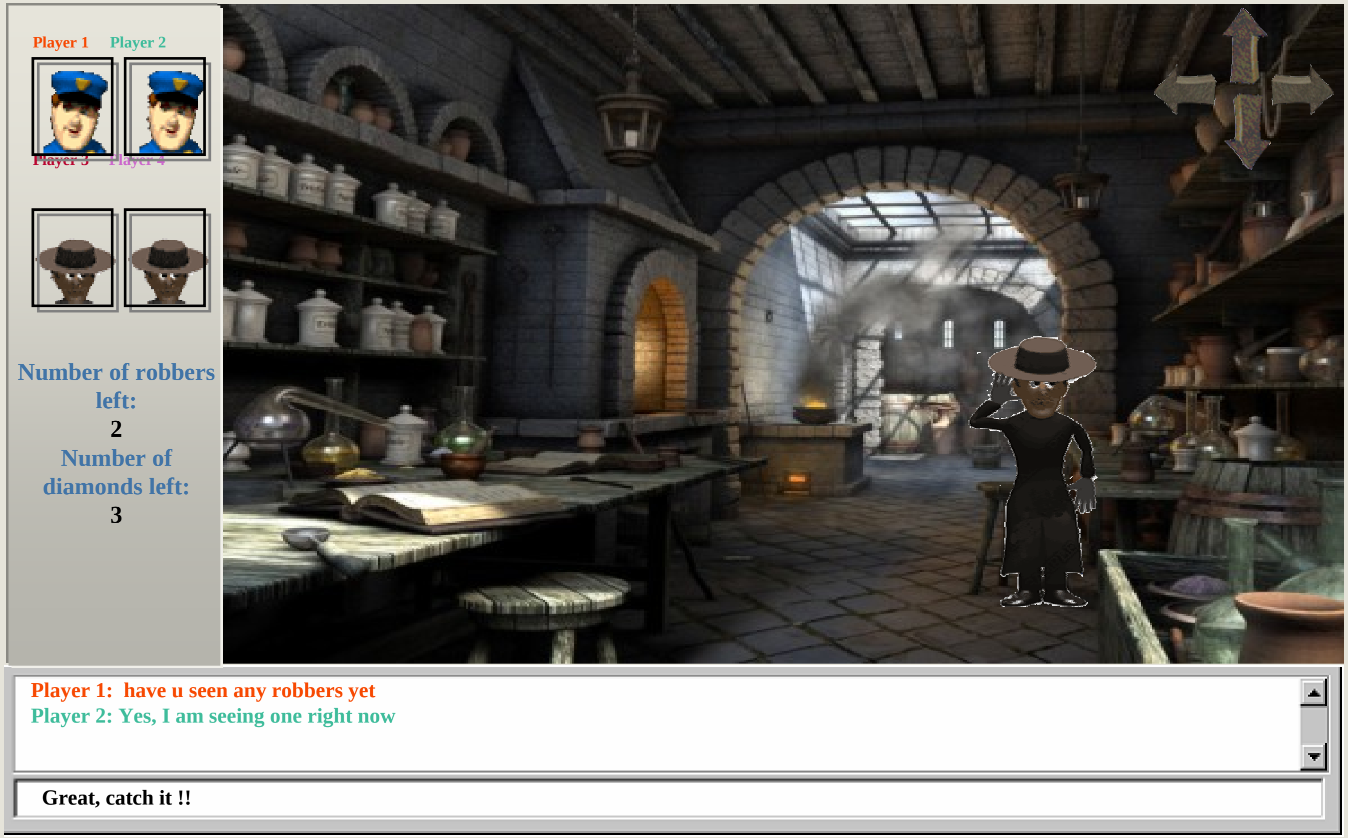


Some of the applications - Presence enabled conferencing

- Five operations provided:
  - Initiate a sub-conference
  - Terminate a sub-conference
  - Add a user from main conference to sub-conference
  - Remove a user from sub-conference to main conference
  - Move a user from one sub-conference to another



Some of the applications - Gaming

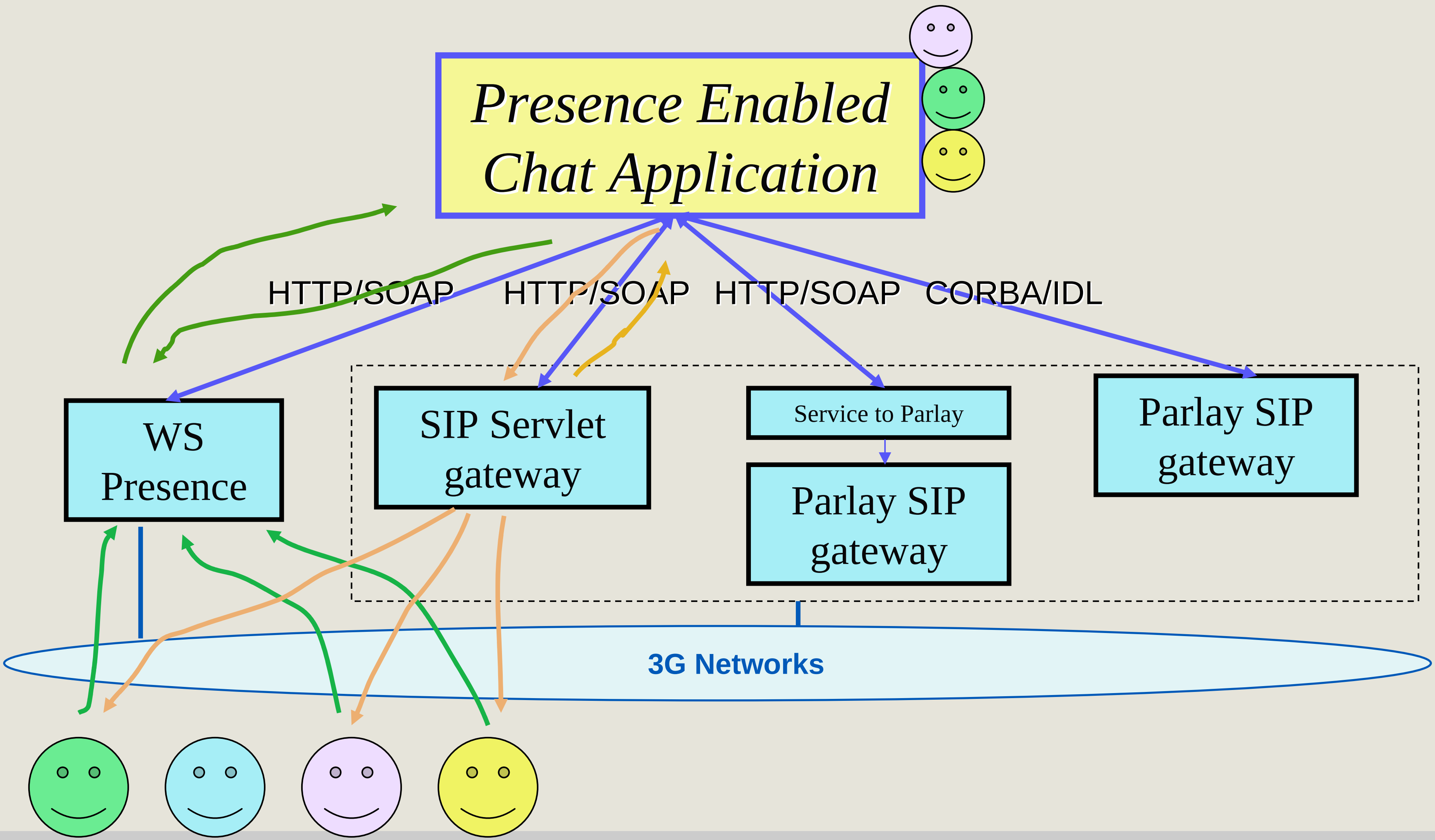


Some of the applications – Gaming

An example of Web service: Conferencing ...

```
boolean
initiateConf( String[] addresses,String mediaType,
String confType,   int duration,
int expectedUsers, String confID);
boolean
endConf( String confID);

boolean
addUser( String userAddress, String confID );

boolean
removeUser(String userAddress, String confID );

boolean
initiateSubConf( String[] users, String ConfID,
String subConfID )

boolean
endSubConf( String ConfID, String subConfID )

boolean
moveUser( String user, String ConfID,
String OriginSubConfID,
String DestinationSubConfID)
```

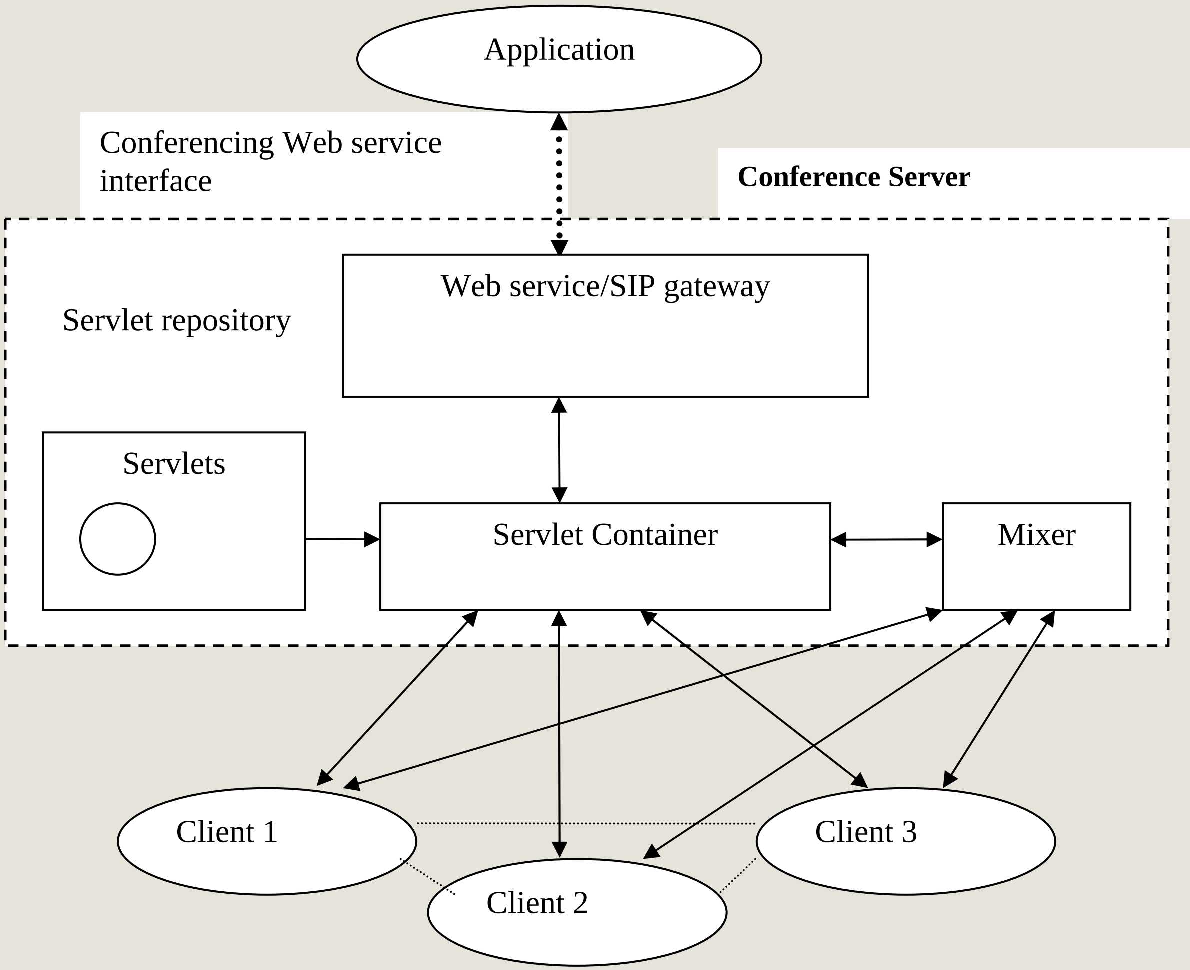
An example of implementation architecture



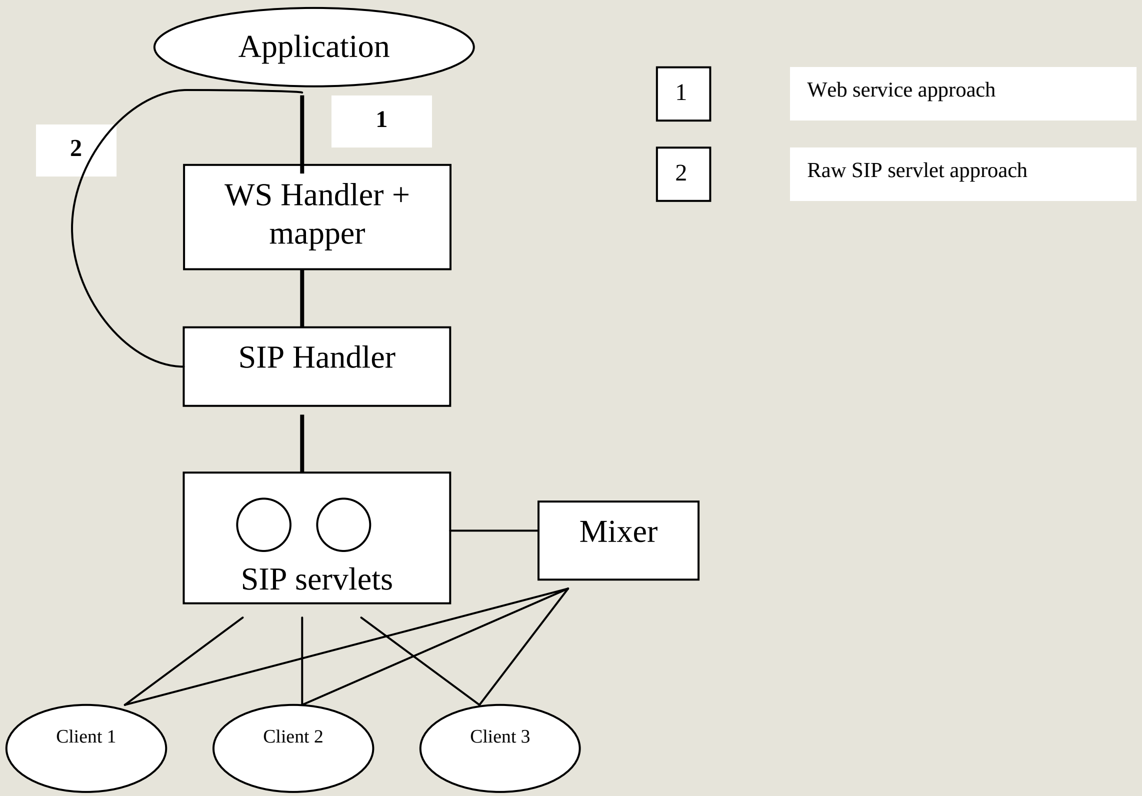
An example of mapping onto Parlay ...

Web Service	Parlay Gateway		
Function Call	Function Call	to/from GW	# times per WS function call
InitiateConf	createConf	to	1/ function call
	createCallLeg	to	1/ party in conference
	routeCallLeg	to	1/ party in conference
	attachMedia	to	1/ party in conference
endConf	release (sub-conference)	to	1/ sub-conference in conference
	release (conference)	to	2/ function call
	callLegEnded	from	1/ party in conference
addUser	createCallLeg	to	1/ function call
	routeCallLeg	to	1/ function call
	attachMedia	to	1/ function call
removeUser	release (callLeg)	to	1/ function call
	callLegEnded	from	1/ function call
initiateSubConf	createSubConference	to	1/ function call
	moveCallLeg	to	1/ party in subConference
EndSubConf	release (sub-conference)	to	1/ function call
	callLegEnded	from	1/ party in subConference
moveUser	moveCallLeg	to	1/ function call

Performance issues …The application



Performance issues …What was measured



Performance issues – The metrics

1. **Time delay**
   - Time Duration between a function call from the application and a feedback response from the container
     - First case: call goes to WS handler
     - Second case: call bypasses WS handler and goes directly to SIP handler

2. **Network load**

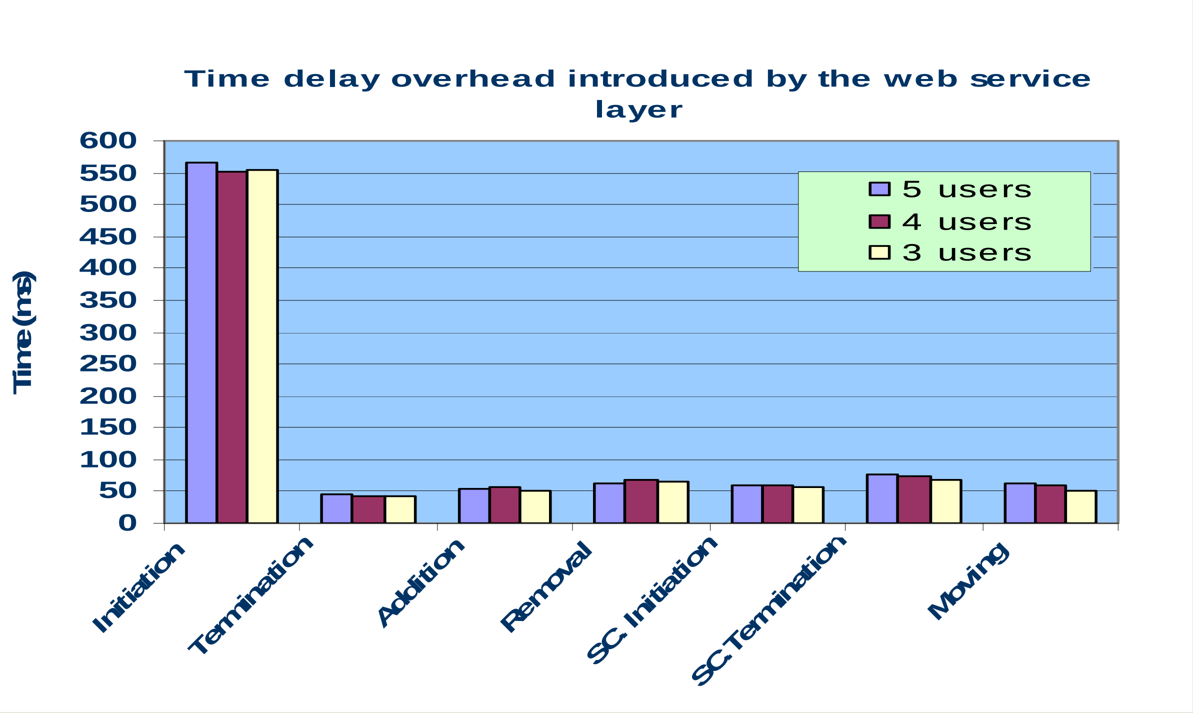
- Total size of all packets exchanged to perform a given function
- Captured using Etheral software

3. **Measurements are made in terms of overhead introduced by Web services**

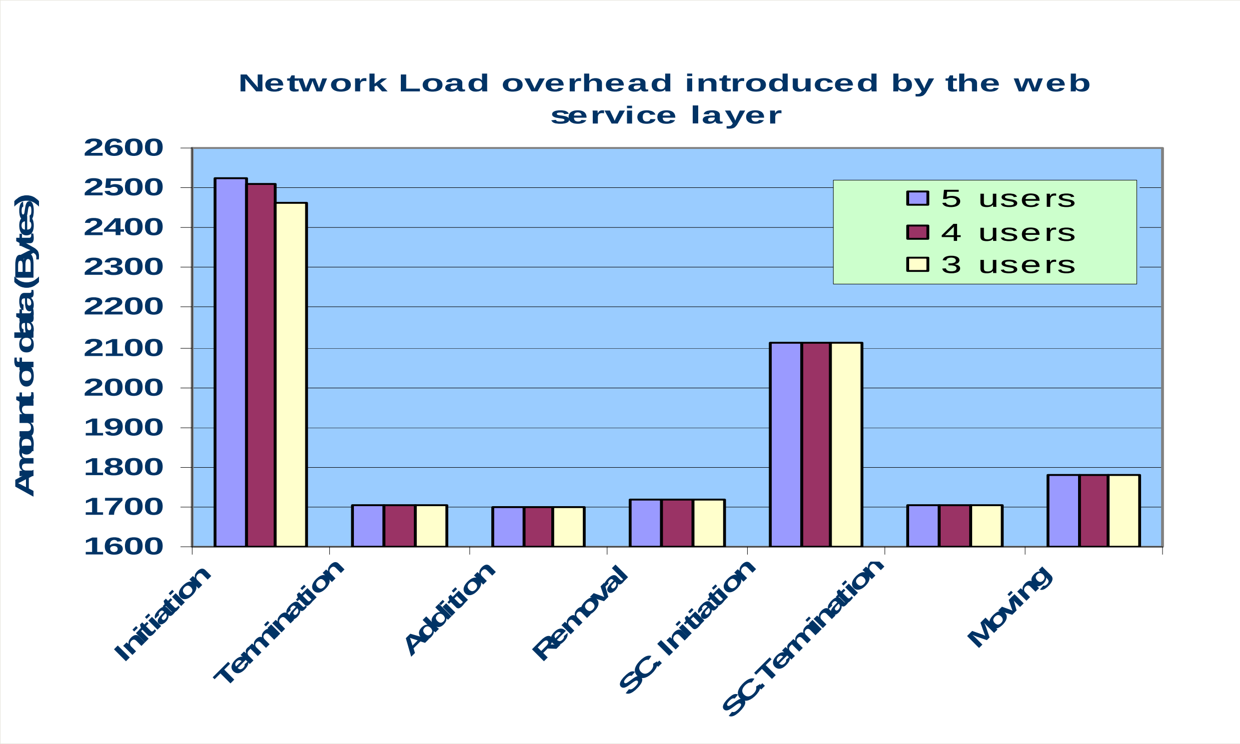
Time delay: Delay in first case – delay in second case

Network load: Load in first case – load in second case

Performance issues ...Some results



Performance issues …Some results



Lessons learned …

- Web service does make life easier to application developers …

A footprint measurement …

Conferencing part of application using Web Service ….

- About 350 lines of code

Same part using Parlay ..

- About 1300 lines of code

-Time delay may be acceptable but extra load is likely prohibitive in wireless environments

To probe further on the case studies ...

- R.H. Glitho and K. Sylla, Developing Applications for Internet Telephony: A Case Study on the Use of Parlay Call Control APIs in SIP Networks, *IEEE Network Magazine*, May/June 2004, Vol. 18, No. 3, pp. 48 - 55
- K. Hassan, J. Dasilva, R. Glitho and F. Khendek: Web services for call control in 3G – A Parlay based implementation, ICIN 2004
- M. El Barachi, R. Glitho and R. Dssouli, Developing Applications in Internet Telephony: A case study on the use of Web services for conferencing in SIP networks, submitted, ACM TOIT
- M. El Barachi, Using Web Services for Application Development in Internet Telephony: A case Study on Conferencing in SIP Networks, MSc thesis, March 2004

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