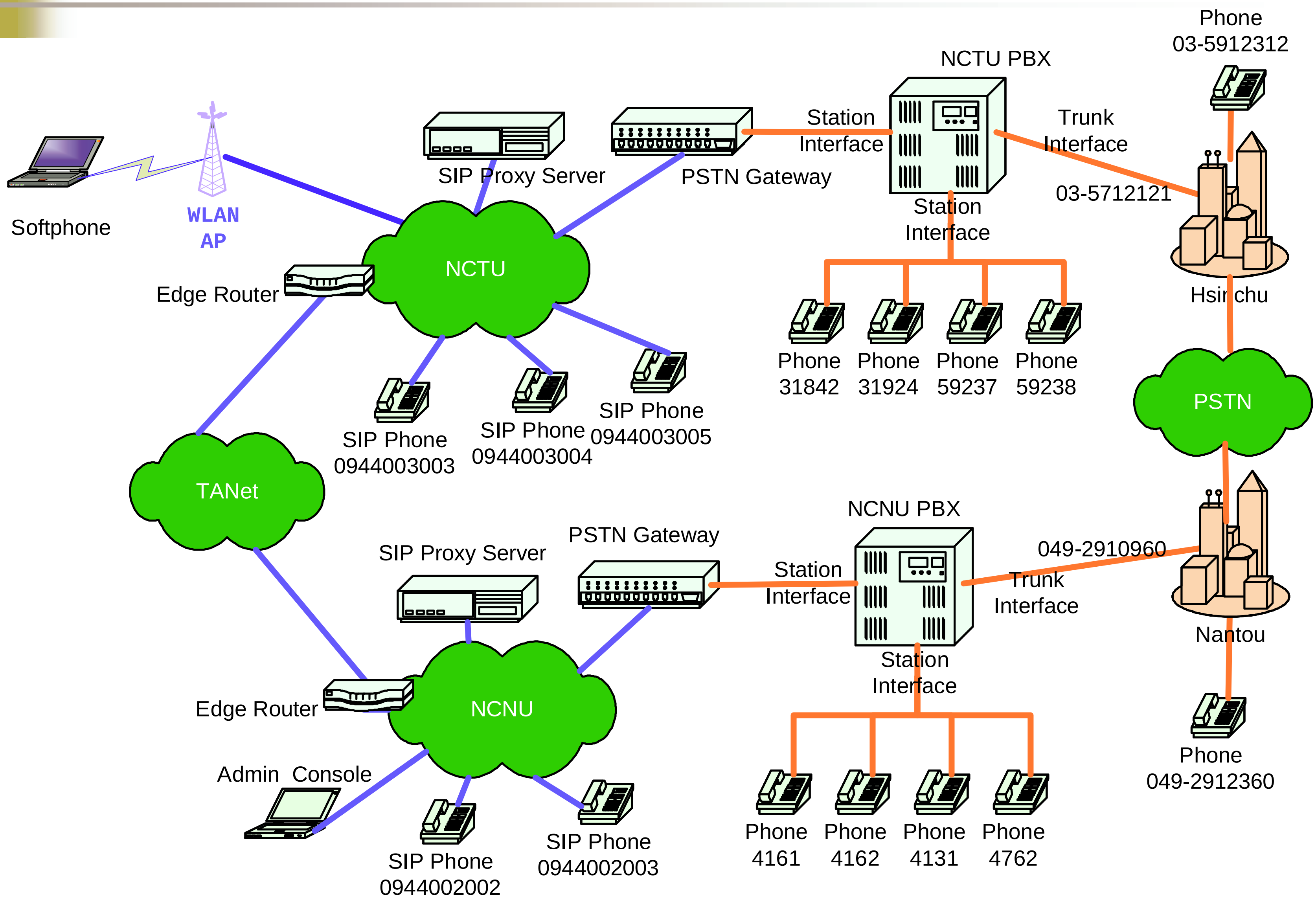
IP Telephony (Voice over IP)

- Instructor
  - Quincy Wu (吳坤熹), solomon@ipv6.club.tw
- Textbook
  - "Carrier Grade Voice over IP," D. Collins, McGraw-Hill, Second Edition, 2003.
- Requirements
  - Homework x 5 20%
  - Mid-term exam 20%
  - Oral presentation 40%
  - Final exam 20%
- Course Webpage
  - http://Course.ipv6.club.tw/VoIP/
- TAs
  - 蔡政霖
  - 陳韋霖

Introduction

Chapter 1

NTP VoIP Platform



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

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 != Voice over Internet
  - The next generation Telcos
    - Access and bandwidth are better managed.

IP

- A packet-based protocol
  - Routing on a packet-by-packet base
- Packet transfer with no guarantees
  - May not receive 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, integration of voice and data applications, potentially lower bandwidth requirements, the widespread availability of IP

Lower Equipment Cost

- PSTN switch
  - Proprietary – hardware, OS, applications
  - High operation and management cost
    - Training, support and feature development cost
- Mainframe computer
- The IP world
  - Standard hardware and mass-produced
  - Application software is quite separate
  - A horizontal business model
    - More open and competition-friendly
- IN (Intelligent Network)
  - does not match the openness and flexibility of IP.
  - A few highly successful services

Voice/Data Integration

- Click-to-talk application
  - Personal communication
  - E-commerce
- Web collaboration
  - Shop on-line with a fried at another location
- Video conferencing
- IP-based PBX
- IP-based call centers
- IP-based voice mail

The Widespread Availability of IP

- IP
  - LANs and WANs
  - Dial-up Internet access
  - The ubiquitous presence
- VoFR or VoATM
  - Only for the backbone of the carriers

VoIP Challenges

- VoIP must offer the same reliability and voice quality as PSTN.
- 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.

Lower Bandwidth Requirements

- PSTN
  - G.711 - 64 kbps
  - Human speech frequency < 4K Hz
  - The Nyquist Theorem: 8000 samples per second
  - 8K * 8 bits
- Sophisticated coders
  - 32kbps, 16kbps, 8kbps, 6.3kbps, 5.3kbps
  - GSM – 13kbps
  - Save more bandwidth by silence-detection
- Traditional telephony networks can use coders, too.
  - But it is more difficult.
- VoIP – two ends of the call negotiate the coding scheme

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
iLBC	13.3kbps	3.9

Speech Quality

- Must be as good as PSTN
- Delay
  - The one-way delay
  - Coding/Decoding + Buffering Time + Tx. Time
  - G.114 < 150 ms
- Jitter
  - Delay variation
  - Different routes or queuing times
  - Adjusting to the jitter is difficult
  - Jitter buffers add delay

Speech Quality

- 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

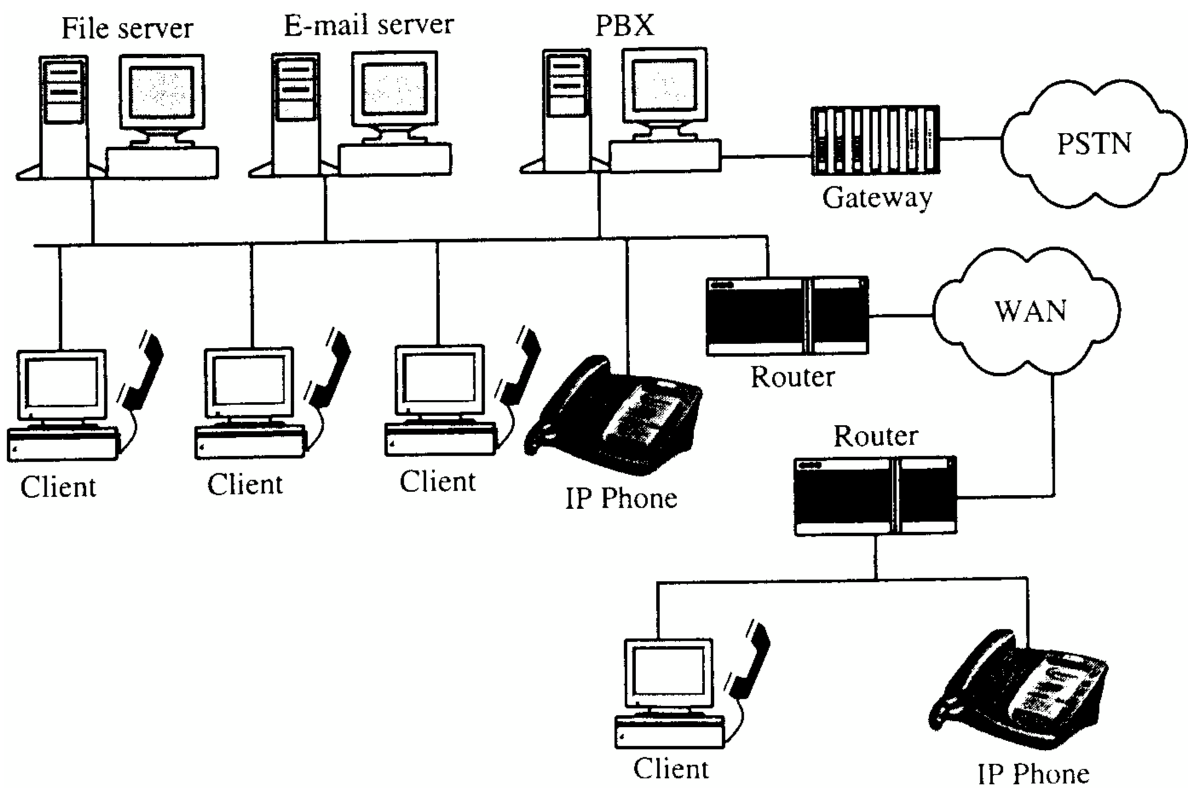
- A single network for a wide range of applications
- Call is admitted if sufficient resources are available
- Different types of traffic are handled in different ways
  - If a network becomes heavily loaded, e-mail traffic should feel the effects before synchronous traffic (such as voice).
- QoS has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

Network Reliability and Scalability

- PSTN system fails
  - 99.999% reliability
- Today's VoIP solutions
  - Redundancy and load sharing
  - Scalable – easy to start on a small scale and then expand as traffic demand increases

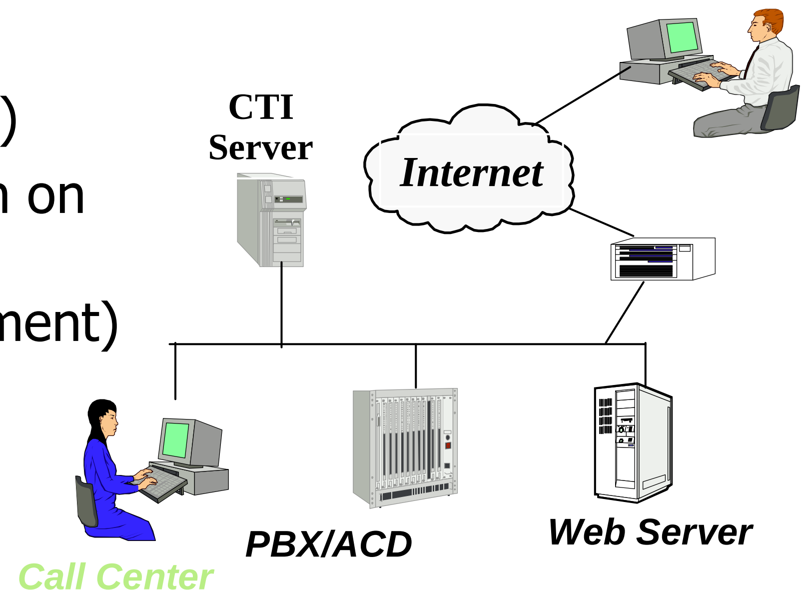
VoIP Implementations

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



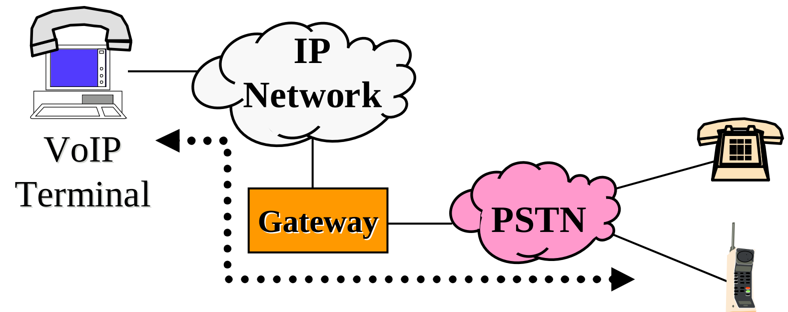
VoIP Implementations

- IP voice mail
  - One of the easiest applications
- IP call centers
  - Use the caller ID
  - Automatic call distribution (ACD)
  - Load the customer's information on the agent's desktop (CRM – Customer Relationship Management)
  - Click to talk

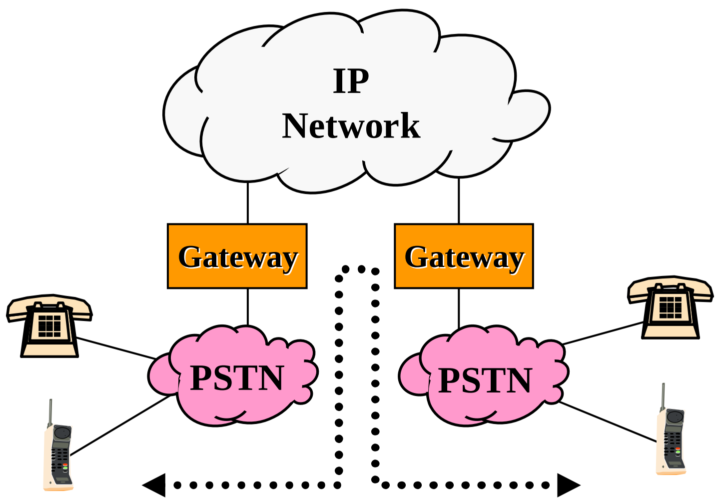


VoIP Evolution

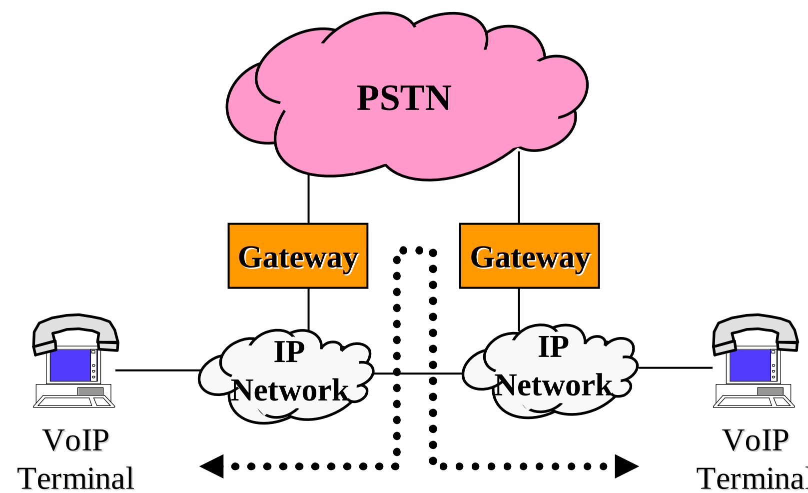
1: PC to PC (iPhone in 1995)



2: Phone to PC over IP



3: Phone to Phone over IP/PSTN



4: PC to PC over IP/PSTN

Overview of the Following Chapters [1/2]

- Chapter 2, "Transporting Voice by Using IP"
  - A review of IP networking in general to understand what IP offers, why it is a best-effort protocol, and why carrying real-time traffic over IP has significant challenges
  - RTP (Real-Time Transport Protocol)
- Chapter 3, "Voice-coding Techniques"
  - Choosing the right coding scheme for a particular network or application is not necessarily a simple matter.
- Chapter 4, "H.323"
  - H.323 has been the standard for VoIP for several years.
  - It is the most widely deployed VoIP technology.
- Chapter 5, "The Session Initiation Protocol"
  - The rising star of VoIP technology
  - The simplicity of SIP is one of the greatest advantages
  - Also extremely flexible (a range of advanced feature supported)

Overview of the Following Chapters [2/2]

- Chapter 6, "Media Gateway Control and the Softswitch Architecture"
  - Interworking with PSTN is a major concern in the deployment of VoIP networks
  - The use of gateways
  - They enables a widely distributed VoIP network architecture, whereby call control can be centralized.
- Chapter 7, "VoIP and SS7"
  - H.323, SIP, MGCP and MEGACO are all signaling systems.
  - The state of the art in PSTN signaling is SS7.
  - Numerous services are provided by SS7.
- Chapter 8, "QoS"
  - A VoIP network must face to meet the stringent performance requirements that define a carrier-grade network.
- Chapter 9, "Designing a Voice over IP Network"
  - How to build redundancy and diversity into a VoIP network without losing sight of the trade-off between network quality and network cost (network dimensioning, traffic engineering and traffic routing)?

Course Outline

- Introduction
- Real-time Transport Protocol (RTP)
- The Session Initiation Protocol (SIP)
- Network Address Translation (NAT) Issues and Countermeasure
- (optional) Deploying a Voice over IP Network
- (optional) VoIP Service