Advanced Networking

Voice over IP: Introduction and H.323 standard

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VoIP: Integrating Services

• Voice on IP Networks is just “another application”

• Nothing “special” or “specialized” as traditional telephony, where the network and the service are joint, coupled and sinergic

• VoIP is realized through end-to-end application level protocols, normally not strictly tailored for voice

• Is QoS required?
Plain Old Telephone Service (POTS)

functional intelligence

signaling network
SS7

transport network

silly user terminals

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IP services

intelligence and functions are in hosts: i.e. terminal devices

a single network for signaling and transport

packet based transport network
Architectural difference

Telephone Network

- User Plane
  - User Protocols
    - physical layer (with switching)
  - Control Plane
    - OMAP
    - ASE
    - TCAP
    - ISDN-UP
    - SCFP
    - MTP - 3
    - MTP - 2

VoIP Network

- User Plane
  - Media
  - Signaling
    - RTP
    - UDP
    - TCP
  - IP
  - network
  - user

Advanced Networking – VoIP: Intro/H.323
A Telephone network ...

Switching Exchange

Switching Exchange

User data

Signaling

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Hierarchical organization

Hierarchy levels, Names and details are not uniform across countries. Architecture is still biased by the original monopoly system. The structure was tailored and optimized for voice transport.
Voice and its transmission

- The voice signal is transmitted with analogic technique on the local loop, filtered between 300 and 3400 Hz to allow direct current for powering the phone and to limit the signal bandwidth to a known extent.
- The local exchange immediately converts the analogic signal to digital PCM.

![Graph showing the frequency range of the phone band from 300 Hz to 3400 Hz]
PCM: Sampling and Quantizing

Amplitude vs. time

[Diagram showing the process of sampling and quantizing a signal]

Amplitude

Time
PCM

- PCM (Pulse Code Modulation) encoding is nothing else than sampling and quantizing (with non-linear quantization for telephone networks)
- Linear quantizing means equal intervals; non linear (companding) means different intervals as a function of amplitude
  - Linear PCM: CD (~44 kHz, 16 bit =~ 1.5 Mbit/s)
  - Companding PCM: telephones (8kHz, 8 bit = 64kbit/s). Based on the fact that human ear sensitivity is logarithmic
“A” Compression Law

\[
Y = \begin{cases} 
\frac{A}{1 + \ln(A)} X \\
\frac{\text{sgn}(X)(1 + \ln|AX|)}{1 + \ln(A)} & \text{if } 1/A < X < 1 \\
A = 87.6 \\
X = \frac{V}{V_{\text{max}}} 
\end{cases}
\]

law A
Service-specific problems

• Voice is “just another service”, but ...
• Is it possible to realize e-t-e conversational services without involving the network layer?
• Signaling in telephony has application-level functionalities
  - access
  - callee identification
  - negotiation of characteristics and quality
  - billing and accounting ...

• But also control function on the transport channel
  - routing and setup
  - resource finding and reservation
Service-specific problems

• Application level signaling are simplified by the IP e-t-e approach 😊
• Network services for the control of the channel (e.g. QoS) simply do not exist in IP ☹
• Routing is not controllable (no alternate routing), hot-swap reliability is not present, QoS control is almost impossible unless by “circuit-like” dimensioning. ☹
Real-Time Transport in IP

• Real Time (Transport) Protocol
• Developed by Audio Video Transport Working Group of IETF
• RFC 1889 obsoleted by 3550/3551
• It is an add-on to UDP building a connection-oriented unreliable channel
• Adds and header with information for:
  - Multimedia data management (coding, timestamping, etc.)
  - Error and QoS control (feedback on the reverse channel)
RTP: characteristics and functionalities

- Independent from the PHY (obvious!!!)
- Scalable
  - Unicast e multicast
- Defines separate logical channels for data and control
  - indeed a “pair” of protocols RTP-RTC\textsubscript{ontrolP}
- Packet reordering at destination
- Delay jitter equalization with buffers (in addition to the playout buffer of the application)
- Sender identification
- Intra-media synchronization
- No predefined Port, but must be even
### RTP: header format

<table>
<thead>
<tr>
<th>Ver.</th>
<th>P</th>
<th>X</th>
<th>CSRC ct.</th>
<th>M</th>
<th>payload type</th>
<th>sequence number</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Timestamp</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>synchronization source identifier (SSRC)</td>
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<td></td>
<td></td>
<td></td>
<td>content source identifiers (CSRC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Extension Headers (optional)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Data</td>
</tr>
</tbody>
</table>
The RTP header (12 bytes)

- **Ver.(2 bits):** Version of the protocol. Current is 2
- **P (1 bit):** Indicate if there are extra padding bytes at the end of the RTP packet.
- **X (1 bit):** Extensions to the protocol used (ELH present)
- **CC (4 bits):** Number of CSRC identifiers that follow the fixed header
- **M (1 bit):** If set means that the current data has some special relevance for the application defined in a profile (external to the protocol)
- **PT (7 bits):** Format of the payload and its interpretation by the application
- **SSRC:** Indicates the synchronization source and timing
- **Extension header:** Length of the extension (EHL=extension header length) in 32bit units, excluding the 32bits of the extension header
RTCP

• **Real Time Control Protocol**

• **Functionalities:**
  - Data Distribution Control
  - Session information advertisement (during the session, not for setup)
  - QoS feedback
  - Error reporting
  - ...

• **RTCP messages are sent on RTP-port+1**
VoIP: Signaling Protocols

Standardized

**H.323**

**SIP**

**H.248**

Proprietary

[Skype]

[Wengo]

[Asterisk]

(IAX, open source)

[Yahoo Messenger]

[AOL Instant Messenger]

(Skinny)

... and many many more...
Applications based on “VoIP” protocols

- Collaboration
- Calendar
- Instant Messaging
- Web Application
- Video Conferencing
- Audio Conferencing
- Voice Messaging
- Email
- Telephone Services

Applications based on VoIP protocols include email, voice messaging, collaboration, calendar, instant messaging, video conferencing, audio conferencing, and telephone services.
Brief History of VoIP (1)

• Sharing expensive lines (end of ’90)
  – VoIP enters the enterprise market as a way to save telecom (transmission) cost by using excess data capacity for Voice
  – using the same lines for data and voice communication
    • utilizing existing Local Area Networks (LANs) and WAN connectivity for voice communication, i.e. reduce enterprises’ bill from PSTN operator
  – ITU-T promotes H.323 as protocol (ISDN-style VoIP protocol)
Brief History of VoIP (2)

• Network Convergence (beginning of ‘00)
  – network convergence:
    • data over ISDN was initially successful in some countries (Ger, J) but usage price was high and bandwidth was soon too limited
    • when Internet bandwidth became abundant – VoIP success started
  – IETF completes standardization of an Internet-style VoIP signaling protocol: Session Initiation Protocol (SIP) and media transport protocol (RTP)
  – Internet (IP) becomes the new Integrated Services Digital Network
  – Operator’s convergence began with VoIP in the backbone
  – only lately moving to the access (2005+)
Brief History of VoIP (3)

- SIP becomes the dominant VoIP protocol (‘00 until now)
  - H.323 had the earlier start, but more oriented towards local networks
  - ISDN-style H.323 was more liked by traditional operators
  - SIP is a text-based protocol on top of IP, much like HTTP and XML
    - therefore easy to understand for IP and/or web experts
    - SIP better suited for large scale applications
    - efficiency is poor
    - security threats
  - but SIP became the choice of Internet community
    - Standardized by IETF
Brief History of VoIP (4)

- **breakthrough:** SIP chosen by 3GPP as basis for IMS, i.e., all multimedia services (including VoIP) in 3G

- The consumer segment becomes aware of VoIP
  - Skype clients are widespread
    - using proprietary protocols
  - consumer market is not interested in standards – only costs
  - the business model of Skype – owned by ebay – is “the whole world can talk for free” – revenue is made through arbitrage:
    - Skype out / Skype in – Gateway to PSTN
    - advertising
  - advertisements, lack of privacy/security, quality are the price consumers pay
Today’s Situation

Three VoIP market segments
1. enterprise
2. public operators
3. consumer

What about protocols?
- H.323 is still in the market but will probably die sooner or later ⇒ no point to get into H.323 market in 2006/7/8/9/....
- SIP is already dominating
  - today new investments are based on SIP
  - SIP large scale deployment still in the beginning
  - already dominating the corporate market
  - entering the operator market
- Proprietary protocols, e.g. Skype, are competing in consumer market only
Signaling and Service Protocols

• H.323
  - Vertical, Hierarchic, Complex, Rigid, Omni-comprehensive “LAN oriented”, not easy to integrate with PSTN

• SIP
  - Horizontale, Flat, Simple, “WAN oriented”, impossible to integrate with PSTN

• MeGaCo (H.248)
  - Vertical, Hierarchical, Complementary to H.323/SIP, Separates data and signaling for management, easy support for soft-switches PSTN-oriented, used locally to control media-gateways, not ment as “entire system”
Protocolli

Internet Telephony

H.323

MeGaCo
H.248

SIP

Need a Gateway!!

device control
media gateway management
Local IP trunking (remote terminals)

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Protocols “philosophy”

Internet Telephony
- voice oriented, try to emulate POTS on top of IP
- “ISDN-like” signaling, protocol piles separated for signaling and data
- aims at the integration with SS7

Internet Telephony
- VoIP $\Rightarrow$ Y.A.I.S. (Yet Another Internet Service) like -casting, conferencing, ...
- voice will be a tiny fraction of the traffic
- integrates voice with mail, web, etc.
- telephone is just a particular case of voice, which is a particular case of media, and sessions can be multi-media
Standard protocols: H.323

- H.323: “Packet-based multimedia communications systems”
  - recommendation from ITU-T
  - used to establish, modify, terminate multimedia sessions (e.g. VoIP calls)
  - it is based on H.320 (ISDN Videoconferencing)
    - multistage signaling
    - good interoperability with PSTN
    - it inherits its complexity
    - recent recommendations extend it to wide deployments
  - some operators deployments are still H.323-based
    - many operators have already SIP in their core network
Standard protocols: SIP

- SIP: Session Initiation Protocol
  - IETF standard
  - used to establish, modify, terminate multimedia sessions (e.g. VoIP calls)
  - it is based on HTTP (light protocol)
  - it inherits its vulnerabilities
  - easily extensible

- It supports name mapping and redirection services transparently
  - personal mobility: one single externally visible identifier regardless of the network location

- Where is SIP used?
  - corporate deployments
  - 3GPP IMS (PS signaling protocol)
  - TISPAN NGN will be based on core IMS and thus on SIP as well
VoIP: architectural elements (I)

- Terminals (end-points)
  - hardware clients
  - software clients
  - optional
    - video codec
    - data transmission
    - instant message
    - presence
VoIP: architectural elements (II)

- Gateway
  - generic: an interface between two worlds
  - specific: interface between packet-based networks and circuit switched networks or between different architectures in packet-based networks (e.g. SIP—Skype)
VoIP: architectural elements (III)

- Rendezvous server
  - H.323 world: Gatekeeper
  - SIP world: Proxy server
  - Main functionalities
    - Managing entities in its domain
      - Endpoint registration
    - Address translation
      - User identity to terminal location
    - Call routing
      - Next hop location

- Additional servers
  - Application servers
    - Registration servers
    - Conferencing server
    - Presence server
    - Etc.
H.323: Delving deeper

• It’s the first architecture developed for audio/video services on packet (not necessarily IP!!) networks

• It has been defined in the “telco” (ITU-T) world, it’s probably still the most diffused protocol for VoIP ..........

• .... but just because it was the first one ....
The architecture is derived from video-conferencing in LANs services defined in the ’80s and early ’90s

- Version 1 (1996) Multimedia over LAN
- Version 3 (1999) + Communications across administrative domains
- Version 5 (2003) + Use of URLs and DNS + Video conferencing support + ...
H.323 architecture

- Enables direct end-to-end signaling
  - terminal interconnection
  - logical channels (for the media) set up
- Uses directly the IP address and the TCP/UDP ports
H.323 elements (logical devices)

- **End-point**: terminals enabled for communications
- **Gateway**: inter-working unit with other networks (PSTN/ISDN and SIP in particular)
- **Gatekeeper**: controls communications (central office)
- **MCU** (Multipoint Control Unit): multicast communications (conferencing) and supplemental services
H.323: compulsory components

- H.225 (connection and status control):
  - Q.931 user signaling
  - RAS (Registration, Authentication and Status) endpoint to gatekeeper signaling
- H.245: e-t-e signaling on terminal capabilities and “media” that support information
- RTP/RTCP: transport and flow control
- G.711: mandatory coding (64 kbps) all other codecs are optional!!
H.323

communication between “internal” terminals

H.225/RAS
H.245
RTP/DATI

gatekeeper

end point
H.323

communication between 1 internal and 1 external terminal

gatekeeper

gateway

eend point

eend point

H.225/RAS

H.245

RTP/DATI

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H.323
communication between external terminals

H.225/RAS
H.245
RTP/DATI

gatekeeper

end point
gateway

eend point

gatekeeper

communication between external terminals

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H.323 architecture

• A H.323 network is composed by one or more “zones”

• One zone is a logical ensable of H.323 devices managed by a single gatekeeper

• Zone boundaries can be based on administrative limits, addressing structures, geography, etc.

• Calls involving more zones are managed involving more gatekeepers, a working mode defined in Version 3 and available in devices 2001-02
Gatekeeper

- It’s the “intelligent” device of H.323 architecture and services
- Each gatekeeper manages a “zone” (a collection of end-points, gateways, MCUs)
- It has the following compulsory functionalities:
  - Admission Control (verification of end-points authorization to place and receive calls)
  - Address translation (telephone alias <-> IP)
  - Bandwidth control (if required by the call)
  - Zone management
Gatekeeper

- May implement optional functions and features
  - Authorization
  - Resource Management
  - Call control signalling (act as rendezvous point also for terminal-to-terminal signaling - H.245)
  - Resource Reservation (for end-point not able to run reservation protocols like RSVP)
  - Call management (multimedia calls and complex services)
  - Gatekeeper management information (remote management via SNMP on standard MIBs)
  - Directory services
H.323 architecture

- Il Gatekeeper can be a proxy signaling
- May be the interface toward additional services
- May also force data-flow switching, behaving as a traditional PBX (computational and traffic burden)
H.323 architecture

- **Gateways** are devoted to interworking with other architectures, and specifically with PSTN
- Also other VoIP architectures (SIP, Skinny, Asterisk (IAX), skype and other proprietary protocols)
A "zone"

H.323

Client H.323

Client H.323

IP phone

Gatekeeper

Router

Internet or intranet

Analog phones

PSTN or ISDN or PBX

H.320 (ISDN)

H.324 (POTS)

IP phone

Client H.323

IP phone

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H.323 protocol stack

User Interface

Audio I/O

Video I/O

Data

System Interface

H.225.0

RAS

H.245

G.711, G.722
G.723.1
G.728, G.729

H.261

H.263

T.120

Receive path delay

RTCP

RTP

IETF

UDP

TPKT

TCP

UDP

UDP
In this phase the caller obtains:
- Logical permits and authorization
- The callee TCP/IP address to send Setup signaling
In this phase end-points exchange TCP ports to open and H.245 session
H.323: H.245 phase

End-points exchange terminal characteristics and UDP ports for the RTP/RTCP flows.

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**H.323: media exchange phase**

![H.323 Diagram](image)
H.225: Annex G

- Introduces methods to implement fundamental services when a call is set-up across multiple administrative domains
  - global address resolution
  - access authorization
  - usage reporting

- Introduces a new network element: the Border Element
H. 225 : Annex G

Is an interface between gatekeepers in different administrative domains.
H. 225 : Annex G

- Border Elements (and Clearing Houses) exchange information on:
  - reachability
  - cost

- “I’ll route calls to 1303*, and I’ll charge 8 cents a minute peak, 5 cents a minute off peak”
- “I can resolve everything for 33*”
- “I can resolve everything for *@cisco.com”
H.323

H.225.0 architecture under Annex G