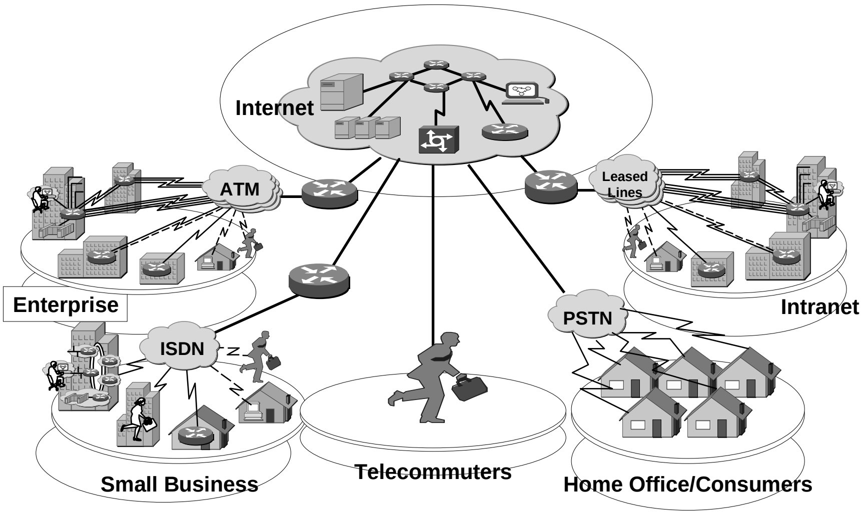
I.3 Introduction to Current Network Infrastructure

Prof. Chia-Wen Lin
Department of CS
National Chung Cheng University
886-5-272-0411 ext. 33120
cwlin@cs.ccu.edu.tw

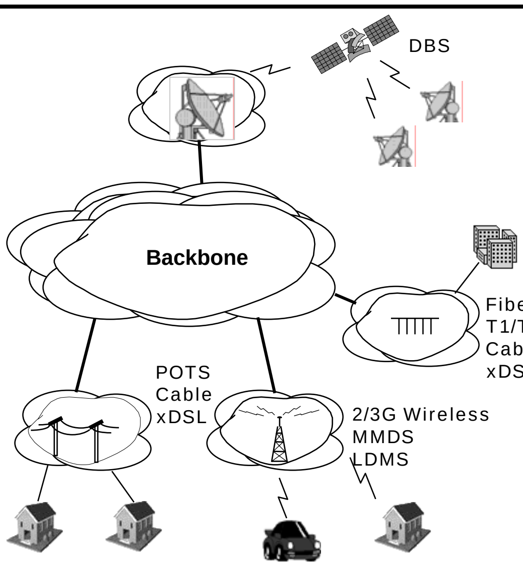
Outline

- Introduction
- IP Networks
- Wireless Systems and Networks
- Summary

Network Infrastructure

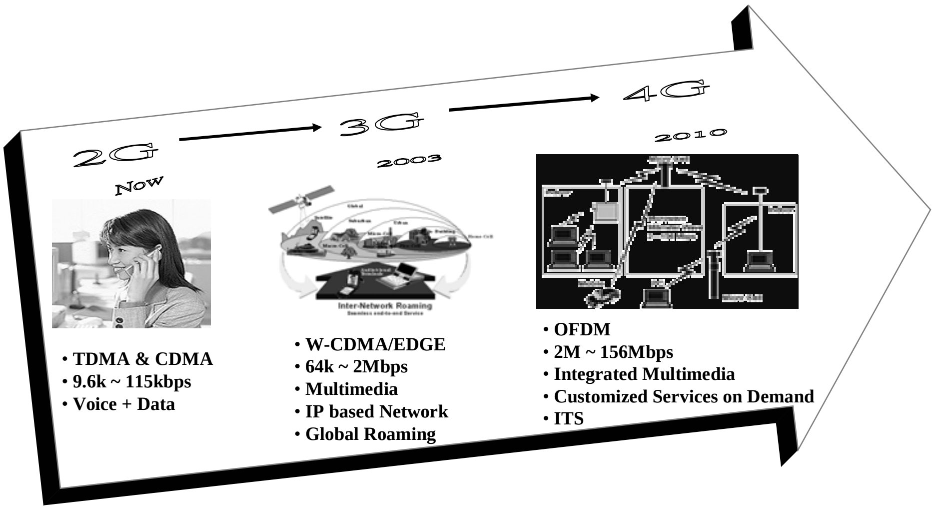


Networking



- Broadband Backbone
  - Multi-Gb/s Backbone
- Narrowband Access
  - POTS
- Broadband Access
  - Cable Modem
  - xDSL
- Wireless Access
  - DBS, LEOS
  - MMDS, LMDS
  - 3G IMT2000

Wireless Network Evolution



Network Characteristics

- PSTN: up to 33.6 kbits/s, ubiquitous, low cost
- N-ISDN: 128 kbits/s, widely available, low cost
- ATM (B-ISDN): broadband cell-switched network, guaranteed QoS, variable bit-rate, priority, not widely available
- Ethernet: packet-switched network, non-guaranteed QoS, delay, delay variation, packet loss, congestion, widely available, low cost
- IsoEthernet: guaranteed QoS, not widely available, higher cost
- Mobile: low-bit-rate, fading, bit errors
- xDSL, cable, satellite, etc.

IP Networks

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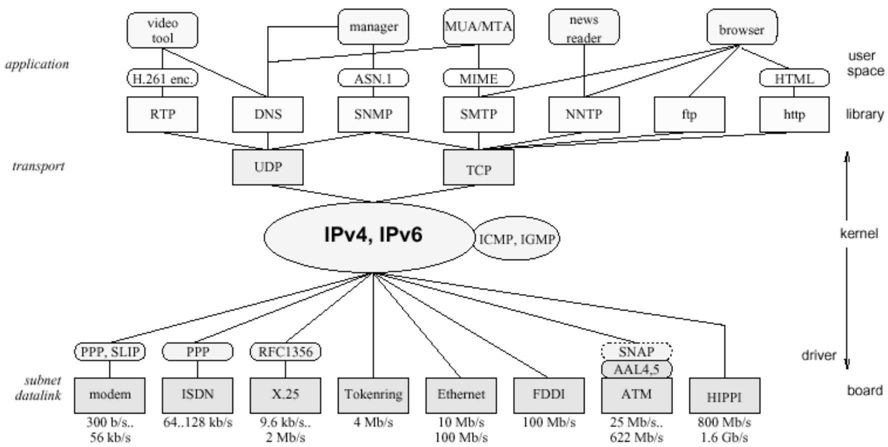
The Internet Protocol Hierarchy



Internet Concepts and Protocols

- Layers:
  - Physical Layer
    - provides point-to-point, point-to multipoint bit transport service over wire, optical fiber, and free space
    - sometimes includes forward error correction
  - Link Layer
    - provides a point-to-point, point-to-multipoint packet service for a relatively small number of nodes
    - may offer the detection of bit errors and the retransmission of lost or errored packets
    - classified into point-to-point (e.g., wide-area links), broadcast (e.g., Ethernet, FDDI, Token-ring, and IEEE 1394) , and non-broadcast multiple-access networks (NBMA; e.g., ATM and frame relay)

Internet Concepts and Protocols

- Network Layer
  - carries packets end-to-end across multiple subnets
  - The path of packets is determined by routing protocols
  - IP (IPv4 and IPv6) is the only routing protocol in the Internet
- Transport Layer
  - operates only within the communication end points (*end systems* or *host*s)
  - The Internet architecture is built primarily on two transport protocols
    - User Datagram Protocol (UDP): unreliable datagram service
    - Transmission Control Protocol (TCP): reliable, sequenced byte stream service
- Application Layer
  - Email, HTTP for WWW, ftp, Telnet, etc.

Internet Protocol (IP)

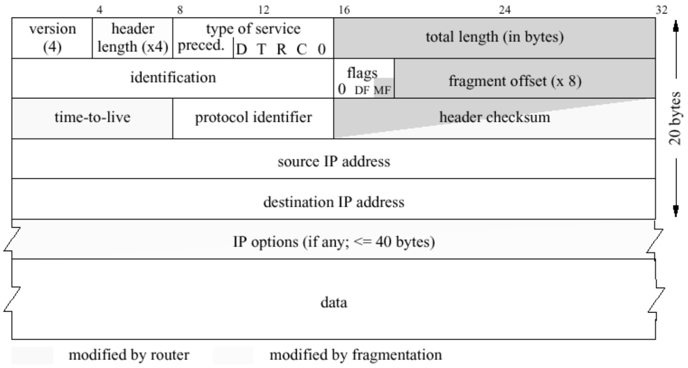
- IP address
  - used for routing and identifying nodes
  - IPv4: 32bits long, e.g. 140.123.102.204
  - IPv6: 128bits long, e.g. 1080::::8:800:200C:417A
  - IP address are divided into a network and a host parts

class	network	host	first octet	hosts per network	nets	delegated (Oct. 1999)
Class A	8	24	< 128	$16 \cdot 10^6$	128	37.5%
Class B	16	16	$128 \ldots 191$	65534	16384	70.3%
Class C	24	8	$192 \ldots 223$	254	$2 \cdot 10^6$	65.6%
Class D			$224 \ldots 239$		$268 \cdot 10^6$	multicast
Class E			$240 \ldots 255$		$134 \cdot 10^6$	reserved

Table 3.1: Internet address classes

Internet Protocol (IP)

- IPv4 packet header
  - At least 20 bytes (the complete packet is up to 65535 bytes long)
  - most link-layer protocols restrict the maximum packet length to between 1,500 and 8,192 bytes (most common 1,500 bytes)
  - Longer IP packets can be *fragmented* into the maximum transmission unit (MTU)



Transport-Layer Protocols: UDP & TCP

- UDP
  - unreliable, connectionless message
- TCP
  - reliable,connection-oriented stream of bytes
  - TCP is less suited for transmitting multimedia data than UDP, if there is an end-to-end delay limit
- Both support multiplexing
  - allow several distinct streams of data between two hosts, streams are labeled by source and destination *port number*s

Comparison of UDP and TCP

Characteristic	UDP	TCP	UDP	TCP
	without resource reservation		reserved resources	
Packet loss	yes	no	no	no
Delay bound	no	no	possible	possible
Abstraction	packet	byte stream	packet	byte stream
Ordering	none	always in order	none	in order
Duplication	possible	no	possible	no
Multicast	yes	no	yes	no

UDP Packet Format

0 16 32

16-bit source port number	16-bit destination port number
16-bit UDP length (incl. header)	16-bit UDP checksum
data (if any)	

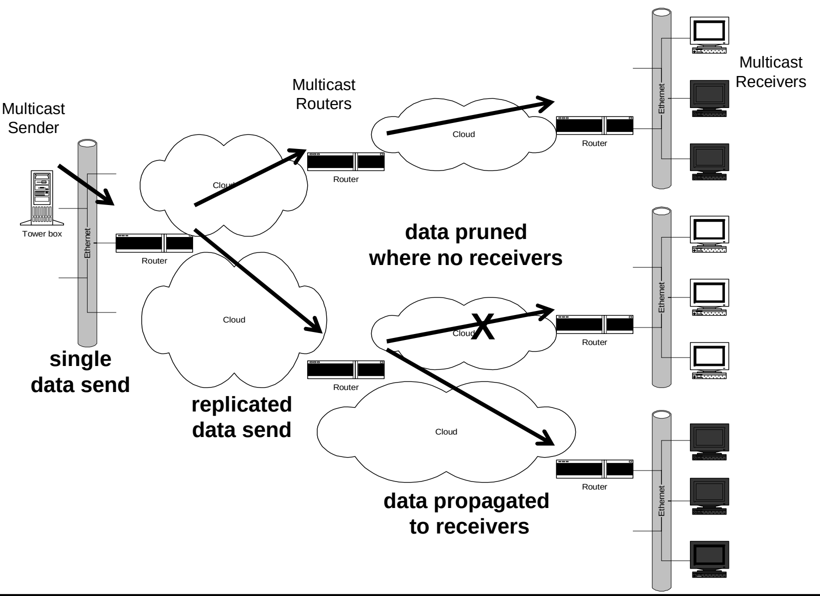
IP Multicast

- allows sender to transmit a single IP packet to multiple receivers
- Three ways:
  - receivers setting up virtual circuits to senders (x)
  - senders including a list of addresses in the packet header (x)
  - radio-like model of host-group multicast (o)

IP Multicast

- Host group model
  - IP address Class-D: 224.0.0.0 to 239.255.255.255
  - IP multicast group can have any number of senders and receivers
  - does not depend on the transport protocol, but TCP can clearly not be used
  - the distribution of multicast packets can be limited by
    - setting time-to-live (TTL) value
    - using scoped multicast addresses (to a single organization or a provider's network)
  - Protocols for discovering the group members:
    - Local-area: Internet Group Management Protocol (IGMP)
    - Routing: MOSPF/DVMRP
    - Interdomain routing: Protocol Independent Multicast (PIM), Core Base Tree (CBT)

An IP Multicast Example



Internet Quality of Service (QoS)

Two major QoS impairments: delay and packet loss

- End-to-End delay

  the time elapsed between sending and receiving a packet or a particular byte

  - Propagation delay- depend on physical distance of path
  - Transmission delay- the sum of the time it takes the network interfaces to send out the packet

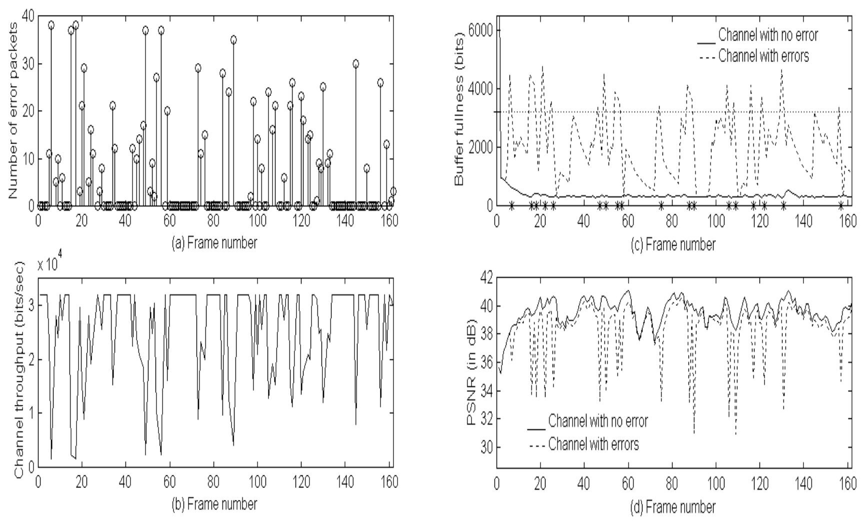
Internet QoS (Cont.)

- Causes of delay
  - resource contention (in routers or end system) and link-layer retransmission cause variable delays
  - packets are dropped rather than delayed if network overload becomes severe
  - bit interleaving and media access resolution
  - play-out buffer for smoothing delay jitter
  - the application & media coding
  - additional end-system delays occur when the receiver has to wait for later packets to reconstruct packet loss

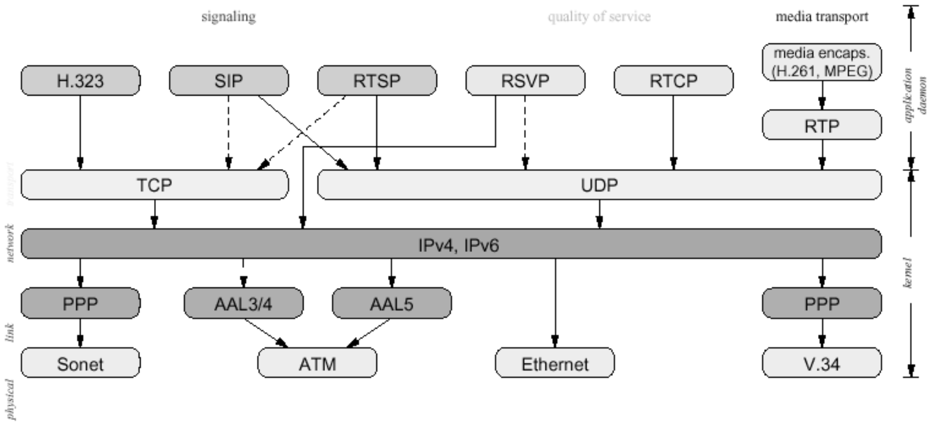
Internet QoS (Cont.)

- Packet loss
  - packets never arrive or arrive too late
  - continuous-media applications are sensitive not only to the packet loss probability, but also to the correlation of packet losses
- Packet Reordering
  - Caused by frequent routes changes
  - can be solved by play-out buffer
- Packet Duplication
  - caused by faulty hardware or drivers, transition in spanning trees, and other anomalies
- Connection Refusal
  - for networks with resource reservation, reservations may be refused by the call admission control (CAC) mechanism if sufficient bandwidth is not available
- Trade-off?

Impact of Burst Packet Loss on Video Quality



Internet Protocol Architecture for Continuous Media Services



QOS-related Protocols

- Real-Time Control Protocol (RTCP) may be used to monitor end-to-end quality-of-service for individual continuous-media streams
- Multicast has specific extensions to IGMP for reporting on connectivity, packet rates and loss
- “Differentiated services” approach has attempted to provide differing levels of service to a relatively small number of packet classes (defined by type-of-service byte in the IP header).

Signaling Protocols

- Different types of applications use different signaling protocols
- Protocol
  - Real-Time Stream Protocol (RTSP) for media-on-demand
  - SIP and H.323 for Internet telephony
  - SAP for broadcast applications
  - The above protocols may be combined into novel applications
  - the Session Description Protocol (SDP) is most commonly used to describe the streams making up the multimedia sessions

Real-time Transport Protocol (RTP)

- Common requirements of real-time multimedia flows:
  - Sequencing
  - Intramedia synchronization
  - Intermedia synchronization
  - Payload identification
  - Frame indication
- RTP has two components: RTP and RTCP

Real-time Transport Protocol (RTP)

- RTP provides some functionality beyond re-sequencing and loss detection
  - Multicast friendly
  - Media independent
  - Mixers and translators
  - QOS feedback
  - Loose session control
  - Encryption
- RTP is used in conjunction with the UDP, but can make use of any packet-based lower-layer protocol

Real-time Transport Protocol (RTP)

RTP

- V field indicates the protocol version
- X flag signals the presence of a header extension
- P bit indicates that the payload is padded to ensure proper alignment for encryption
- SSRC distinguishes user in multicast group
- CSRC lists all the SSRC that “contributed” content to the packet

Real-time Control Protocol (RTCP)

- real-time control protocol (RTCP)

  Media senders (sources) and receivers (sinks) “periodically” send RTCP packets to the same multicast group (but different ports) used to distribute RTP packets

Sender report (SR):

  the amount of data sent so far
  correlating the RTP sampling timestamp and absolute
  time to allow synchronization between different media

Receiver report (RR):
  contains one block for each RTP source in the group.
  Each block describes the instantaneous and cumulative loss rate
  and jitter from that source.

Source descriptor (SDES):
  used for session control.

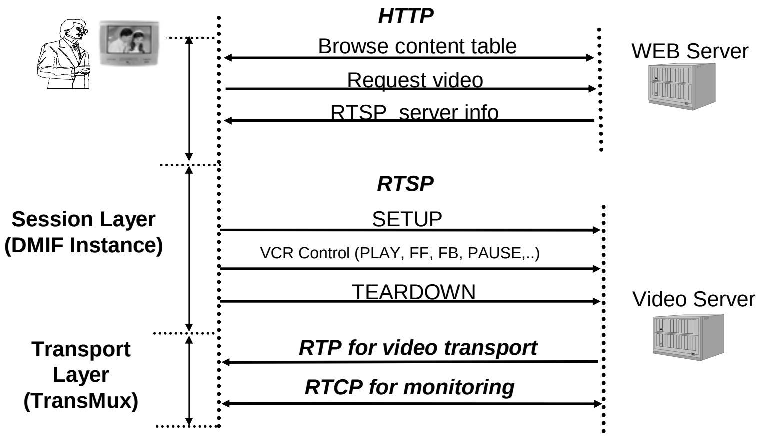
Real-Time Streaming Protocol (RTSP)

- allows a client to open a media session consisting of one or more streams, where the streams may be located on a single server or multiple servers
- can be used to tell a sever to record packets from a particular network address
- Syntax: similar to HTTP, textual protocol
- Request: request URL, protocol version, a number of parameter-value header lines and then the message body
- Document: IETF RFC 2326 (http://www.ietf.org)

RTSP Methods

OPTIONS	$S \leftrightarrow C$	determine capabilities of server or client
DESCRIBE	$C \rightarrow S$	get description of media stream
ANNOUNCE	$S \leftrightarrow C$	announce new session description
SETUP	$C \rightarrow S$	create media session
RECORD	$C \rightarrow S$	start media recording
PLAY	$C \rightarrow S$	start media delivery
PAUSE	$C \rightarrow S$	pause media delivery
REDIRECT	$S \rightarrow C$	use another server, please
TEARDOWN	$C \rightarrow S$	destroy media session
SET_PARAMETER	$S \leftrightarrow C$	set server or client parameter
GET_PARAMETER	$S \leftrightarrow C$	read server or client parameter

Video Streaming Protocols: An Example



Session Description Protocol (SDP)

- a text format for describing multimedia sessions
- not really a protocol, but rather similar in spirit to a mark-up language like HTML
- Convey the following information
  - the type of media (video, audio, shared applications)
  - the media transport protocol (typically, RTP/UDP/IP, but could be an ATM virtual circuit)
  - the format of the media (such as H.261 video, MPEG video, G.723.1 audio), with a mapping between encoding names and RTP payload types
  - the time and duration of the session
  - security information, such as the encryption key for a media session;
  - session names and subject information for presentation in “TV Guide”-style session directories
  - human contact information related to a session
- SDP message consists of a set of global headers describing the session, followed by a set of media descriptions

SDP: An Example

```
o=- 2890844526 2890842807 IN IP4 192.16.24.202
s=Twister PG-13 (c) Warner Bros.
m=audio 0 RTP/AVP 98
a=rtpmap:98 L16/16000/2
a=control:rtsp://audio.example.com/twister/audio.en
m=video 0 RTP/AVP 31
a=rtpmap:31 MPV
a=control:rtsp://video.example.com/twister/video
```

IP Telephony Signaling Protocol

- Functionality of an IP telephony signaling protocol
  - Name translation and user location
  - Feature negotiation
  - Call participant management
  - Feature changes
- Two IP signaling protocols
  - IETF: SIP
  - ITU-T: H.323

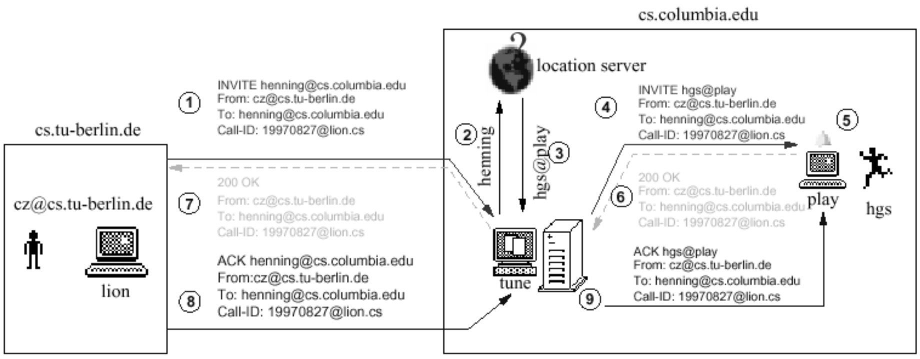
Session Initiation Protocol (SIP)

- SIP is a client-server protocol
- SIP-enabled end systems include a protocol client and server (generally called a user agent server), the user agent server generally responds to the requests based on human interaction or some other kind of input
- SIP requests can traverse many *proxy server*s, each of which receives a request and forwards it towards a next hop server, which may be another proxy server or the final user agent server
- A server may also act as a *redirect serve*r, informing the client of the address of the next hop server, so that client can contact it directly

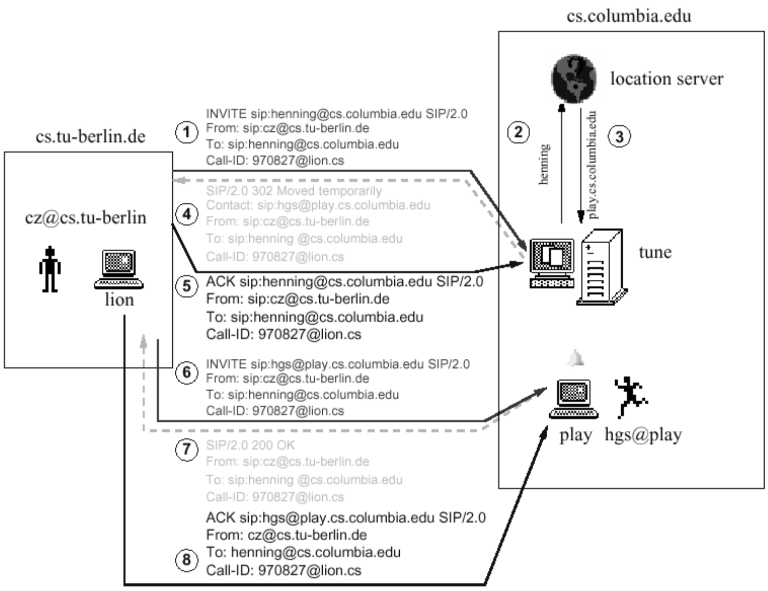
Session Initiation Protocol (SIP)

- Information conveyed in SIP header
  - Call identifier
  - Logical connection source
  - Logical connection destination
  - Media destination
  - Media capability
- Methods in SIP
  - INVITE
  - BYE
  - OPTIONS
  - STATUS
  - ACK
  - CANCEL
  - REGISTER

SIP Proxy Server Operation



SIP Redirect Sever Operation



Summary

- Internet multimedia services can be built without changing the basic IP infrastructure
- Major challenges
  - Protocol foundation
    - there is no session control protocol that can be used to perform floor control in distributed multimedia conferences
    - Protocols for sharing computer applications are limited, mostly proprietary, and not well suited for Internet use
  - Operational infrastructure
    - Network reliability and deployment multicast of services with predictable QOS

Wireless Systems and Networking

Prof. Chia-Wen Lin
Department of CS
National Chung Cheng University
886-5-272-0411 ext. 33120
cwlin@cs.ccu.edu.tw

Outline

- Introduction
- Overview of Wireless System Characteristics
- Wireless Networks
- Video Applications over Wireless Networks
- Summary

Introduction

- 2G wireless systems open up market not only for voice services but also for data services
- To offer a cost-effective multimedia service, the air interface needs to be able to support data rates higher than 2G rate
- With the advent of the technology advancement, spectrum availability, and innovative algorithms developed, a wireless multimedia service will be available in the near future

Propagation Characteristics

The signals transmitted over the radio channel are subject two types of signal variations :

•Large-scale signal variation:

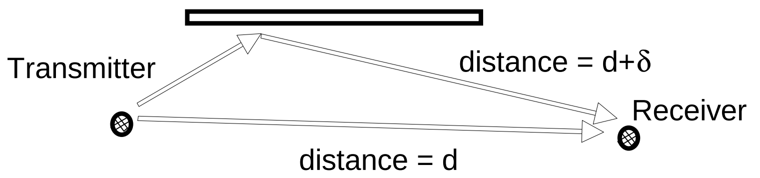
- path attenuation

•Small-scale signal variation:

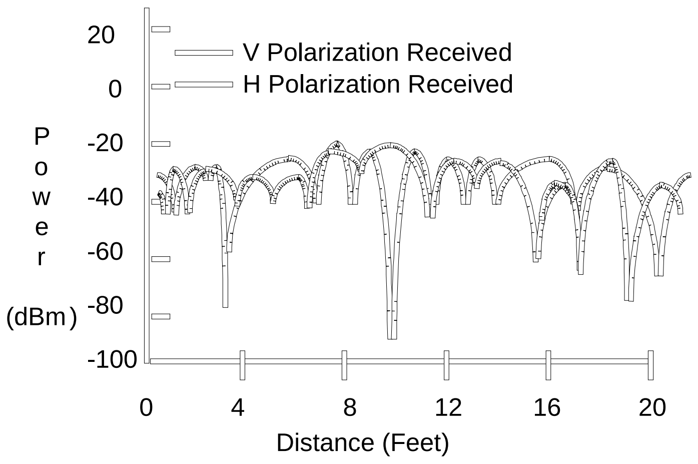
- multi-path signal propagation

Propagation Characteristics

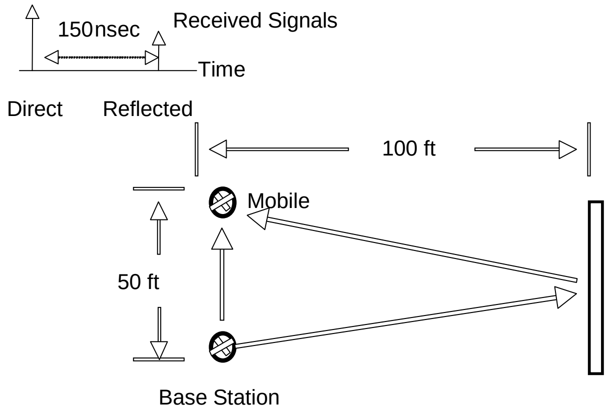
- Multi-path signal propagation :
- Between each transmitter/receiver pair, many scattered or reflected signal exist due to signal reflections from walls,ceilings or other objects, resulting in multi-path signal propagation.



Received Signal Power from Two Antennas



Multi-path Delay Spreading



Antenna Diversity

- Use two orthogonal antenna and chose the best received signal
- Improve the overall received signal quality substantially
- Mitigate the effect of multi-path fading

Multiple Access & Duplexing Schemes

- Duplex techniques
  - Time Division Duplex (TDD)
  - Frequency Division Duplex (FDD)
- Multiple Access Schemes
  - Frequency Division Multiple Access (FDMA)
  - Time Division Multiple Access (TDMA)
  - Code Division Multiple Access (CDMA)

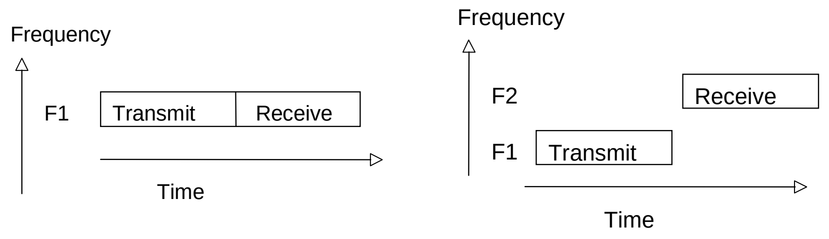
Time Division Duplex (TDD)

- TDD requires only one frequency band but the transmission rate in each direction is only half of the radio channel rate
- In systems using TDD multiplex the uplink and the downlink information are transmitted together in the same radio frequency channel but at different time
- Time synchronization among base stations is required to maintain good system performance and frequency reuse efficiency

Frequency Division Duplex (FDD)

- Use a pair of frequencies for uplink and downlink transmission separately
- Avoid the need for base station synchronization
- Most of the 2nd generation digital wireless systems employed FDD to avoid strong inter-base station interference

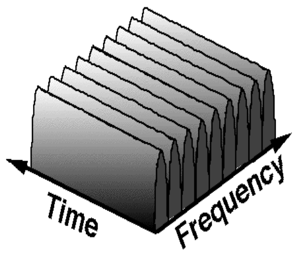
TDD vs. FDD



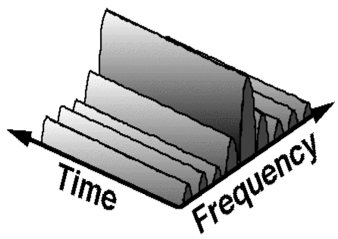
Frequency Division Multiple Access (FDMA)

- Each user is assigned to an non-overlapping and exclusive frequency segment
- At the receiver, a band-pass filter is used to filter out signal carried in each frequency channel, then the individual user signal will be recovered

FDMA Before and After Filtering



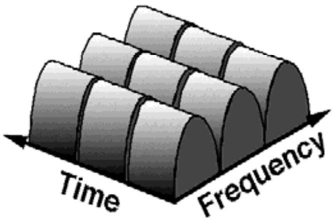
After Filtering



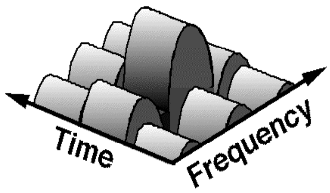
Time Division Multiple Access (TDMA)

- User communication with the same base station transmit data on the same carrier frequency but at different times
- The transmission time is segmented into frames and each frame is divided into time slots
- A user with a transmission rate equivalent to the time slot duration uses that time slot to communicate
- Require frame/time slot synchronization

TDMA Before and After Filtering



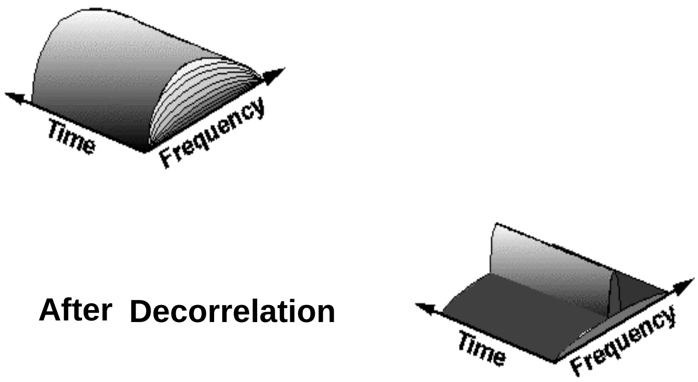
After Filtering



Code Division Multiple Access (CDMA)

- A unique binary spreading Pseudo Random code is assigned to each user and all active users share the same frequency spectrum at the same time
- The signal of each user is separated from the others at the receiver by using a selected binary spreading code
- CDMA is more robust to the multi-path delay spread environment than the other access schemes

CDMA Before and After De-Spreading



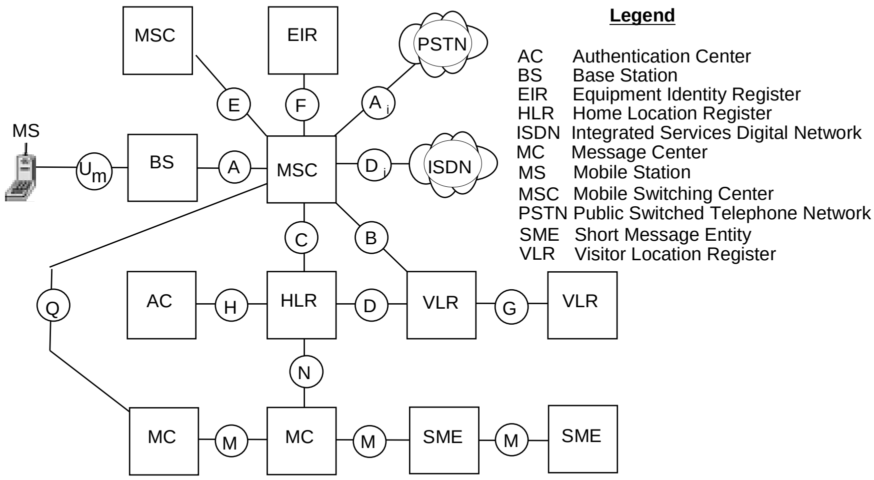
1G & 2G Wireless Systems

- 1G wireless systems
  - introduced in the 1980s
  - designed for local or national coverage and primarily for voice services
  - use analog frequency modulation
  - demand has continuous exceeded expectation
  - several different systems: AMPS in North America, TACS in UK, NMT in Nordic countries

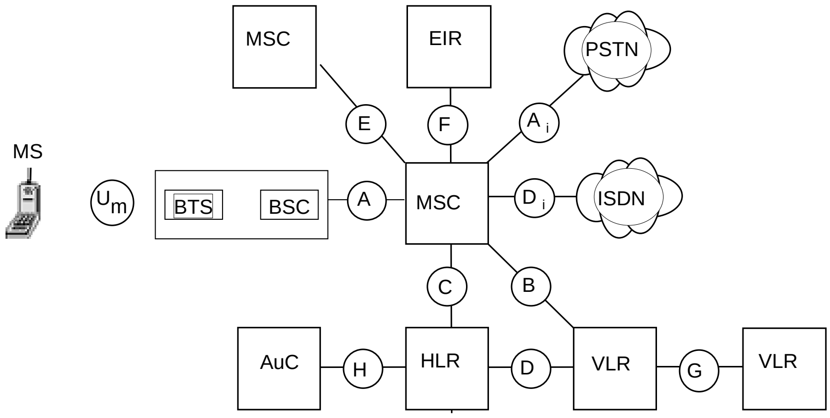
1G & 2G Wireless Systems

- 2G wireless systems
  - The "all digital" 2G wireless systems were designed to solve the capacity problem in 1st generation and were deployed in early 1990s
  - To meet the demand for wireless access and to enlarge the range of applications
  - They can be classified into high-tier and low-tier systems

ANSI41 Network Reference Model



GSM900/DCS1800 Network Reference Model



3G Wireless Systems

- Provides high-speed packet data services and to offer higher capacity for the voice services
- Offers smoother interconnecting between different networks
- Offers global roaming to the users
- Provides variable rate services for different propagation environment

Minimum Bearer Capability in Different Environments

Test environment	**Indoor**	**Outdoor-to-Indoor and Pedestrian**	**Vehicular**
Speech	32 kbits/s BER $\leq 10^{-3}$	32 kbits/s BER $\leq 10^{-3}$	32 kbits/s BER $\leq 10^{-3}$
Circuit-switched data	2 Mbits/s BER $\leq 10^{-6}$	384 kbits/s BER $\leq 10^{-6}$	144 kbits/s BER $\leq 10^{-6}$
Packet-switched data	2 Mbits/s BER $\leq 10^{-6}$, exponentially-sized packets, Poisson arrivals	384 kbits/s BER $\leq 10^{-6}$, exponentially-sized packets, Poisson arrivals	144 kbits/s BER $\leq 10^{-6}$, exponentially-sized packets, Poisson arrivals

Emerging Air Interface Approaches for Packet Data Services

- Enhanced Data rates for GSM Evolution (EDGE)
- CDMA2000
- Wideband CDMA (WCDMA)

EDGE

- EDGE is designed to operate on the GSM and GPRS's infrastructure to provide higher data rates services
- Dynamic multi-time slots allocation and link adaptation are used in EDGE to provide data rates between 64 to 384 kbps
- The modulation scheme has been changed to 8-PSK due to the better performance at the higher data rate

EDGE vs. GSM: Physical Layer Designs

Attributes	GSM	EDGE
Frame Structure	TDMA, 8 time slots/frame, 4.62 ms frame duration	Same as GSM
Modulation (downlink)	GMSK, 1 bit/symbol	8-PSK, 3 bits/symbol
Modulation (uplink)	GMSK	8-PSK and GMSK
Payload/burst	114 bits	346 bits
Gross rate/time slot	22.8 kbps	69.2 kbps
Channel spacing	200 kHz	200 kHz
Time slot assignment	Fixed, single	Dynamic, up to 8 time slots
Link adaptation	No	Yes (8 classes)
Handover	MAHO (Mobile Assisted Hand Over)	MAHO for transparent mode. Cell re-selection for packet data operation
Channel coding	Rate ½ convolutional code with memory 5	Rate 1/3 convolutional code with memory 6 and different puncturing paterns

WCDMA

- a wideband, spread spectrum radio interface that uses CDMA technology
- variable rate transmissions are possible by changing the spreading factor
- introduces a Synchronization Channel (SCH) in the downlink for MS to perform cell search

WCDMA Features & Characteristics

Frame duration	10 ms
Number of time slots	16 (time slot duration= power control period)
Duplexing	FDD (outdoor), TDD (indoor)
Modulation	QPSK (downlink), BPSK (uplink)
Channel bit rate (FDD)	$32*2^k$ kbps (downlink), $16*2^k$ kbps (uplink), k=1,10
Channel bit rate (TDD)	512 kbps
Spreading factor	variable ranging from 4-256
Spreading bandwidth	5 Mhz (10 Mhz, 20 Mhz available for FDD)
Chip rate	4.096 Mchps (8.192 Mcps, 16.384 Mcps available for FDD)
Power-control	1.6 kHz for FDD, 100 Hz for TDD
Power control step size	0.25-1.5 dB for FDD, 2 dB for TDD
Inter-BS synchronization	FDD: not required, TDD: required
Handover	MAHO (Mobile Assisted Hand Off)
Multi-rate/variable rate scheme	variable spreading factor+multi-code

IS-95 vs. CDMA-2000

	TIA/EIA-95 B	CDMA2000
Frame duration	20 ms	20 ms or 5 ms
Duplexing	FDD	FDD, TDD
Uplink modulation	64 orthogonal (Walsh)	Pilot Coherent
Channel bit rates	1.2 kbps to 19.2 kbps	1.2 kbps to 2 Mbps
Spreading bandwidth	1.25 MHz	N*1.25 MHz, N=1,3,6,9,12
Downlink power control	Slow open loop	800 Hz close loop
High data rate support	Multiple codes	Variable spread
Transmitter diversity	No	Yes
Low latency 5ms control frame	No	Yes

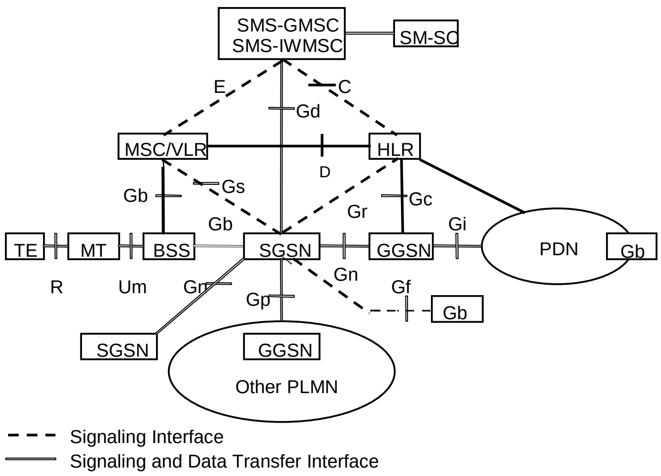
General Packet Radio Service (GPRS) Network

- GPRS is a new packet data service introduced in the GSM phase 2 standard
- The system consists of the packet wireless access network and IP-based backbone and is designed to provide access to packet data networks
- The basic GPRS network offers a payload bit-rate ranging from 9 to 21.4 kbps (single time slot), while the enhanced GPRS (i.e., EDGE) will provide bit-rates ranging from 8.8 to around 59.2 kbps.
- designed to support QoS negotiation for different service classes

GPRS Network

- Three MS classes are defined in GPRS to serve different needs of various market segments
  - Class A : can support GSM circuit calls and GPRS packet services concurrently
  - Class B : support GSM circuit calls and GPRS packet services sequentially
  - Class C : only support GPRS services

GPRS Logical Architecture

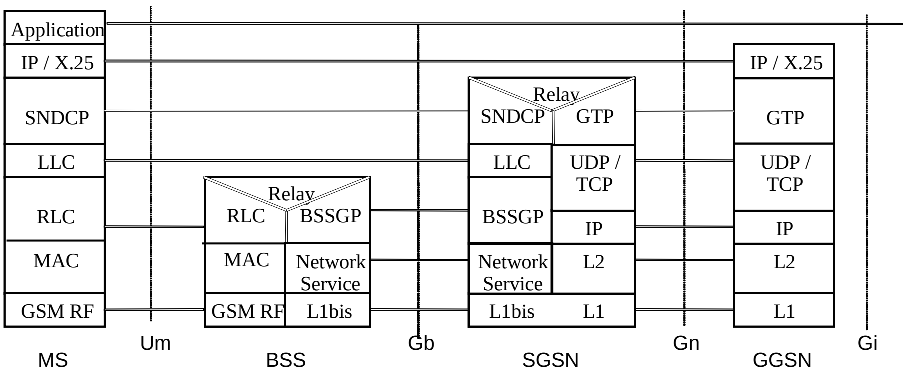


SMS-GMSC: Short Message Service-Gateway Mobile Switching Center
SM-SC: Short Message Service Center IWMSC: Inter-Working MSC
PLMN: Public Land Mobile Network
TE: Terminal Equipment
MT: Mobile Terminal

GPRS Delay Class: Maximum Values

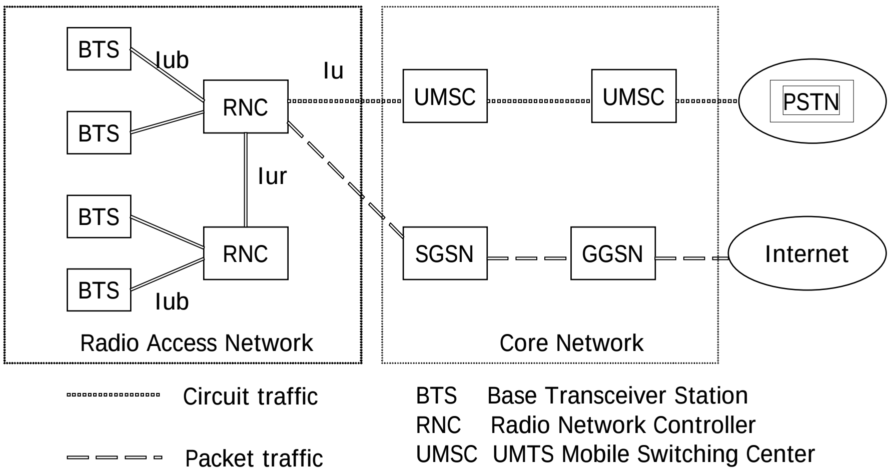
Delay Class	Delay (maximum values)			
	SDU size: 128 octets		SDU size: 1024 octets	
	Mean Transfer Delay (sec)	95 percentile Delay (sec)	Mean Transfer Delay (sec)	95 percentile Delay (sec)
1. (Predictive)	< 0.5	< 1.5	< 2	< 7
2. (Predictive)	< 5	< 25	< 15	< 75
3. (Predictive)	< 50	< 250	< 75	< 375
4. (Best Effort)	Unspecified			

GPRS Data Transmission Plane



GTP GPRS Tunnelling Protocol
SNDCP Sub-network Dependent Convergence Protocol
BSSGP Base Station System GPRS Protocol
LLC Logical Link Control
RLC Radio Link Control

UMTS Network Architecture



Video Applications over Wireless Networks

- Considerations in separate design of video coder:
  - Scalable video compressing rate to adaptive to the variations of the wireless channel quality
  - Robustness to the transmission errors
  - Minimum end-to-end delay to improve the visual quality
  - Capable to handle missing frames to avoid noticeable degradation of the video quality
- Joint optimization of source & channel coding

Summary

Although the wireless channel is a very harsh environment for real time data applications, the intelligent 3rd generation packet air interface and protocol designs and rate adaptation in the application will make interactive wireless multimedia services possible