The IETF Approach to Internet Telephony: An Overview

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Internet Telephony …

Definition:

Telephony over the Internet (or more generally over packet switched networks)

Telephony

- Conversational voice (or more generally multimedia) exchange over a network
  - Signalling component: Logical links between the parties
  - Media component: actual media exchange

Key differentiating factor of Internet Telephony

Use of packet switched networks instead of circuit switched networks

"Synonyms" used in the industry: IP Telephony, Voice over IP, Third generation telecommunication networks, packet telephony …

Internet Telephony …

Milestones

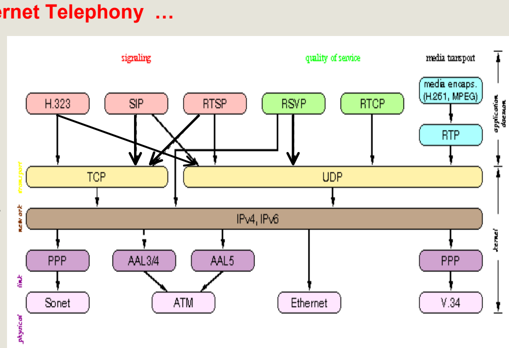
- Late 70s:
  - First two party voice calls over Internet (Network Voice Protocol (NVP - RFC 741 - November 1977)
- 80s:
  - Emergence of proprietary systems for Internet Telephony
- 90s:
  - Emergence of standards (e.g. SIP, H.323)
- 00s:
  - Backing by telcos (e.g. 3GPP specifications)
  - Backing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

Internet Telephony …

Current Approaches

- ITU-T: H.323 standards
- IETF: SIP standards
- Focus of this tutorial
  - IETF approach

Internet Telephony …



Objectives assigned to this tutorial …

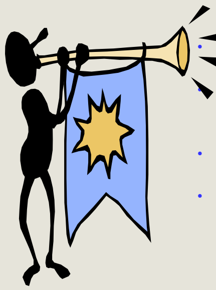
Discuss the basics of telephony and introduce briefly H.323

Discuss in detail the IETF specifications for Internet Telephony

- **Session Initiation Protocol (SIP) and accompanying specifications**
  - **Media transportation**
  - **QoS**
- **Megaco / Soft-switches**

Discuss the IETF and related specifications for value added services in SIP environment

Outline

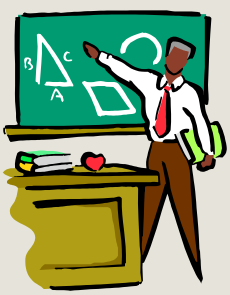


- Part I - Circuit switched telephony and H.323
- Part II - SIP
- Part III – Megaco / H.248
- Part IV – Value added services for SIP environment

Part I: Circuit switched telephony and H.323

- Circuit Switched Telephony
- H.323

Essentials of circuit switched telephony

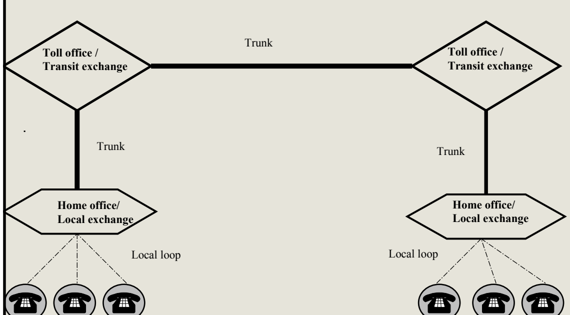


- Circuit switching vs. packet switching
- Local loops, telephone exchanges and trunks
- Signaling
- Beyond fixed telephony

Circuit switching vs. packet switching

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

A simplified telephony network ...

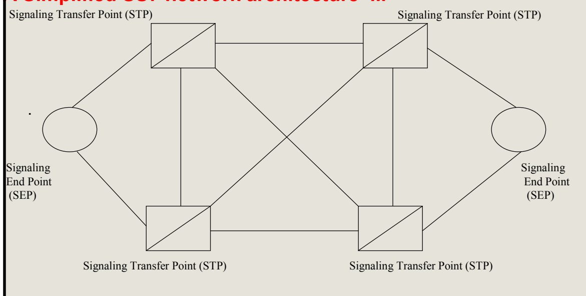


Signaling ...

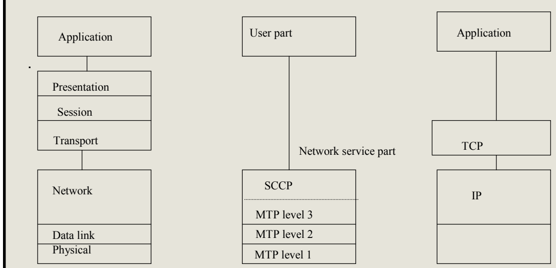
Establishment, modification and tear down of calls:

- User Network Signalling
  - Between user and home office
  - On/off hook, dial tone …
  - Carried over local loops
- Network – Network signalling
  - Between telephone exchanges
  - Initially in-band (Same trunks as voice)
  - Out-band in modern circuit switched telephony
    - Signalling data carried over a separate and overlay packet switched network (Signalling System no7 – SS7)

A Simplified SS7 network architecture ...



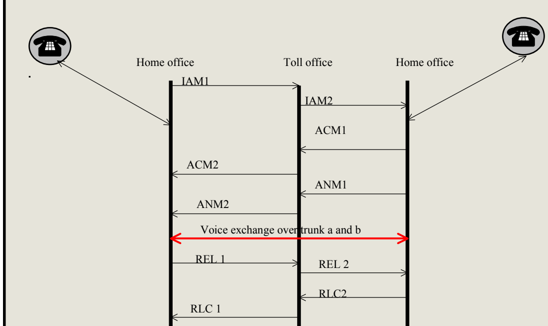
SS7 Protocol stack ...



SS7 Network Service Part ...

- **Message Transfer Part (MTP)**
  - Level 1
    - Physical layer (signalling data layer functions)
    - Bit rates: 56 kbps / 64 kbps
  - Level 2
    - Similar to data network bit oriented protocols (e.g. HDLC)
    - Adaptation to stringent performance requirements (e.g Fill in signalling units when there is no traffic)
    - Error correction, monitoring
    - Flow control
  - Level 3
    - Message handling (e.g. routing, distribution)
    - Signalling network management (e.g. diversion from an unavailable route with loss or duplication)
- Signalling Connection Control Part (SCCP): Add to MTP the possibility of having connection oriented communication

Integrated Service Digital Network (ISDN) - User Part



Beyond fixed telephony ...

Cellular telephony

- Mobile Switching Centre
  - Switches used in cellular telephony – Additional features for mobility management
- Home location register (HLR) /Visitor location register (VLR)
  - Keep information on user location
- Base stations
  - Access point to cellular networks
  - Communicate with end user terminals
  - Control cells
- Signalling in cellular networks
  - SS7 based

Beyond fixed 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

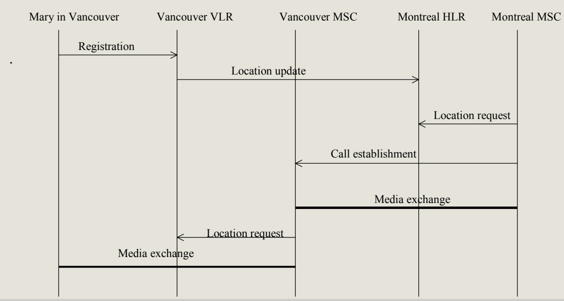
Second Generation (90s – early 00s)

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

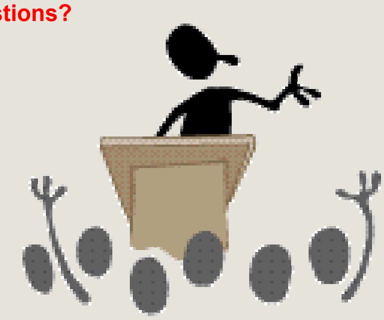
Third Generation (90s – early 00s)

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

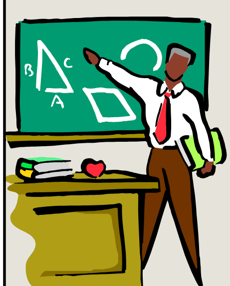
Mary a Montreal subscriber receives a call while in Vancouver



Questions?



H.323



1. Introduction
2. Functional entities
3. Signaling protocols
4. H.323 vs. SIP

H.323: Introduction

First standard to emerge for Internet Telephony

1996: First release by ITU-T
- Target initially conferencing in LAN/enterprise environment, but subsequently enhanced for WAN

2000: Lost the battle to SIP for signaling in third generation cellular networks
- Lost some momentum as a consequence

Current Status:
- Remains widely deployed, especially in enterprise environment
- Wide range of commercial products
- No freeware (as far as I know)

http://www.h323forum.org/

H.323: Introduction

First standard to emerge for Internet Telephony

- Signalling standards reminiscent of circuit switched telephony:
  - H.225.0
  - Q.931
  - H.245
- Re-use IETF specifications
  - Media (RTP, RTCP)
  - QoS (e.g. RSVP, DiffServ)
- Many other specifications (e.g. H.324 Terminal for low bit rate multimedia communications)

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

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

Key features

- ASN.1 based messages
- Request / reply protocol
- Signaling between end-points
  - Terminal or gateway

  and
  - Gatekeeper
- Use unreliable channels
  - Retries
  - Timeouts

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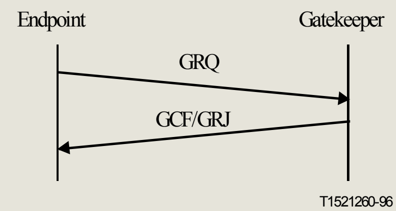
RAS: Gatekeeper discovery …



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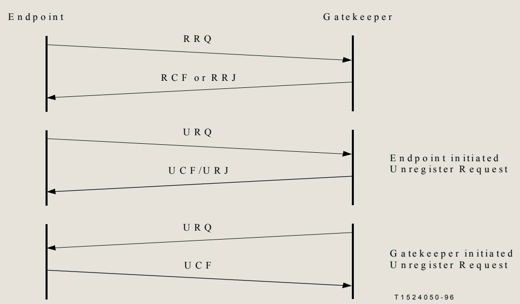
RAS: Admission request …



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

Key features

- ISUP signaling (Q.931) based
- ASN.1 based messages
- Transaction oriented protocol
- Signaling between end-points
  - Terminal or gateway

  and
  - Gatekeeper
- Use reliable channels

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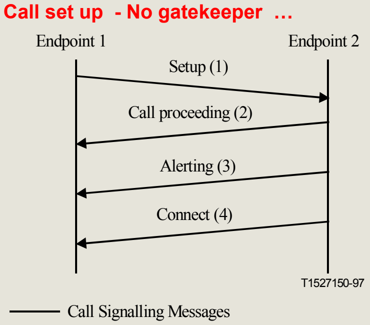
RAS: Call set up - No gatekeeper …



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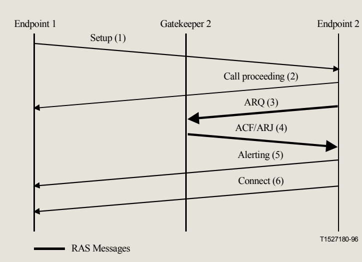
RAS: Call set up - 1 gatekeeper …



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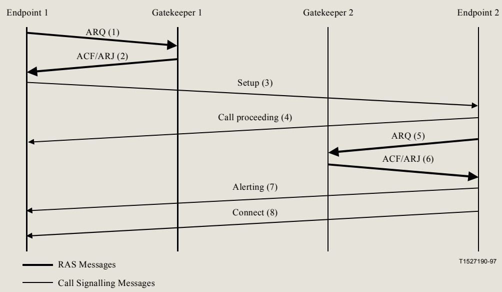
RAS: Call set up - Two gatekeepers …



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

Key features
- ASN.1 based messages for
  - Master/slave determination
  - Capabilities negotiation
  - Logical channel signaling
- Several modes
  - Request/reply
  - Commands
  - Indications
- Signaling between end-points
  - Terminal or gateway
  and
  - Gatekeeper
- Use reliable channels

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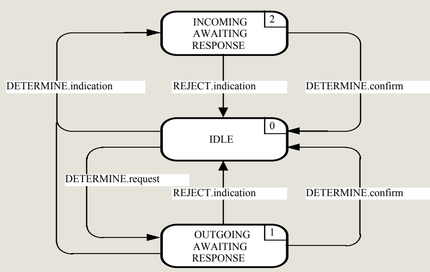
H.323 signaling: Master / slave determination



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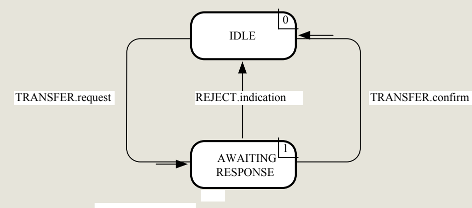
H.323 signaling: Capabilities exchange



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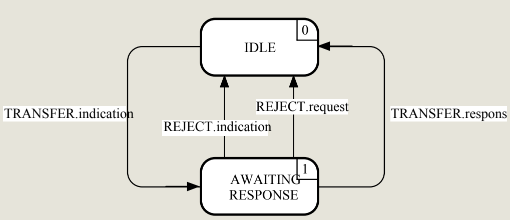
H.323 signaling: Capabilities exchange



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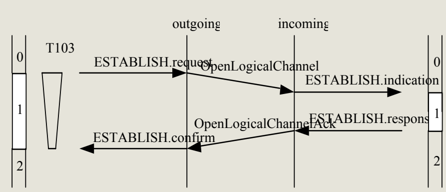
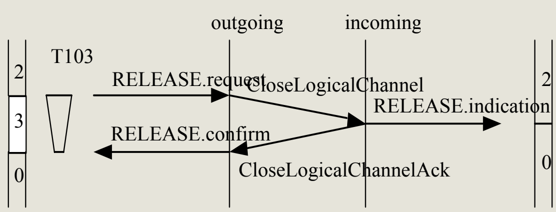
H.323 signaling: Logical channels



H.323 signaling: Logical channels



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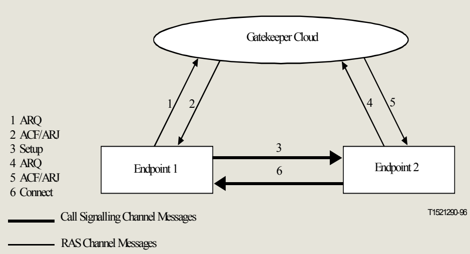
H.323 signaling: An important feature - Fast connect

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 … alternative 1



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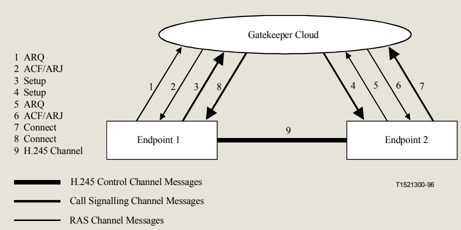
H.323 signaling : Putting it together …alternative 2



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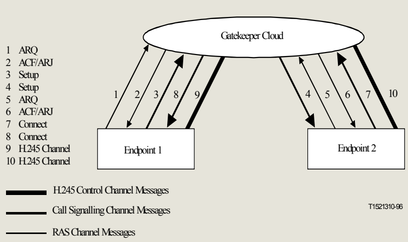
H.323 signaling: Putting it together - alternative 3



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References …

- A. R. Moderassi and R. Skoog, Signaling System No7: A Tutorial, IEEE Communications Magazine, July 1990, available at:http://www.comsoc.org/livepubs/surveys/public/4q98issue/reprint4q.html
- A. Tanembaum, Computer Networks, 4th edition, Prentice Hall 2003 (Chapter 2.5 – The public switched telephone system network)
- H. Liu et al., Voice over IP Signaling: H.323 and Beyond, IEEE Communications Magazine, October 2000
- H.323 ITU-T specifications

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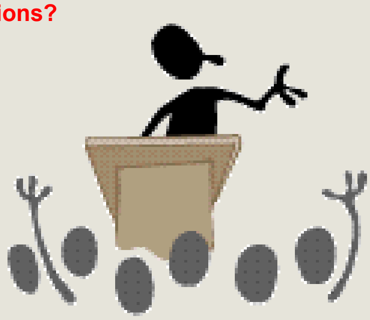
Questions?



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Part II: SIP and accompanying specifications

- SIP core
- SIP extensions
- Accompanying specifications
  - Session description
  - Media transportation
  - QoS

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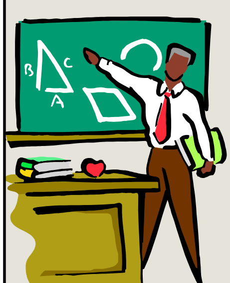
Session Initiation Protocol (SIP) - Core



1. Introduction
2. Functional entities
3. Messages
4. A digression on SDP
5. Examples

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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)
- 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)
- Current status
  - Gaining momentum due to adoption for third generation mobile networks
  - Replacing some of the H.323 installed basis
  - Availability of both commercial products and freeware
  - http://www.cs.columbia.edu/sip/

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SIP: Introduction

SIP core Signaling

- A signalling protocol for the establishment, modification and tear down of multimedia sessions
- Based on HTTP

A few key features

- Text based protocol
- Client/server protocol (request/response protocol)

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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

Generic structure

- Start-line
- Header field(s)
- Optional message body

Request message

- Request line as start line
  - . Method name
  - . Request URI
  - . Protocol version

Response message

- Status line as start line
  - . Protocol version
  - . Status code
  - . Reason phrase (Textual description of the code)

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)

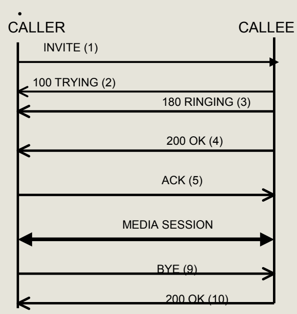
SIP: The messages

Response message

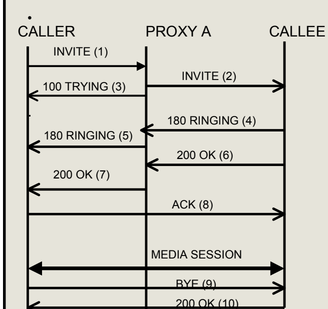
- Provisional
- Final

Examples of status code
1xx: Provisional
2xx: Success
6xx: Global failure

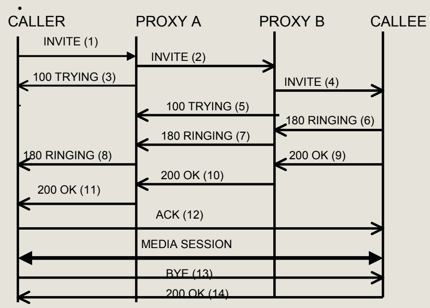
SIP: Examples of call case



SIP: Examples of call case



SIP: Examples of call case



SIP: Examples of messages from the RFC

An example of an INVITE

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

SIP: Examples of messages from the RFC

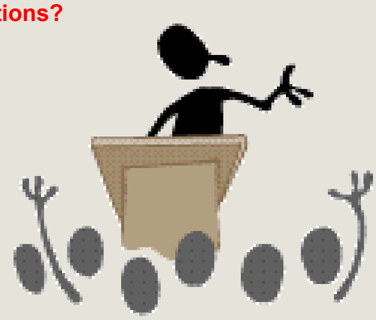
An example of an OPTIONS message

OPTIONS sip:carol@chicago.com SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKhjhs8ass877
Max-Forwards: 70
To: <sip:carol@chicago.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 63104 OPTIONS
Contact: <sip:alice@pc33.atlanta.com>
Accept: application/sdp
Content-Length: 0

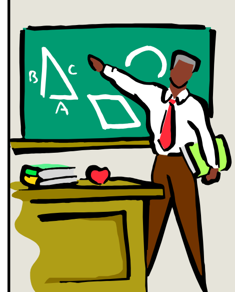
SIP: Examples of messages from the RFC

An example of RESPONSE to the OPTIONS request
SIP/2.0 200 OK
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKhjhs8ass877
;received=192.0.2.4
To: <sip:carol@chicago.com>;tag=93810874
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 63104 OPTIONS
Contact: <sip:carol@chicago.com>
Contact: <mailto:carol@chicago.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
Accept: application/sdp
Accept-Encoding: gzip
Accept-Language: en
Supported: foo
Content-Type: application/sdp

Questions?



SIP – Selected Extensions



1. Event framework
2. Others

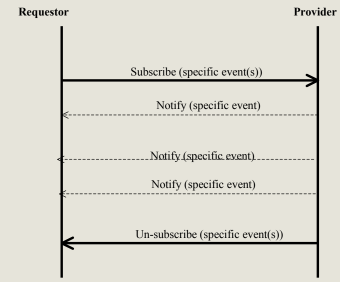
Event Notification

Motivation

- Necessity for a node to be asynchronously notified of happening (s) in other nodes
  - Busy / not busy (SIP phones)
    - A client A can call again a client B when notified that B is now not busy
  - On-line / Off-line
    - Buddy list

Event Notification

Conceptual framework



Event Notification

The SIP Event Notification Framework

- Terminology
  - Event package:
    - Events a node can report
    - Not part of the framework – Part of other RFCs
  - Subscriber
  - Notifier
- New Messages
  - Subscribe
    - Need to be refreshed
    - Used as well for un-subscribing (expiry value put to zero)
  - Notify

Event Notification

The SIP Event Notification Framework

- More on the methods
  - New headers
    - Event
    - Allow-Events
    - Subscription state

Event Notification

An example of use: REFER Method

- Recipient should contact a third party using the URI provided in the CONTACT field
  - Call transfer
  - Third party call control
- Handled as Subscribe / notify
  - REFER request is considered an implicit subscription to REFER event
    - Refer-TO: URI to be contacted
    - Expiry determined by recipient and communicated to sender in the first NOTIFY
    - Recipient needs to inform sender of the success / failure in contacting the third party

Event Notification

Another example of use: Presence

- Dissemination/consumption of presence information (e.g. on/off, willingness to communicate, device capabilities, preferences)
  - Numerous applications
    - Multiparty sessions initiated when a quorum is on-line
    - News adapted to device capabilities
- Several standards including SIMPLE (SIP based)
  - Handled as Subscribe / notify in SIMPLE
    - Watchers / presentities
      - Explicit subscriptions
      - Explicit notifications

INFO Method

Allow the exchange of non signalling related information during a SIP dialog

- Semantic defined at application level
- Mid-call signalling information
  - DTMF digits with SIP phones
- Info carried as
  - Headers and/or
  - Message body

References

Core SIP

- **SIP core signalling:**
- H. Schulzrinne, an J. Rosenberg, SIP: Internet Centric Signaling, IEEE Communications Magazine, October 2000
- RFC 3261, June 2002 (Obsoletes RFC 2543)
- RFC 2327 (SDP)

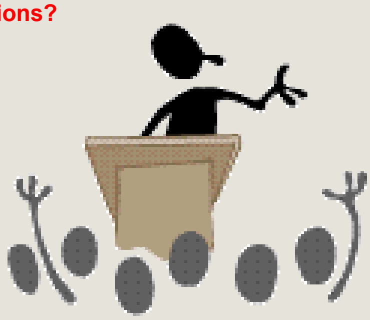
- **SIP extensions**

No overview paper

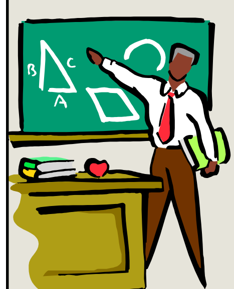
- **RFC 3265, 3515 (Event framework)**
- **RFC 2976 (INFO Method)**

- **3GPP**
- **No overview paper**
- **3GPP TS 23.228**
- **3GPP TS 2302**

Questions?



Accompanying standards: SDP



1. **Introduction**
2. **Message structure**
3. **Examples of messages**

SDP: Introduction …

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)

SDP: Introduction …

Session Description Protocol

Used with SIP

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

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SDP: Message structure …

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)

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SDP: Examples of messages from the RFC …

Session Description Protocol

An example from the RFC …

v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
s=SDP Seminar
i=A Seminar on the session description protocol
u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
e=mjh@isi.edu (Mark Handley)
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
a=recvonly
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
m=application 32416 udp wb
a=orient:portrait

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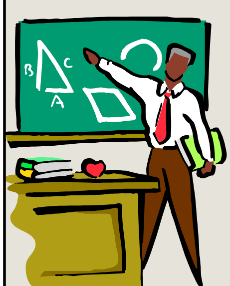
Accompanying standards: Media transportation



1. Introduction
2. Use cases
3. RTP
4. RTCP

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Introduction …

Two complementary protocols

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

Provide guarantee for neither out-of-order delivery nor QoS

- Rely on other IETF protocols (e.g. IntServ, DiffServ,)
- Independent of underlying transport protocol
  - Usually run on top of UDP

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Use cases …

Multicast audio conference on Internet

- Information distributed to the participants
  - Multicast address
  - Ports
    - 1 RTP port
    - 1 RTCP port

Multicast audio / video conference on Internet

- Information distributed to the participants
  - Multicast address
  - Ports
    - 1 RTP port for audio
    - 1 RTCP port for audio
    - 1 RTP port for video
    - 1 RTCP port for video

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Some RTP concepts …

End system

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

Synchronization source

- Grouping of data sources for playing back purpose (e.g. voice vs. video)
- Same timing and sequencing space Multicast address
- An end system can act as several synchronization sources (e.g. IP phone with video capabilities)

Mixers / translators

- Group RTP streams
- Example: conference bridges

RTP packets: Examples of fields …

- Protocol version
- CSRC count:
  - Part of packets sent by translators / mixers
  - Number of contributing sources
- Payload type
- Sequence number
- Time stamp
- Synchronization source
  - Generated randomly
- List of contributing sources

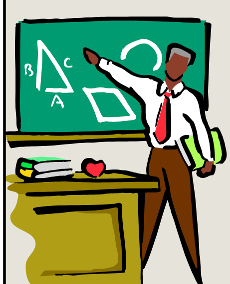
Note: The actual data is contained in the message body

RTCP …

Purpose: Feedback on the quality of RTP message reception
- Adaptive encoding
- Distribution fault detection
- Third party monitoring (e.g. network operators)

Mechanism: Distribution of periodical feedback packets
- Sender report
- Receiver report
- End of participation

Accompanying standards: QoS



1. Introduction
2. Integrated services
3. Differentiated services

Introduction …

The three models

1. Fat dumb pipe
- Classical model used in circuit switched telephony
- Over-provisioning
- Very inefficient from resource usage point of view

2. Integrated services
- Mid 1990s
- Provide QoS guarantee in absolute terms

2. Differentiated services
- Aim at addressing IntServ shortcomings
- Provide QoS guarantee in relative terms

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

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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

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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

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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

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Differentiated services - DiffServ …

Fundamental principle: A code point – Differentiated service code point (DSC) 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

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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

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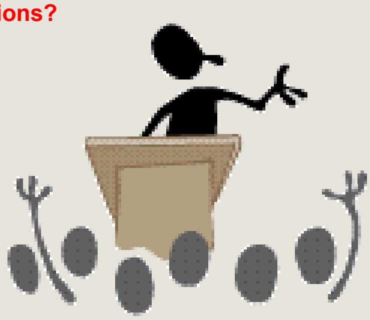
References …

1. IETF RFCs
   RFC 2327 (SDP)
   RFC 1889 (RTP/RTCP)
   RFC 1633 (IntServ)
   RFC 2205 (RSVP)
   RFCs 2430, 2474 … (DiffServ)

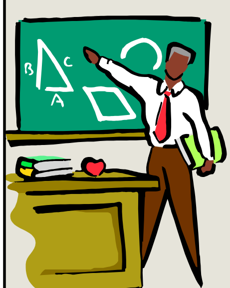
Questions?



Part III: Megaco and soft-switches

- Megaco
- Soft-switches

Megaco / H.248



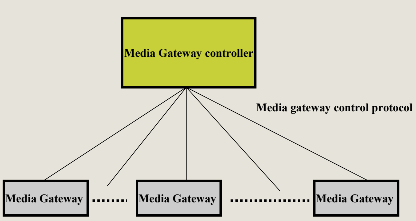
1. Introduction
2. Genesis
3. Concepts
4. Protocol
5. Call cases

Megaco/H.248: 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

Megaco/H.248 : 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)

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Megaco/H.248: Concepts - Termination

Source or sink of media
- Persistent (circuit switched) or ephemeral (e.g. RTP)
- IDs
  - Unique or wildcard mechanism (ALL or CHOOSE)
- Properties/descriptors
  - Unique ids
  - Default values
  - Categorization
    - Common (I.e. termination state properties) vs. stream specific
    - For each media stream
      - Local properties
      - Properties of received streams
      - Properties of transmitted streams
    - Mandatory vs. optional
      - Options are grouped in packages

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Megaco/H.248: Concepts - Termination

Examples of properties/descriptors
- Streams
  - Single bidirectional stream
    - Local control: Send only – send/receive …
    - Local: media received
    - Remote: media sent
- Events
  - To be detected by the MG and reported to the controller
    - On hook / Off hook transition
- Signals
  - To be applied to a termination by the MG
    - Tones
    - Announcements
- Digit map
  - Dialling plan residing in the MG
  - Detect and report events received on a termination ..

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Megaco/H.248: Concepts - Context

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)
  - Null context
    - Terminations that are not associated with no other termination (e.g. idle circuit switched lines)
  - Topology
  - Precedence

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Megaco/H.248: 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)

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Megaco/H.248: Protocol - Transactions

Possibility to send several commands in one go

- Transaction Request
- Transaction Reply
- Transaction pending

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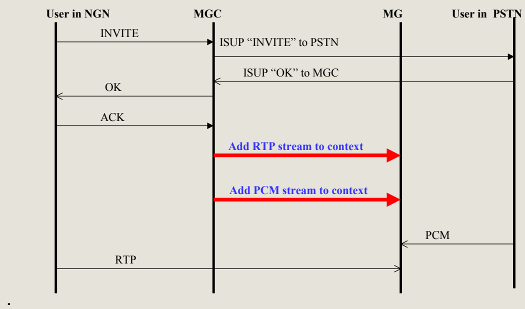
Megaco/H.248: Protocol - Transportation

Several alternatives

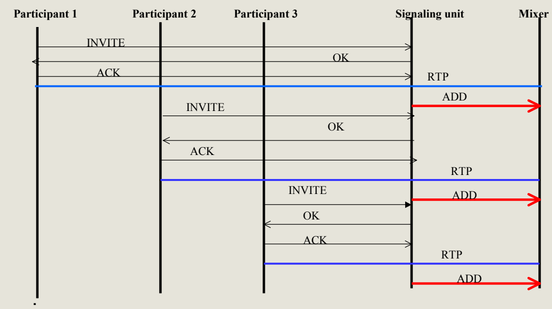
An example

- 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 …

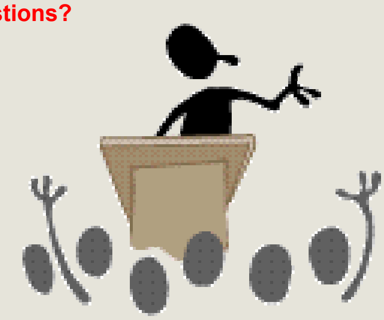


Megaco/H.248: Megaco IP phones

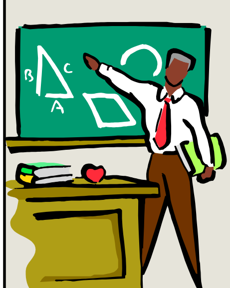
Phone considered as a media gateway …

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

Questions?



Soft-switches



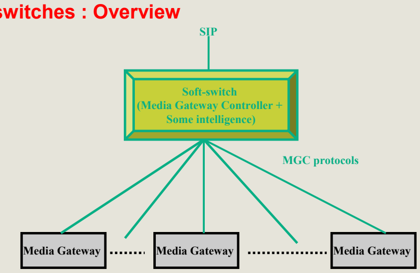
1. Introduction
2. Overview
3. A simplified call case

Soft-switch: Introduction

A "side effect" of media gateway decomposition

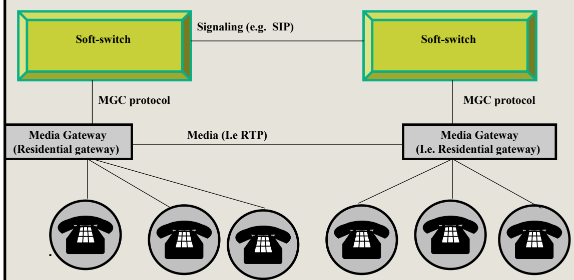
- Aggressively promoted by the soft-switch consortium, now known as the International Packet Communication Consortium (IPCC)
  - Adoption of existing standards (e.g. SIP, H.323, MGCP, Megaco)
- Gateway controller (plus some additional features) acts as a switch
  - Switching in software instead of hardware
- 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

Soft-switches : Overview



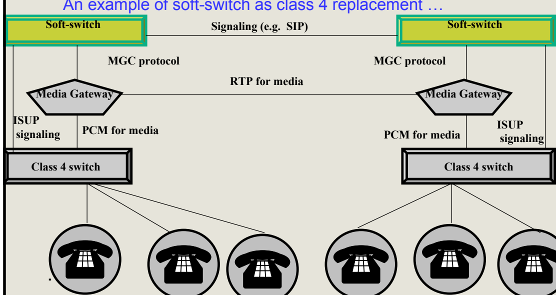
Soft-switches : Overview

An example of soft-switch as class 5 replacement …

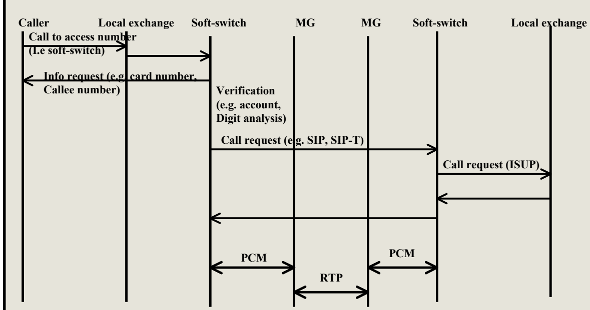


Soft-switches : Overview

An example of soft-switch as class 4 replacement …



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

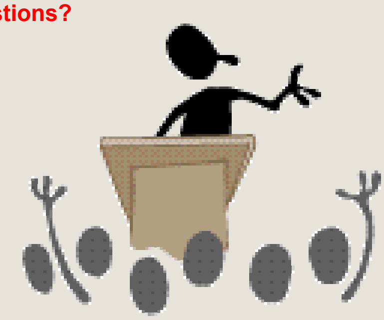


References …

1. T. Taylor, Megaco / H.248: A new Standard for Media Gateway Control, IEEE Communications Magazine, Ocotber 2000

2. Additional references on Megaco/H.248
   RFC 3525 (The protocol)
   RFC 3054 (IP Phone)

Questions?



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Part IV: Value Added Services for SIP

- SIP CGI
- SIP servlets

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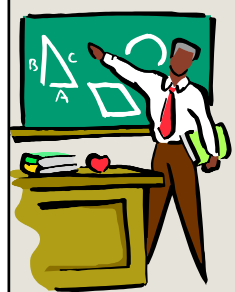
Introduction



1. Value added services
2. Service life cycle
3. Service engineering

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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

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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

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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

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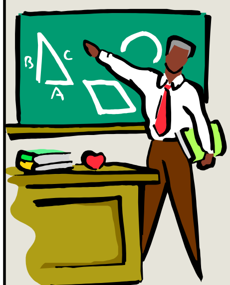
Service Engineering ...

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.

SIP CGI



Introduction ...

Key features

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

.

HTTP ...

Object oriented application level request/reply protocol for distributed multimedia information systems

- Clients
  - establish connections
- User agents
  - Initiate requests
- Servers
  - accept connection, serve requests and send responses back
  - Origin servers:
    - servers on which resources are created and reside
  - Resource
    - data object or service
- Proxies
  - act as both servers and clients

.

HTTP ...

Message

- Type: request or reply
- Headers
  - General header
    - Applicable to both request and reply (e.g. date)
  - Request (or reply) header
    - Method to be applied to the resource (e.g. GET, POST)
    - Resource id
    - Protocol version
    - Additional information (e.g. host, user agent)
    - In case of reply: status line
  - Entity header: Optional information on body
- Body (optional) (e.g. HTML file)
- length

.

HTTP CGI ...

Creation of dynamic Web content

- Script that can work with most programming language
- Generate resource identified in a request on the fly
  - interface between HTTP request and data bases
  - Forms
  - Dynamic information (e.g. date, number of visitors)

Environment variables allow the script to access

- HTTP headers
- Non request specific information (e.g. server host name)

HTTP CGI ...

Pros

- Programming language independence

Cons

- Poor performance
  - Scripts are not persistent: connection to a data base needs to be established each time
- Lack of scalability
  - Scripts need to reside on same server as resource

SIP CGI ...

Examples of adjustments

- Script output is not necessarily the response to send
  - Case of call forward
    - Script will instruct the server to proxy the request to right location
- Scripts are persistent
    - Several interactions are required between script and server for some services

.

SIP CGI ...

Algorithm implemented by a script for call forward

- Get the destination from the SIP request
  - Done by retrieving the To_Field from the environment variable HTTP_TO
- Obtain the forwarding address from a data base
- Forward the call
  - Done by using the CGI-PROXY-REQUEST-TO CGI action

.

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
- Language independence

Cons:

- CGI is less and less used in the Web world
- SIP CGI is not exactly the same thing as HTTP CGI
- Lack of scalability (e.g. scripts need to reside on same server)
- Performance issues

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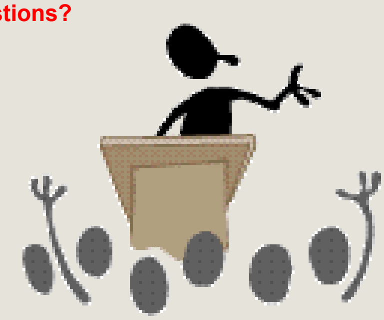
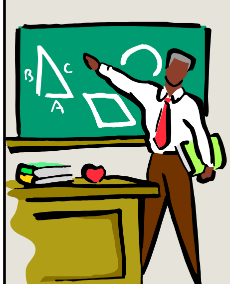
Questions?



SIP Servlet API



1. Introduction
2. HTTP servlet API
3. SIP servlet API
4. Examples
5. 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 HTTP servlet API
  - HTTP 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

.

HTTP servlet API ...

Creation of dynamic Web content

- Servlet
  - Java component
  - Generate content on the fly, just like HTTP CGI
    - interface between HTTP request and data bases
    - Forms
    - Dynamic information (e.g. date, number of visitors)

HTTP servlet API ...

Servlet container (also know as servlet engine)

- Servlet container (or servlet engine)
  - Contains the servlets
  - Manage the servlets through their life cycle
    - Creation
    - Initialisation
    - Destruction
    - Receives and decodes of HTTP requests
    - Encodes and sends of HTTP responses

HTTP servlet API ...

Pros

Address most HTTP CGI shortcomings

- **Performance**
  - Can keep data base connections open
- **Scalability**
  - Servlet containers can be accessed remotely

Cons

- Language dependence

SIP servlet API...

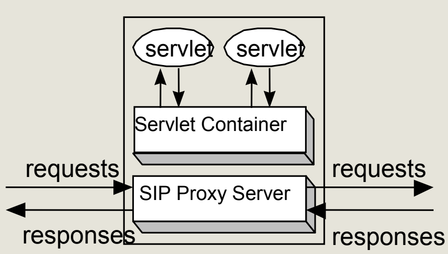
Adjustments made to HTTP servlet:

- 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

.

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

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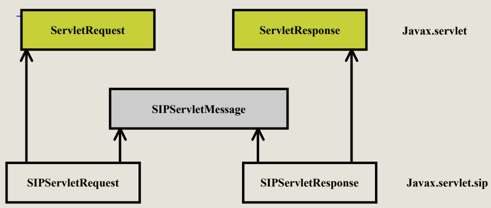
SIP servlet request response hierarchy...

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:
- SIP Servlet is not exactly the same thing as HTTP Servlet
- Language dependence

.

References ...

SIP specific approaches

IETF RFCs: SIP CGI - RFC 3050

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Questions?