Voice over IP Quality of Service and Reliability

David Tipper
Associate Professor
Department of Information Science and Telecommunications
University of Pittsburgh
tipper@tele.pitt.edu
http://www.sis.pitt.edu/~dtipper/tipper.html

Course Outline

- Introduction (Ch. 1)
  - Overview of Voice over IP systems, QoS issues
- Speech Coding and Voice Quality/Cognition (Ch. 2)
  - A/D converters, Vocoders, Hybrid coding, Perception
- VoIP Network Quality Factors (Ch. 3)
  - Packetization, loss, delay, jitter, echo, reliability
- Internet QoS and VoIP (Ch. 4)
- Network Dimensioning and Reliability (Ch. 5)
- Exam
• Overview of VoIP technology and issues
  – Review of Telephony
  – What is VoIP and Why?
  – Basic IP telephony models
  – VoIP Protocols: H.323, RTP, SIP, MGCP, RTSP
  – QoS issues

Voice over IP Systems

• Voice over IP (VoIP):
  – Technology providing voice telephony services using the Internet Protocol.
  – Involves converting analog audio to digital, packetization and transport over an IP network.
  – Internet Telephony (IP Telephony) more general concept then VoIP
    • Includes VoIP plus signaling/call control and telephony features (call waiting, 800 calls, three way calling, billing, etc.), interworking with existing telephone network
  – Will use VoIP = Internet Telephony = IP Telephony interchangeably
  – Review of Telephony
Public Switched Telephone Network (PSTN)

- Long History 1876 invention of telephone
  - Switch from Analog to digital phones began in 60’s
  - Advanced services and digital signaling added 70’s
  - Cell phones 90’s (now more than wired phones)

- PSTN characteristics
  - Developed for single basic service: two-way voice
  - Circuit switched (guaranteed bandwidth) end-to-end connection
  - 64 kb/s continuous voice transmission 8000 samples/sec 8 bits per sample,
  - Everything clocked a multiple of 125 $\mu$s
  - low end-to-end delay
  - Call establishment “out-of-band” using packet-switched signaling system (SS7)

PSTN Architecture

- Standardized interface between CO and end-system => digital handsets, cordless/cellular phones
- CO supplies the power for the phone (power life line when electricity out)
- Hierarchically allocated telephone number space (country, area, local)
Transmission Multiplexing

- Calls multiplexed to higher data rates in core
- Multiplexed trunks can be multiplexed further

<table>
<thead>
<tr>
<th>Digital Signal Number</th>
<th>Number of previous level circuits</th>
<th>Number of voice circuits</th>
<th>Bandwidth</th>
</tr>
</thead>
<tbody>
<tr>
<td>DS0</td>
<td>1</td>
<td></td>
<td>64 Kbps</td>
</tr>
<tr>
<td>DS1</td>
<td>24</td>
<td>24</td>
<td>1.544Mbps</td>
</tr>
<tr>
<td>DS2</td>
<td>4</td>
<td>96</td>
<td>6.312 Mbps</td>
</tr>
<tr>
<td>DS3</td>
<td>7</td>
<td>672</td>
<td>44.736 Mbps</td>
</tr>
</tbody>
</table>

Telephone Switch

- Transfers data from an input to an output
- Switching systems come in two parts: switch and switch controller
- Switch: time and space division
Signaling

- Signaling/switch controllers are special purpose computers
- Linked by their own internal packet switched network
  - Common Channel Interoffice Signaling (CCIS) network
- Messages on CCIS conform to Signaling System 7 (SS7)
- Switch controller is in the control plane
  - Does not touch voice samples
  - Circuit-setup (i.e. the signaling call) is what is routed.
  - Voice then follows route, and claims reserved resources.

Signaling (contd)

- Manages the network
  - call routing (collect dialstring and forward call)
  - alarms (ring bell at receiver)
  - billing
  - directory lookup (for 800/888 calls)
- Switch controller: keep track of state of every endpoint
- State transition diagram
Features of PSTN

- **Resource reservation aspects:**
  - Resource reservation is coupled with path reservation
  - Connections need resources (same 64kbps)
  - Signaling to reserve resources and the path
  - Stable load
    - Network built for voice only.
    - Can predict pairwise load throughout the day
    - Can choose optimal routes in advance – high call completion rate
  - Simplified topology:
    - Very highly connected network

- **Technology and economic aspects:**
  - Extremely reliable switches (99.999% availability – 5mins downtime per year)
  - End-systems (phones) dumb because computation was non-existent in early 1900s
  - Advanced features provided by signaling network (911, 800 number, etc)
  - Regulated monopoly or oglopolo in many areas

- **Organizational aspects:**
  - Single organization controls large sections of the network (core, LEC)
  - Afford the scale economics to build expensive
  - International Telecommunications union
  - Since the Plain old Telephone (POTS) in place and works why VoIP?

---

Trends: Data vs Voice Traffic

Service provider thinking - since we are building future networks for data, can we slowly junk the voice infrastructure and move over to VoIP?
New services? Security with encryption, multi-media web, etc.
Internet

- Internet characteristics:
  - Internet Protocol developed in 60’s
  - Network of networks - no centralized control/management
  - Standards developed by open consensus IETF
  - Dynamic routing – adaptive to congestion and failure
  - Largely unregulated
  - Packet switching – bandwidth as needed
  - Best-effort service
    (no performance guarantees)
  - Packet-by-packet variations – packets can take different routes, arrive out of order, be dropped, etc.

Packet Switching

- Cost: self-descriptive header per-packet, buffering and delays due to statistical multiplexing at switches.
- Need to either reserve resources or dynamically detect and adapt to overload for stability
Voice over IP Systems

- Voice over Internet Protocol (VoIP):
  - Technology providing voice telephony services using the Internet Protocol.
  - Involves converting analog audio to digital, packetization and transport over an IP network.
  - Internet Telephony (IP Telephony) more general concept then VoIP includes VoIP plus signaling/call control and telephony features (call waiting, 800 calls, three way calling, billing, etc.), interworking with existing telephone network
  - Will use VoIP and Internet Telephony interchangeably
  - Several categories/configurations for VoIP/Internet Telephony each with different motivation

VoIP Configurations/Categories

1. End-to-End over the Internet
   - Motivation – free/cheap phone calls
2. Enterprise VoIP
   - Small IP phone deployments, IP PBX, Call manager, etc.
   - Managed vs. Unmanaged
3. IP-switched local access phone carrier:
   - Motivation cheaper phone service – competitive local service long distance toll bypass services
   - VoIP wholesale, using equipment by vendors such as Cisco, Lucent, Avaya, etc.
4. VoIP through middleware
5. IP-switched service provider core
6. Telco Grade: Local and long distance service
   - Last mile delivery, total phone services, high dependability and availability.
1. End –to-End VoIP

- End-to-End VoIP telephony over the Internet (using PCs)
  - Skype, Microsoft Netmeeting, Asterisk*, Marratech, etc.
  - No gateway with public switched telephone network (PSTN)
  - PSTN by-passed except for access (dialup, DSL)
  - An Internet application between users over unmanaged network
  - Not possible to detect or regulate
  - Business model?
  - Quality, features less than PSTN

2. Enterprise VoIP

- Motivation Integrating private voice & data network
  - No traditional economy of scale
    - Integrated network must be over-designed →
    - To provide acceptable voice & data service
  - $ managing 2 networks >> $ managing 1??
  - Which technology for common network?
    - In Enterprise data traffic > voice traffic then integrate on one
    - Managed or Unmanaged Enterprise core??

- Issues Quality and Reliability
  - Interest in this in the US → example General Motors (GM)
Enterprise Example

Corporate/Campus

Private Branch Exchange

External line

Telephone switch

Corporate/Campus LAN

Internet

If Managed then over Private network or VPN rather than Internet

Corporate/Campus

IP PBX

Another campus

Private Branch Exchange

External line

Telephone switch

Corporate/Campus LAN

Internet

Enterprise IP Telephony
3. VoIP-Switched CLEC

- Competitive local phone provider
  - For example Cable TV
- Home controller (set-top box)
  - Could be a card in a PC, Also provides CATV
- Call Processing at Cable head-end
  - Provides features, 800,…
- Can control end-to-end delay
  - Local: Private internet, not The Internet
  - Long Distance: Inter-Exchange Carrier
- Example Comcast in U.S.

4. VoIP thru Middleware

- PC users only, with IP interface
  - Excludes non-PC market
  - Subscriber calls ISP thru local phone company (LEC)
- Artificial economy - don’t pay:
  - LD access charge, LD excise tax (voice & data)
- ISP does Call Processing at its gateway
  - Features, 800, etc?
  - Signaling?
  - Charge users?, Include in monthly flat rate?
  - AOL is offering
5. IP-Switched IXC Core

- Conventional telephone
  - Conventional local service, call processing, SS7 etc.
  - Subscriber calls an IXC long distance carrier
- True economy - subscriber pays:
  - LD access charge, LD excise tax
- IXC switches calls internally by IP
  - Private internet, DS0-IP interface at IXC boundary
  - LEC at both ends
- No advantage for user at PC
- Quest is doing this in the U.S., Telus in Canada

VoIP-Switched End-to-End Telco Grade Carrier

- Connect VoIP phones and PCs
  - Can't push any cost into ILEC
  - CATV leverage (bundling)
  - Bypass LD access charge
    - Avoid Universal Service in U.S., but pay all excise taxes
- Economics? Features? Quality?
IP Telephony - Regulatory Issues

- Bypassing the “Access Charge”
  - Local exchange carriers have some legitimate cost
  - Long Distance subsidy of Universal Service
  - Inequity → “bypass” industry
    - For enterprise, not residential

- Why no charge for Internet access?
  - IXC could “looks like” an ISP
  - Significant cost/price advantage

- VoIP has variety of implementations to go along with different configurations/categories
  - Standards bodies (IETF, ITU, ETSI, etc.)

VoIP Camps

<table>
<thead>
<tr>
<th>Conference</th>
<th>Netheads</th>
<th>Circuit switch</th>
<th>“Convergence”</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.323</td>
<td>SIP</td>
<td>“Softswitch”</td>
<td>ITU standards</td>
</tr>
<tr>
<td>ISDN LAN conferencing</td>
<td>I-multimedia WWW</td>
<td>Call Agent SIP &amp; H.323</td>
<td>BISDN, AIN H.xxx, SIP</td>
</tr>
<tr>
<td>IP</td>
<td>IP</td>
<td>IP</td>
<td>“any packet”</td>
</tr>
</tbody>
</table>

Our focus
VoIP Protocols

- No one standard for Signaling:
  - SIP, H.323, MGCP
- RTP for transport
  - RTP (Real-Time Transport Protocol)
  - RTP is a UDP stream with no intelligence for QoS or resource reservation
  - Contains a packet number for detection of packet loss and re-sequencing of out of order packets.
  - Unidirectional: two streams in any call

How they fit in: The ISO Model

<table>
<thead>
<tr>
<th>ISO Model Layer</th>
<th>Protocol or Standard</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presentation</td>
<td>VoIP CODECS</td>
</tr>
<tr>
<td>Session</td>
<td>H.323 &amp; SIP &amp; MGCP</td>
</tr>
<tr>
<td>Transport</td>
<td>RTP / UDP / TCP</td>
</tr>
<tr>
<td>Network</td>
<td>IP</td>
</tr>
<tr>
<td>Data Link</td>
<td>ATM, FR, PPP, Ethernet, SONET, etc.</td>
</tr>
</tbody>
</table>
VoIP Signaling

- Most VoIP implementations use H.323 protocol
  - Same protocol that is used for IP video.

- H.323 Multimedia Standard developed by ITU
  - H.225 RAS - Registration, Admission, Status
  - Q.931 - Call Signaling (Setup & Termination)
  - H.245 - Call Control (Preferences, Flow Control, etc.)
  - Lots of G.7XX CODECS for audio

H.323 Entities

- Terminals
  - An endpoint that supports 2-way streaming with another H.323 terminal or gateway
  - Originates and terminates calls
  - Includes videoconferencing stations, hard phones, & soft phones

- Gateways (Optional but really useful)
  - Device that connects H.323 voice network to non-H.323 voice network (SIP or PSTN)
  - Allows H.323 terminals to communicate with non-H.323 terminals

- Gatekeepers (Also optional)
  - Provides address translation (H.323 & E.164 to IP)
  - Admission control for H.323 terminals and gateways
  - Manage bandwidth allocation

- MCU (multipoint control unit)
  - MC – multipoint controller
    - Routes call and control signaling to ensure endpoint compatibility
  - MP – multipoint processor
    - Switches, mixes and processes voice and video streams to conferencing equipment
Call Setup using H.225 RAS

• Registration, Admission and Status (RAS), is responsible for registration, admission, and disengaging procedures between H.323 Gatekeeper and Gateway.
• Discovery: GRQ, GCF, GCR
  – Unicast or multicast Discovery
• Registration by terminals, Gateways & MCUs using H.323 ID
  – RRQ Registration Request
  – RCF Registration Confirm
  – RRJ Registration Reject
  – URQ Un-registration Request
  – URF Un-registration Confirm
  – URJ Un-registration Confirm

H.323 – H.225 RAS Messages

• LRQ – location request
  – Gatekeeper A requests contact information from directory gatekeeper.
• LCF – location confirm
  – Gatekeeper B returns IP address of destination gateway to gatekeeper A.
• ARQ – admission request
  – Gateway A requests admission to make a call.
• ACF – admission confirm
  – Gatekeeper A responds with IP address of destination gateway.
Signaling using Q.931 messages

- Q.931 is a signaling protocol used to setup, manage, and terminate H.323 connections between endpoints.
  - ARQ, ACF, ARJ Admission messages
  - LRQ, LCF, LRJ Location Request messages
  - IRQ, IRR, IACK, INAK Status messages
  - BRQ, BCF, BRJ, RAI, RAC Bandwidth messages
- Release Complete
  - H.225 (Q.931) call has been released, signaling channel is now open
- Status
  - Sent when unknown call signaling message or a status inquiry message is received
- Status Inquiry
  - Requests a call’s status

H.323 – H.245 Messages

- Establishes logical channels for transmission of H.323 data
- Master/Slave Determination
  - Determines which terminal will be master which will be slave in the call
- Terminal Capability Set
  - Contains information on a terminal’s ability to send and receive multimedia streams
- Open Logical Channel
  - Opens logical channel for transport of multimedia data
- Close Logic Channel
  - Closes the logical channel between two endpoints
H.323 – H.245 Messages

- Request Mode
  - Receive terminal requests type of transportation from a transmit terminal
  - Types of Modes:
    - Video
    - Audio
    - Data
    - Encryption

- Send Terminal Capacity Set
  - Instructs far-end terminal to send transmit and receive capabilities

- End Session Command
  - Indicates the end of the H.245 session

---

H.323 Call Setup via Gatekeepers

1. IP Phone
2. ARQ
3. RIP
4. LRO
5. LCF
6. ACF
7. Q.931 Call Setup
8. ARQ
9. ACF
10. Q.931 Proceed
11. Q.931 Proceed
12. H.245
13. RTP

Site A
- Gateway A
- Gatekeeper A
- Directory GK
Site B
- Gateway B
- Gatekeeper B
- IP Phone
VoIP Signaling Protocols

- Session Initiation Protocol (SIP)
- IETF-standardized *peer-to-peer* signaling protocol (RFC 2543):
  - Locate user given email-style address
  - Setup session (call)
  - (Re-)negotiate call parameters
  - Manual and automatic forwarding
  - Personal mobility: different terminal, same identifier
  - Call center: reach first (load distribution) or reach all (department conference)
  - Terminate and transfer calls
- Light-weight generic signaling protocol

SIP Addresses Food Chain

yellow pages                      “president of the United States”
                                  \ `WWW search engines`
common names                     “Bill Clinton, Whitehouse”
                                  \ `directory services`
host-independent                 `president@whitehouse.gov`
                                  \ `SIP`
host-specific                    sip:bubba@oval.eop.gov sip:+1-202-456-1111@net2ph
                                  \ `DNS`
IP address                       `198.137.241.30`
SIP Architecture

- SIP uses client/server architecture
  - Client - originates message
  - Server - responds to or forwards message
- Elements:
  - SIP User Agents (SIP Phones)
    - User Agent Client (UAC)
      - Initiates SIP requests
    - User Agent Server (UAS)
      - Returns SIP responses
  - SIP Servers (Proxy or Redirect - used to locate SIP users or to forward messages.)
    - Can be stateless or stateful
  - SIP Gateways:
    - To PSTN for telephony interworking
    - To H.323 for IP Telephony interworking

SIP Messages

- INVITE: Initiates call
  - Session description included in message body
- Re-INVITEs used to change session state
- ACK confirms session establishment, can only be used with INVITE
- BYE terminates a session/call (hanging up)
- CANCEL cancels a pending invite and ringing
- REGISTER: binds a permanent address to current location, may convey user data
- OPTIONS: capability inquiry, that is determine features supported by the other side of the call
SIP-based Architecture

Example Call

- Bob signs up for the service from the web as “bob@himolde.no”
- He registers from multiple phones
- Alice tries to reach Bob
  INVITE ip:Bob.Wilson@himolde.no
- sipd canonicalizes the destination to sip:bob@himolde.no
- sipd rings both e*phone and sipc
- Bob accepts the call from sipc and starts talking
**Another Example**

**SIP operation in proxy mode**

**PSTN to IP Call**
Authentication & Encryption

- SIP supports a variety of approaches:
  - end to end encryption
  - hop by hop encryption
- Proxies can require authentication:
  - Responds to INVITEs with 407 Proxy-Authentication Required
  - Client INVITEs again with Proxy-Authorization header.
  - Stacked!!
- SIP Users can require authentication:
  - Responds to INVITEs with 401 Unauthorized
  - Client INVITEs with Authorization header

SIP vs H.323

- What makes SIP > H.323?
  - Simpler & faster call setup
  - Multicast experience (MBONE)
  - Easier to add third-party services
  - SIP uses UDP for setup gives better timing control than TCP used by H.323
  - SIP uses an HTTP-like syntax much easier to write/debug.
- What makes H.323 > SIP?
  - Separates signaling & media channels
  - Conference control
  - Compatible with Q.931
  - More stable and encompasses broader system (ITU standard)
  - Embedded base of H.323 telephones
- European Telecommunications Standards Institute Tiphon group working on SIP ↔️ H.323 gateway/portal
  http://www.etsi.org
MGCP and Megaco

- Media Gateway Controller Protocol (RFC 2705)
- Controlling Telephony Gateways from *external* call control elements called media gateway controllers (MGC) or call agents
  - Gateways: Eg: RGW : physical interfaces between VoIP network and residences
  - MGCP is a master/slave protocol where all the smarts resides in the gateway controller and not the gateway.
  - Goal: Scalable gateways between VoIP telephony and PSTN
    - Simplify PSTN- VoIP interworking
    - Carriers don’t want H.323’s direct-call model
    - IP signaling must interface to SS7:
  - Successor to MGCP: H.248/Megaco

MGCP Architecture

**Goal: large-scale phone-to-phone VoIP deployments**

RGW: Residential Gateway
TGW: Trunk Gateway
SoftSwitch

- MGCP signaling controller
  - SW that controls calls to the PSTN
  - Billing
  - Assumes/controls Indirect calls
- Evolved - may or may not include:
  - Gatekeeper functionality
  - Directories & Feature SW
  - Conference control $\rightarrow$ VoIP, PSTN mix
  - More later (time permitting)
- Be careful of this word

MGCP Analysis

- What makes MGCP $>$ H.323v1?
  - Better with complex calls & carrier functions
    - Network services, Billing
  - H.323v1 uses TCP (could be changed)
- H.323v2 $>$ MGCP?
  - Better with embedded base
    - PCs, IP telephones
  - Better with intelligent end-points
- It’s a mess
### Comparison of Packet vs. Circuit Switching

<table>
<thead>
<tr>
<th></th>
<th>Circuit</th>
<th>Packet</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Setup</td>
<td>Database / SS 7 Overlay</td>
<td>H.323 &amp; SIP</td>
</tr>
<tr>
<td>Communications Channel</td>
<td>Dedicated</td>
<td>Shared</td>
</tr>
<tr>
<td>Addressing</td>
<td>International Numbering Plan</td>
<td>IPv4 &amp; IPv6</td>
</tr>
</tbody>
</table>

### VoIP Issues and QoS

- VoIP needs to replicate POTs features?
  - 911, Power to phone, 800 numbers, etc
- Quality of Service issues
  - Voice codec choice
  - Echo
  - Delay
  - Delay variation (jitter)
  - Packet loss
  - Reliability/Availability
- Lots of Hype for VoIP the last 7 years or so – reality is small percentage of voice traffic is VoIP
  - Forrester Group predicts 10% of residential voice service VoIP by 2010
  - Hope for VoIP is adoption by business – leading to buildout of service and lower cost