IP Telephony (Voice over IP)
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Textbook

Requirements
- Homework x 2 (Homework I and Homework II) 30%
- Mid-term exam 15%
- Final exam 20%
- Term project (or Homework III) 20%
- Oral presentation 15%

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Course Outline

- Introduction
- Transporting Voice by Using IP - RTP (Real-Time Transport Protocol)/RTCP (RTP Control Protocol)
- H.323
- Session Initiation Protocol (SIP) and ENUM
- VoIP over Network Address Translation (NAT)
- Skype – Voice over Overlay Networks
- Media Gateway Control and the Softswitch Architecture
- VoIP and SS7
- Quality of Service
- Designing a Voice over IP Network
- Mobile All IP Network
- VoIP over Wireless LAN (WLAN)
Next Generation Networks [1/2]

- Internet Telecom & Wireless Communication

Reference: CCL/ITRI
Next Generation Networks [2/2]

- Internet Telecom & Wireless Communication

Reference: CCL/ITRI

IP Telephony
Introduction

Chapter 1
Carrier Grade VoIP

- Carrier grade and VoIP
  - Mutually exclusive
  - A serious alternative for voice communications with enhanced features

- Carrier grade
  - The last time when it fails
  - 99.999% reliability (high reliability)
    - Fully redundant, Self-healing
  - AT&T carries about 300 million voice calls a day (high capacity).
    - Highly scalable
  - Short call setup time, high speech quality
    - No perceptible echo, noticeable delay and annoying noises on the line
  - Interoperability
VoIP

- Transport voice traffic using the Internet Protocol (IP)
- One of the greatest challenges to VoIP is voice quality.
- One of the keys to acceptable voice quality is bandwidth.
- Control and prioritize the access
- Internet: best-effort transfer
  - VoIP ≠ Internet telephony
  - Next generation Telcos
    - Access and bandwidth are better managed.
    - QoS solutions
    - Service-level agreements between providers
A packet-based protocol
- Routing on a packet-by-packet base

Packet transfer with no guarantees
- May not be received in order
- May be lost or severely delayed

TCP/IP
- Retransmission
- Assemble the packets in order
- Congestion control
- Useful for file-transfers and e-mail
Data and Voice

- Data traffic
  - Asynchronous – can be delayed
  - Extremely error sensitive

- Voice traffic
  - Synchronous – the stringent delay requirements
  - More tolerant for errors

- IP is not for voice delivery.

- VoIP must
  - Meet all the requirements for traditional telephony
  - Offer new and attractive capabilities at a lower cost
Why VoIP?

- Why carry voice?
  - Internet supports instant access to anything.
  - However, voice services provide more revenues.
    - Voice is still the killer application.

- Why use IP for voice?
  - Traditional telephony carriers use circuit switching for carrying voice traffic.
  - Circuit-switching is not suitable for multimedia communications.
  - IP: lower equipment cost, lower operating expense, integration of voice and data applications, potentially lower bandwidth requirements, the widespread availability of IP.
Lower Equipment Cost

- PSTN switch
  - Proprietary – hardware, OS, applications
    - New software application development for third parties
  - High operation and management cost
    - Training, support, and feature development
  - Mainframe computer

- The IP world
  - Standard mass-produced computer equipment
  - Application software is quite separate
  - A horizontal business model
    - More open and competition-friendly

- Intelligent Network (IN)
  - does not match the openness and flexibility of IP solutions.
  - A few highly successful services
  - VoIP networks can interwork with Signaling System 7 (SS7) and take advantage of IN services build on SS7.
Voice/Data Integration

- Click-to-talk application
  - Personal communication
  - E-commerce
- Web collaboration
  - Shop on-line with a friend at another location
- Video conferencing
  - Shared whiteboard session
  - With IP multicasting
- IP-based PBX
- IP-based call centers
- IP-based voice mail
- Far more feature-rich than the standard 12-button keypad
Lower Bandwidth Requirements

- **PSTN**
  - G.711 - 64 kbps
  - Human speech frequency < 4K Hz
  - The Nyquist Theorem: 8000 samples per second to fully capture the signal
  - 8K * 8 bits
- **Sophisticated coders**
  - 32kbps, 16kbps, 8kbps, 6.3kbps, 5.3kbps
  - GSM – 13kbps
  - Save more bandwidth by silence suppression
- **Traditional telephony networks can use coders, too.**
  - But it is more difficult.
- **VoIP** – two ends of the call to negotiate the coding scheme
- **The fundamental architecture of VoIP systems lends itself to more transmission-efficient network designs.**
  - Distributed (Bearer traffic can be routed more directly from source to destination.)
The Widespread Availability of IP

- IP
  - LANs and WANs
  - Dial-up Internet access
  - IP applications even reside within hand-held computers and various wireless devices.
  - The ubiquitous presence
- VoFR or VoATM
  - Only for the backbone of the carriers
VoI P Challenges

- VoI P must offer the same reliability and voice quality as traditional circuit-switched telephony.

- Mean Opinion Score (MOS)
  - 5 (Excellent), 4 (Good), 3 (Fair), 2 (Poor), 1 (Bad)
  - International Telecommunication Union Telecommunications Standardization Sector (ITU-T) P.800
  - Toll quality means a MOS of 4.0 or better.
Speech Quality [1/2]

- Must be as good as PSTN

- Delay
  - The round-trip delay
  - Coding/Decoding + Buffering Time + Tx. Time
  - G.114 < 300 ms

- Jitter
  - Delay variation
  - Different routes or queuing times
  - Adjusting to the jitter is difficult.
  - Jitter buffers add delay.
Speech Quality [2/2]

- **Echo**
  - High Delay ===> Echo is Critical

- **Packet Loss**
  - Traditional retransmission cannot meet the real-time requirements

- **Call Set-up Time**
  - Address Translation
  - Directory Access
Managing Access and Prioritizing Traffic

- A single network for a wide range of applications, including data, voice, and video
- Call is admitted if sufficient resources are available
- Different types of traffic are handled in different ways
  - If a network becomes heavily loaded, e-mail traffic should feel the effects before synchronous traffic (such as voice).
- QoS has required a huge effort.
Speech-coding Techniques

- In general, coding techniques are such that speech quality degrades as bandwidth reduces.
  - The relationship is not linear.
- G.711 64kbps 4.3
- G.726 32kbps 4.0
- G.723 (celp) 6.3kbps 3.8
- G.728 16kbps 3.9
- G.729 8kbps 4.0
- GSM 13kbps 3.7
- MOS values are still subjective in nature.
Network Reliability and Scalability

- PSTN system fails
  - 99.999% reliability

- Today’s VoIP solutions
  - Redundancy and load sharing
    - A balance must be struck between network cost and network quality.
    - Finding the right balance is the responsibility of the network architect.
  - Scalable – easy to start on a small scale and then expand as traffic demand increases
VoIP Implementations

- IP-based PBX solutions
  - A single network
  - Enhanced services
VoIP Implementations

- IP voice mail
  - One of the easiest applications
- IP call centers
  - Use the caller ID
  - Automatic call distribution
  - Load the customer’s information on the agent’s desktop
  - Click to talk
VoIP Evolution

1: PC to PC

2: Phone to PC over IP

3: Phone to Phone over IP

4: PC to PC over PSTN

VoIP Terminal

VoIP Terminal

IP Network

Gateway

Gateway

PSTN

PSTN

IP Network

Gateway

PSTN

IP Network

Gateway

PSTN

IP Network

Gateway

PSTN