Emerging Technologies for Multimedia Networks

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Outline

- Communications and Technologies
- Video Standards
- Multimedia Networks
- Quality of Services (QoS)
- Technical Challenges
- Researches

Communications and Technologies

- Wired:
  - Transmission Line Loss, Echo, Delay, Insertion Loss, Impedance Matching, Crosstalk, Return Loss, Clock Sync.
- Wireless:
  - Signal Bandwidth vs Noise/Interference
  - Antenna Gain
  - Congestion
  - Modulation and Multiplexing
- Multimedia Networks:
  - Video/Audio over RTP/UDP/IP, SCTP/IP, TCP/IP

Wired Digital Transmission Highlights

Digital Signal	Optical Transmit	Electrical Transmit	Line Bit Rate	Effective Data Rate	# DS0s in Payload	#DS1s in Payload	#DS3s in Payload	Others	SDH Level
DS-0		E0 /J0	64 Kbps	64 Kbps	1				
DS-1		T1 /J1	1.544 Mbps	1.536 Mbps	24	1			
		E1	2.048		32				
DS-2		T2	6.312		96	4			
		E2	8.448		128				
		E3	34.368		512				
DS-3		T3	44.736		672	28	1		
	OC-1	STS-1	51.84	50	672	28	1		
		E4	139.264		2048				
	OC-3	STS-3	155.52	150	2016	84	3		STM-1
DS-4			274.176		4032	168	6		
	OC-9	STS-9	466.56	451	6048	252	9		STM-3
	OC-12	STS-12	622.08	601	8064	336	12	4 OC-3	STM-4
	OC-24	STS-24	1.244 Gbps	1.20 Gbps	16128	672	24		STM-8
	OC-96	STS-96	4.976	4.81	64512	2688	96		STM-32
	OC-256		13.271		172032	7168	256		
	OC-768		39.813		516096				

Wireless: Modulation and Multiplexing

- Modulation

  Frequency-Modulated Signals, Amplitude-Modulated Signals

  Phase/Angle-Modulated Signals, Phase/Amplitude Modulated
  Pulse Duration Modulated, Pulse Code Modulation

- Multiplexing

  Frequency Division Multiplex (FDM)

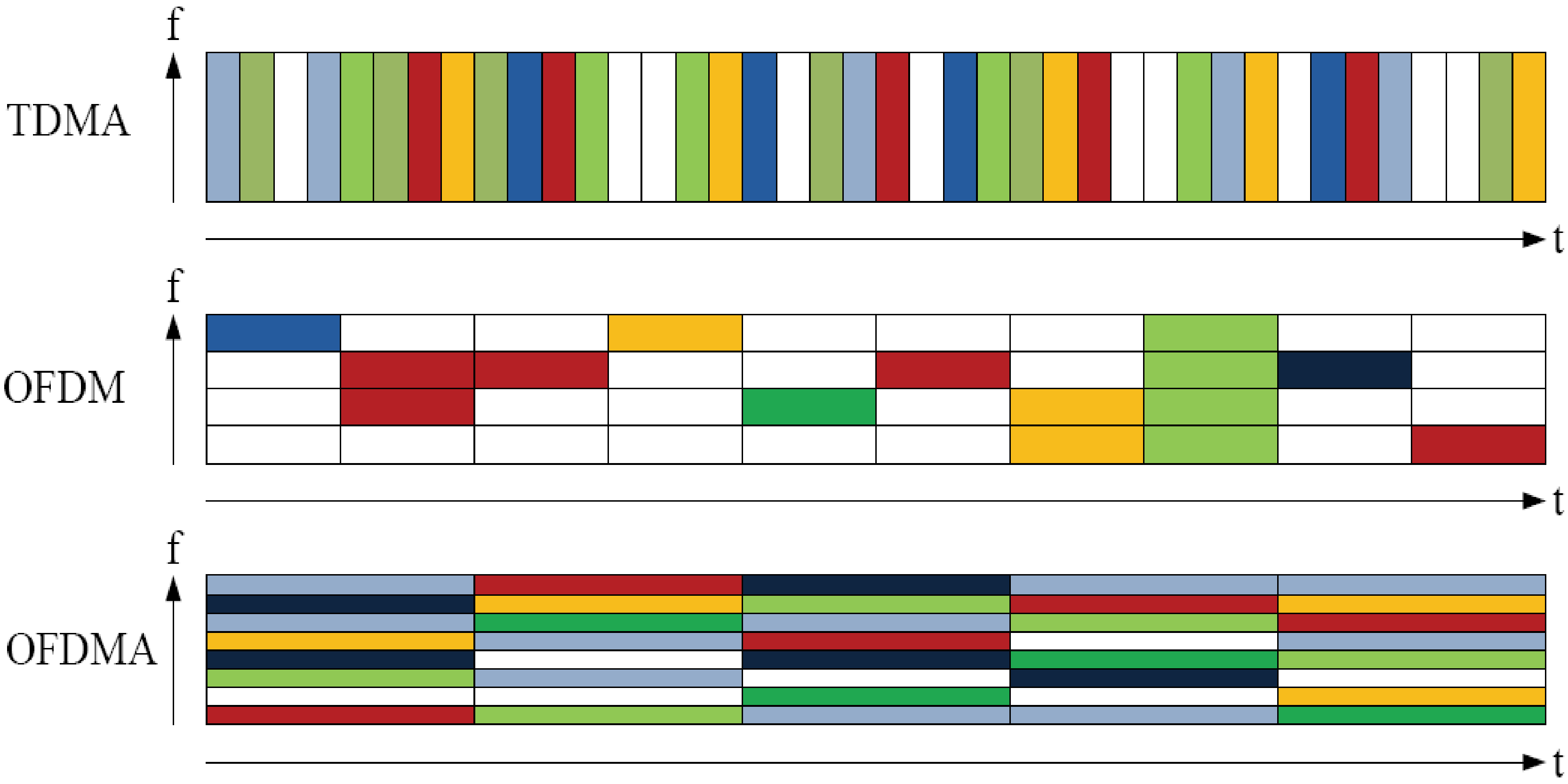
  Time-Division Multiplex (TDM)

  Space-Division Multiplex (SDM)

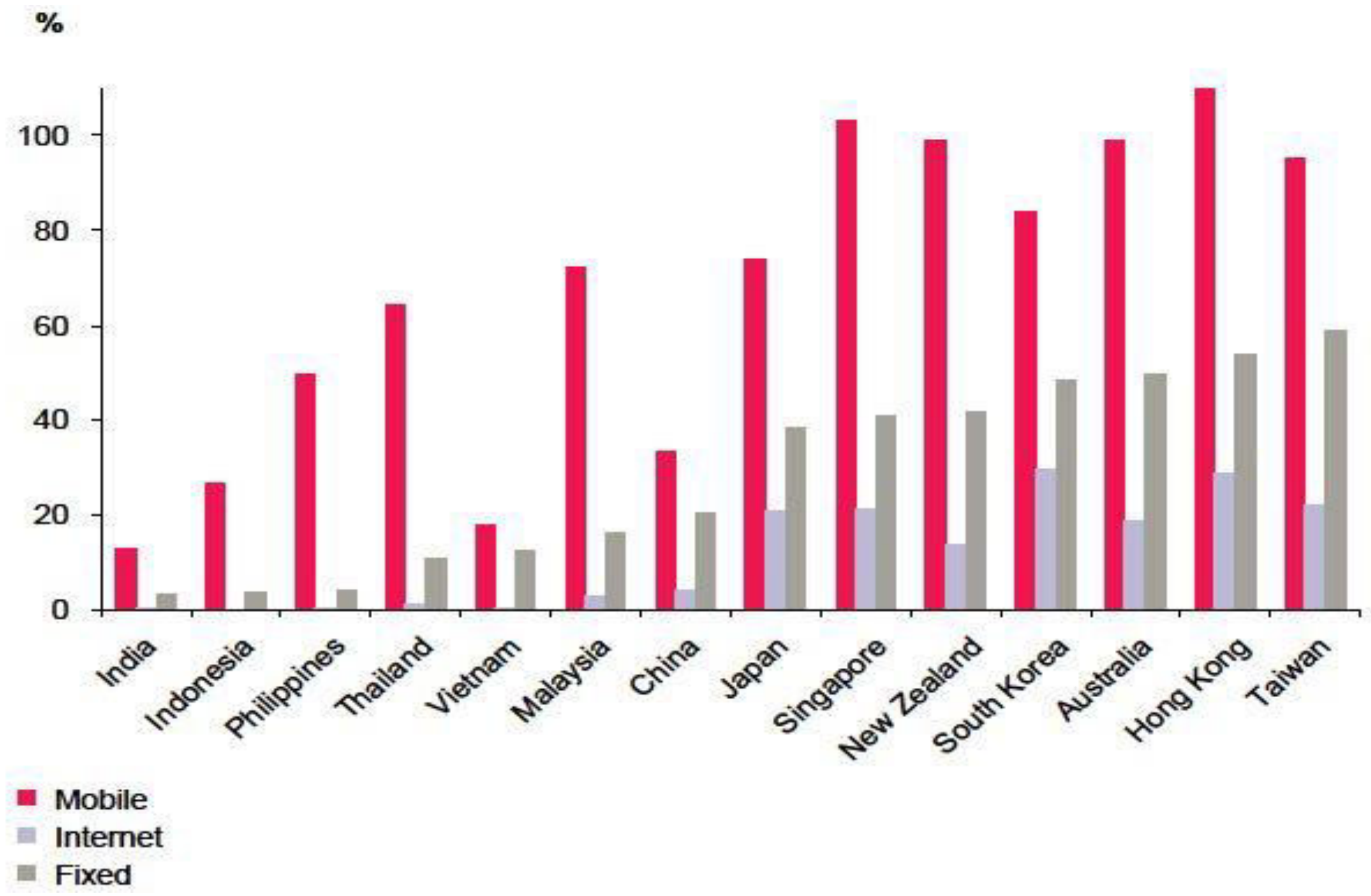
  Wave-Division Multiplex (WDM)

  Orthogonal-Frequency Division Multiplex (OFDM)

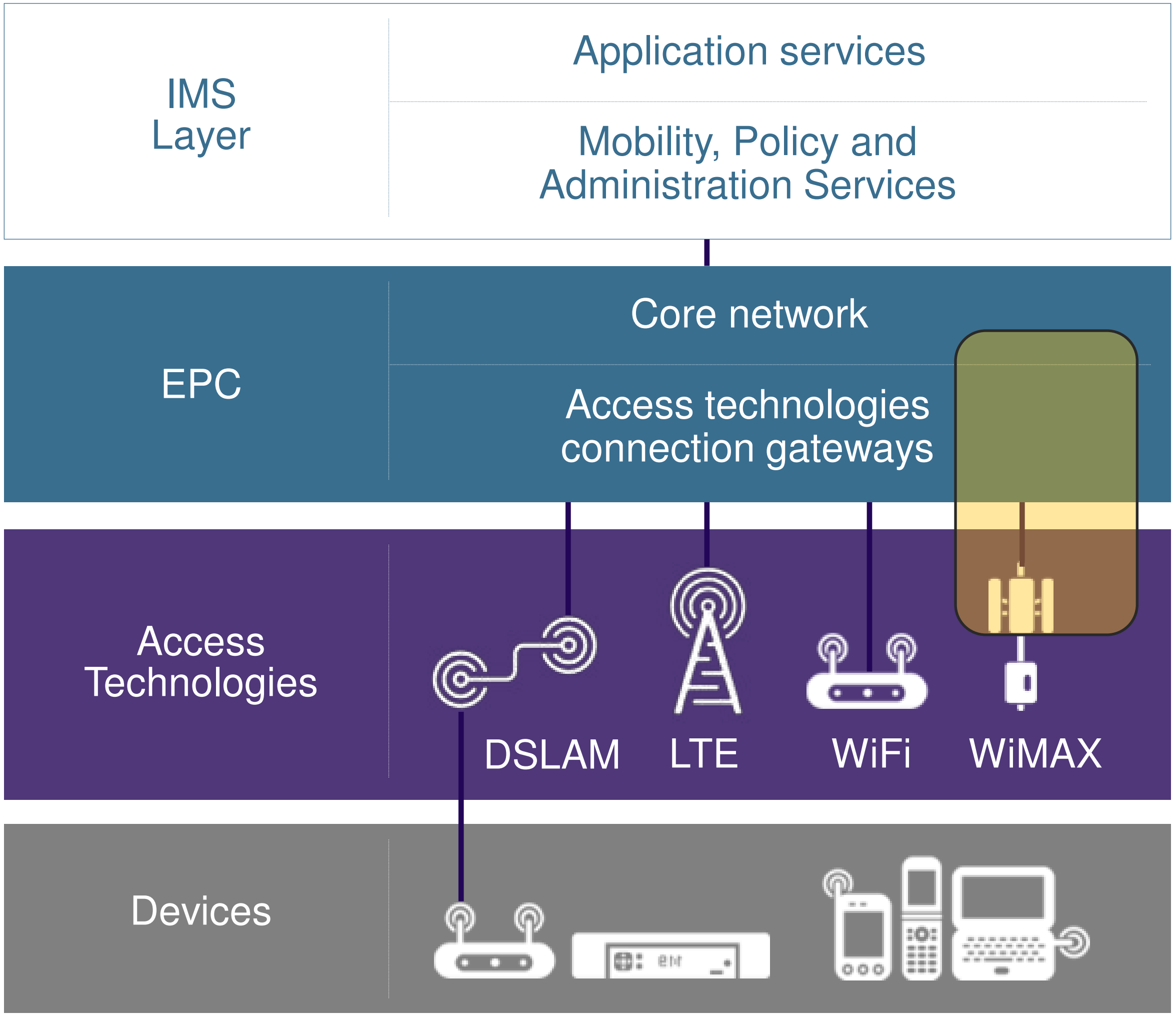
TDMA/OFDM/OFDMA



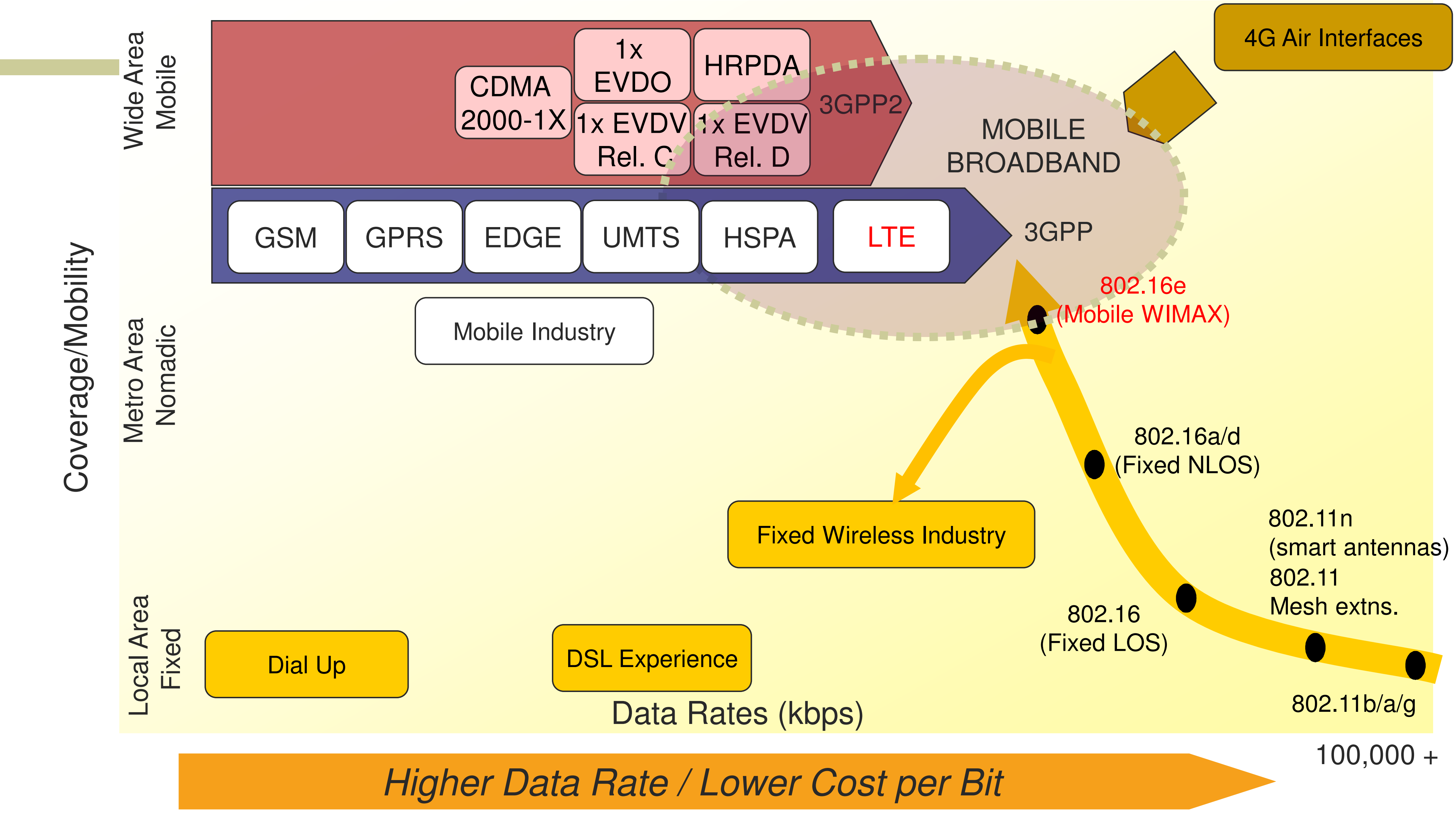
Population penetration of mobile vs fixed across Asia-Pacific



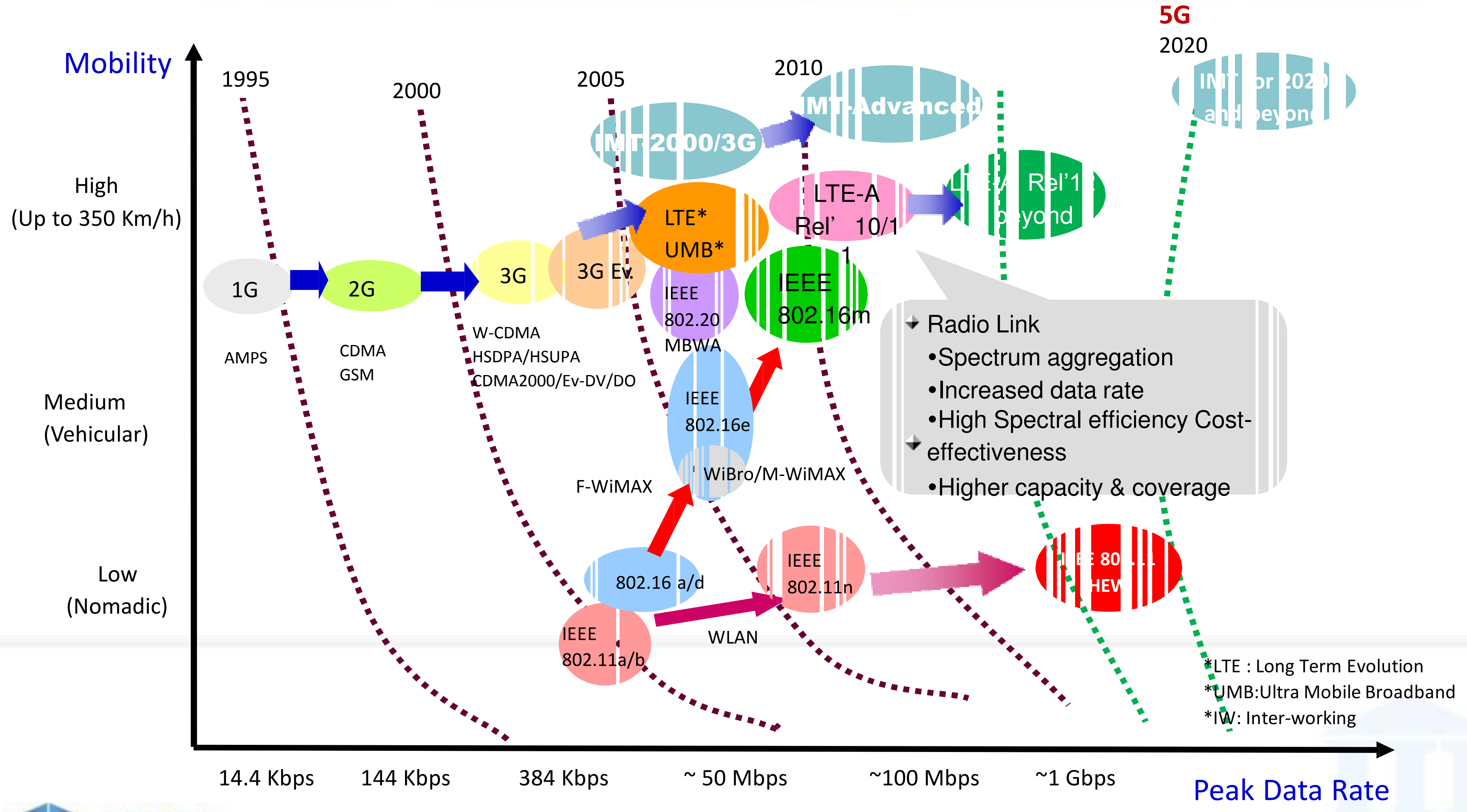
Access Service Stack



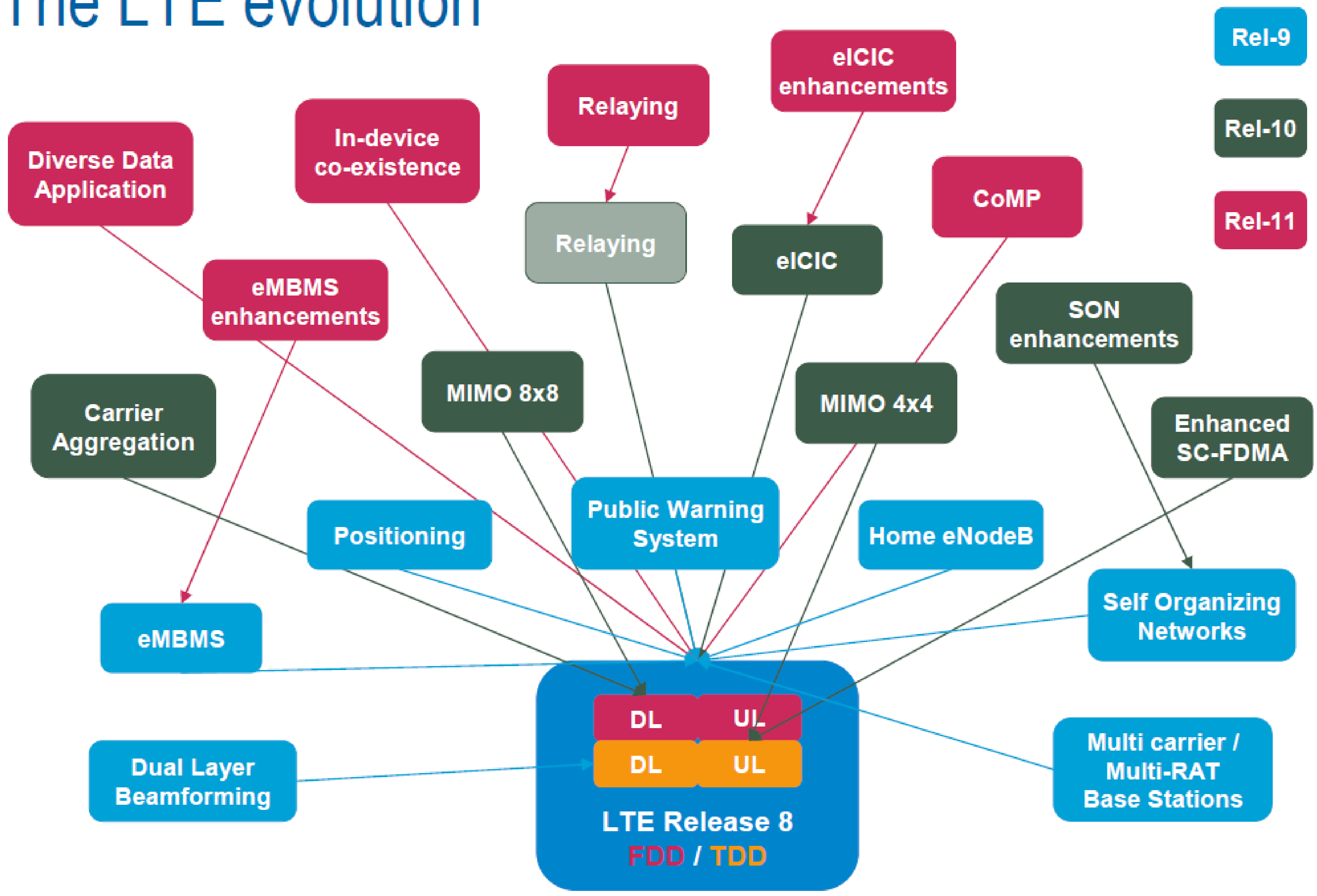
Evolution of Wireless Access Technologies



Standard Evolution for Wireless Access



The LTE evolution

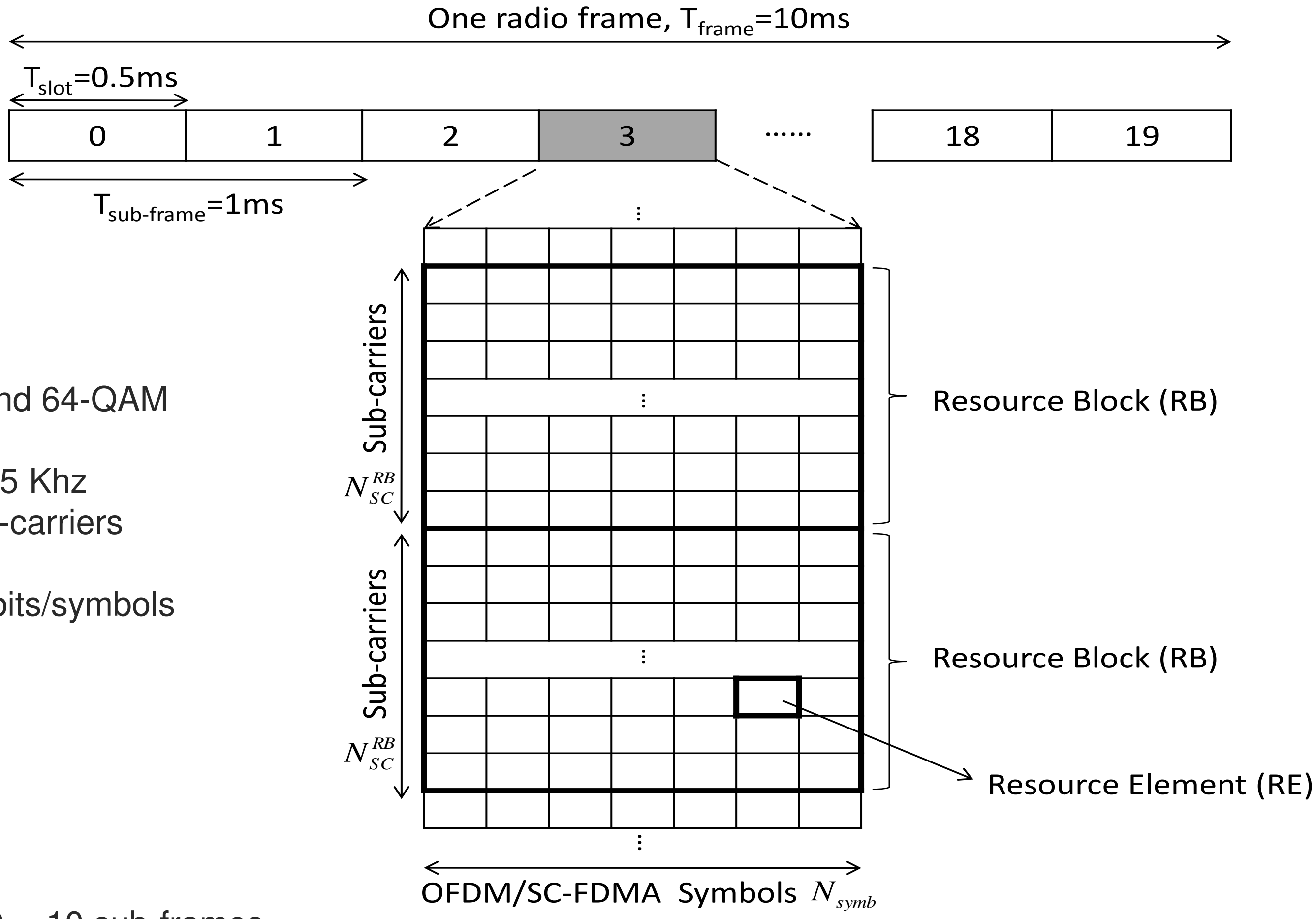


4G: IEEE 802.16m and LTE-A

- ITU-R’s IMT-Advanced (4G) requirements
  - up to 1 Gbps in static or low mobility environment
  - up to 100 Mbps in high-speed mobile environment

- Multicarrier is the technology to utilize wider bandwidth for parallel data transmission across multiple RF carriers.
  - IEEE 802.16m
  - LTE-A
    - Carrier Aggregation (CA)
    - Component Carrier (CC)

LTE-A Frame Structure



Assume total BW = 30 Mhz and 64-QAM
One sub-carrier = 15 Khz
Total sub-carriers = 30 MHz/15 Khz
= 2000 sub-carriers
Total capacity (Data rate)
= 2000 x 14000 symbols x 6 bits/symbols
= 168 Mbps

One OFDMA frame (10 msec) = 10 sub-frames
One sub-frame (1 msec) = 2 slots
One slot (0.5 msec) = 7 symbols
Symbol rate = 7/0.5 msec = 14000 symbols/sec
One RE = One symbol x one sub-carrier One RB = 7 symbols x N sun-carriers

Interactive Multimedia Display System

- A Bi-Directional Interactive Communication System
- Image Plans and Video Graphic Mode
- Texts and Graphics Mixed Mode
- Video Graphics and Texts Display Processors in A Digital Format
- Information Retrieval Between Video Display Terminal and Terminal
- Information Retrieval Between Video Display Terminal and Database (Information Provider)

Worldwide Video Standards

	NTSC	PAL	SECAM
Line / Field	525 / 60	625 / 50	819 / 50 “E” Mono 625 / 50 “L” Color
H. Frequency	15.734 KHz	15.625 KHz	20.745 KHz “E” 15.625 KHZ “L”
V. Frequency	59.94 Hz	50 Hz	50 Hz “E” & “L”
Color Subcarrier	3.579545 MHz	4.433618 MHz	4.40625 MHz OR 4.25000 MHz OB
Sound Carrier	4.5 MHz (FM)	6.0 MHz (FM)	6.5 MHz (AM) “L”
Video Bandwidth (Y)	4.2 MHZ	5.5 MHz	10 MHz “E” 6.0 MHz “L”
Video Component	R G B Or Y I Q or Y B-Y R-Y	R G B Or Y U V	R G B Or Y U V
Interlaced	2 : 1	2 : 1	2 : 1
Frames / Second	30	25	25
Aspect Ratio	4 : 3	4 : 3	4 : 3

HDTV Standards

	Japan	USA	Europe
Line / Field	**1125 / 60**	**1050 / 59.94**	**1152 / 50**
H. Frequency	**33.7495 KHz**	**31.468 KHz**	**31.25 KHz**
V. Frequency	**60 Hz**	**59.94 Hz 50 Hz**	
Video Bandwidth (Y)	**30 MHz**	**40 MHz**	
Chrominance BW (B-Y)	**15 MHz**	**20 MHz**	
Chrominance BW (R-Y)	**15 MHz**	**20 MHz**	
Interlaced	**2 : 1**	**2 : 1**	**2 : 1**
Frames / Second	**30**	**30**	**25**
Aspect Ratio	**16 : 9**	**16 : 9**	**16 : 9**

Video Compression Techniques

Type Compression (CODEC)	Rate	Formats	Application
H.261	p x 64Kbit/s (p is in the range 1-30).	QCIF, CIF	PSTN, PSDN
H.263	20-30kbps and above	QCIF, CIF SQCIF, 4CIF 16CIF. SQCIF	PSTN, PSDN, Video Conferencing, Video Telephony
H.264	Less than 1 Mb/s	MPEG-4 AVC	Internet Protocol-based broadcast-quality video
MPEG 2 IS-13818	4 Mbps or higher	Progressive coding	broadcast quality video
MPEG4 'ISO/IEC 14496'	Less than 1.15Mb/s	MPEG-4	Digital television, Interactive graphics applications, Interactive multimedia

Circuit-Switched Network

- Characteristics - Constant Bit Rate; Full Bandwidth After Call Setup; Low Latency, Constant Delay; Incremental Bandwidth Available (Add B Channels)
- Protocols - ISDN, Robbed Bit Signaling
- Medium - T1 / Fractional T1, DDS at 56 kbps Lines
- Addressing Schemes - Use Phone Numbers; Statically Assigned; Public Directory Assistance if Unknown
- H.320 Terminals
- Intended for Voice Transmission
- Data Transmission Using Modems
- High Quality connections – Low Delay, High Reliability, full Duplex 56 / 64 Kbps Channels
- Basis for Toll Quality

Packet-Switched Network

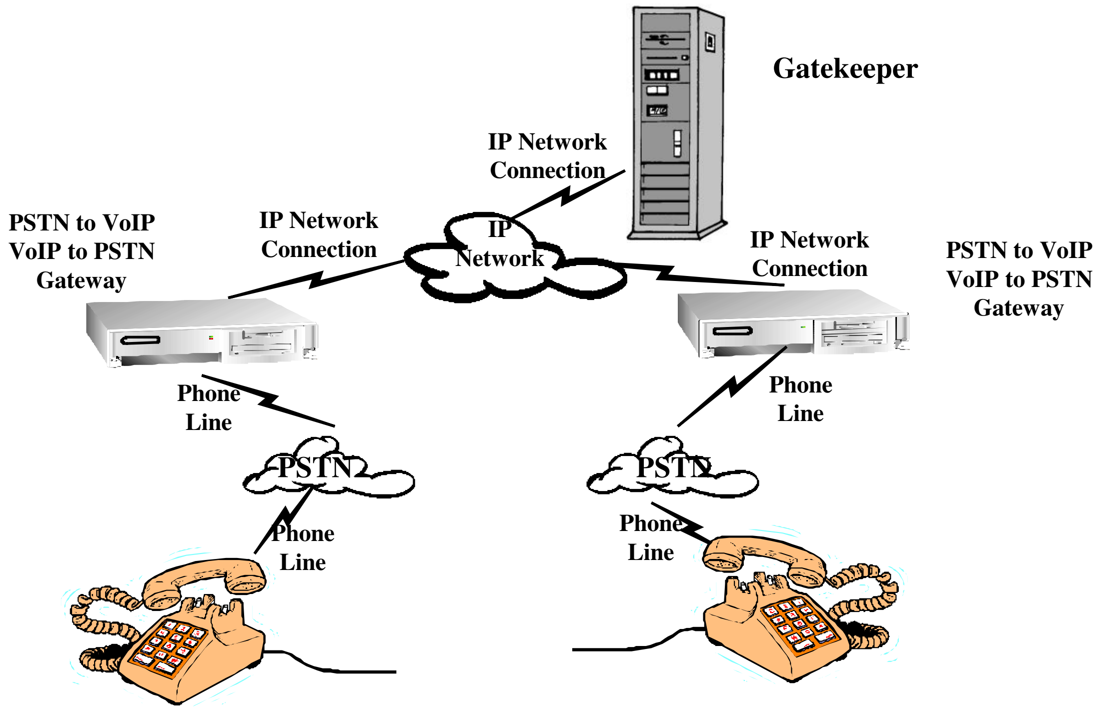
- Characteristics - Burst Mode; Variable Bit Rate; Variable Latency and Variable Delay; Non-Guaranteed Quality of Services in Current Network Topologies; Incremental Bandwidth Quickly Becoming More Available
- Protocols - TCP/IP, ICMP (Internet Control Message Protocol), DHCP (Dynamic Host Configuration Protocol)
- Wired Internet – MAC layer: Ethernet, Frame Relay, ATM, MPLS, and TCP/IP Transport layer
- Addressing Schemes - IP Address; Static or Dynamic Assignment; Directory Servers - Maintain User IP Address by Name or Alias; IP Address Can Change Depending on Your Location (Mobile IP)

Multimedia Networks

The Adoption and Implementation of Digital Contents over IP Can be justified by the Following:

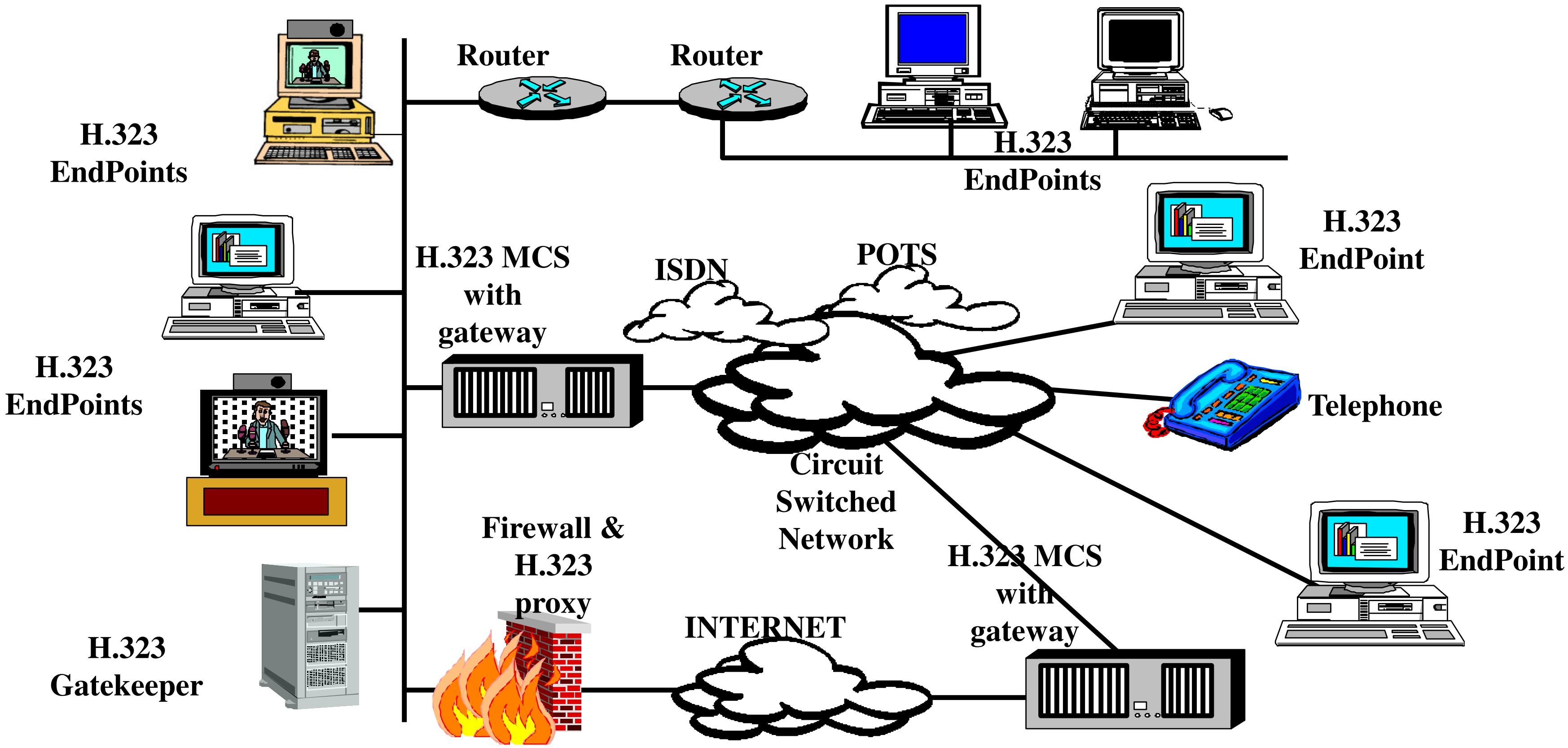
- Increasing Voice/Data/Video Convergence
- IP is Now the "Common Protocol"; RSVP Protocol for Bandwidth Reservation and RTP Protocol for Detecting Missing Packets to Improve Quality of Services
- Packetized Compressed Voice Has Shown Cost-Effective Solutions (High-Class Coding Algorithms)
- Intranets and Extranets are Growing Rapidly
- Voice over IP (VoIP) is Being Successfully Deployed in Major Corporate Network
- The Rapid Growth of Digital Multimedia Contents in Internet

VoIP Network Topology

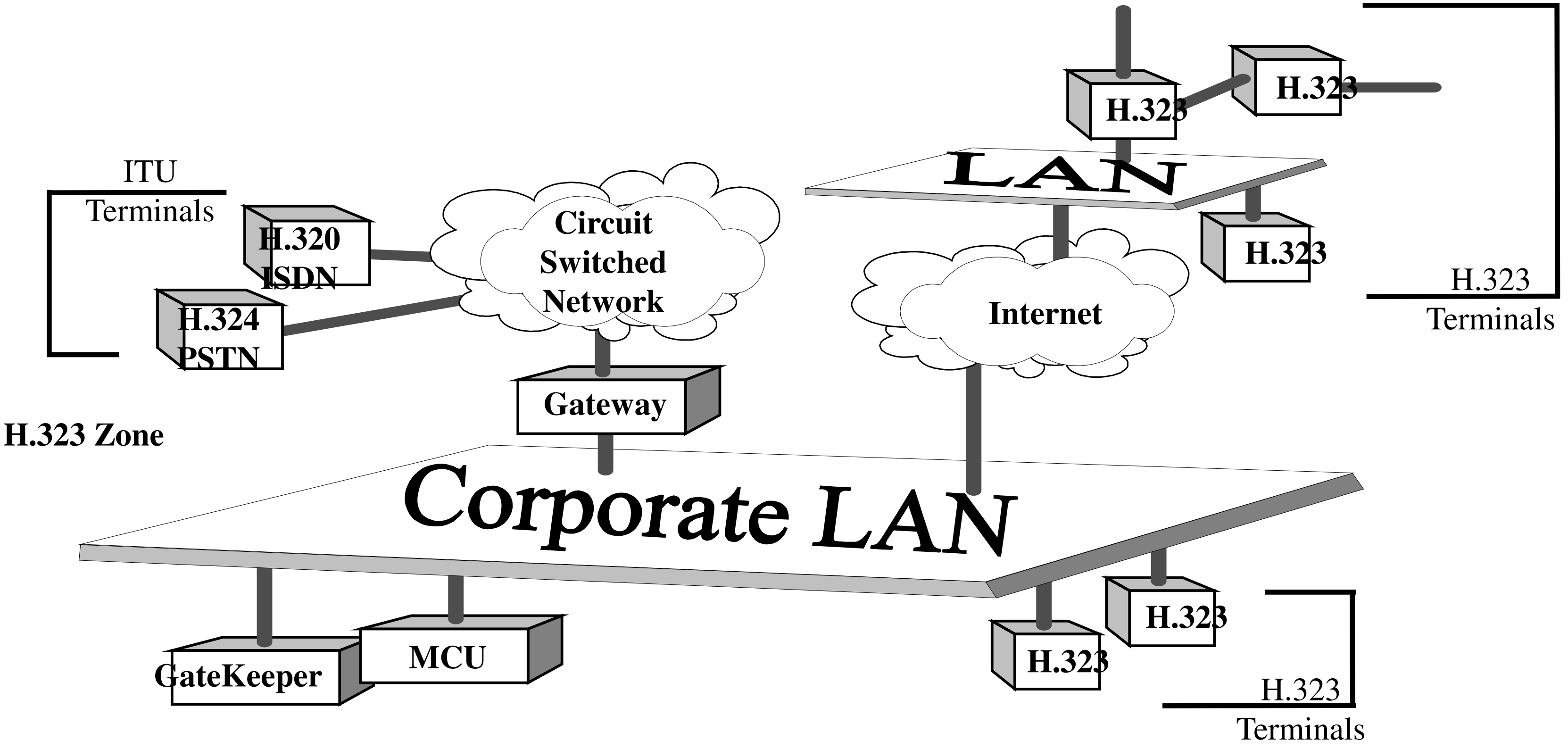


Equipment to Bridge the Circuit-Switched Network
and Packet-Switched Network

MoIP Network Topology



MoIP Network Topology



Major Entities in an H.32X Environment: H.323 Terminals, Gateways, Gatekeepers and MCUs.

Major System Components

Terminals

- Bi-Directional Real-Time Communication for the User
- Enable Voice Connections, Video and Data Communication are Optional
- Supporting H.245 for Describing the Negotiation of the Appropriate or Required Terminal Functionality

Gateways

- Translate Between Physical Media, Network Protocols, Conferencing Protocols and Addressing
- Translate Between Different Audio or Video Codes in Real Time, Allow H.320 Endpoints to Use T.120 Conferencing Effectively
- Support Voice Over IP and Multimedia Conferencing

Major System Components (Continued)

- Locate at Multimedia Conference Servers, Stand-Alone Equipment, Network Servers, Routers, Remote Access Servers, Multimedia PBX, Network Services Authorization and Authentication
- Serve Conferences Interoperability Between H.320 and H.323 Endpoints Complying With Different ITU Standards, Diverse Network Transport, Diverse Audio or Video Codecs
- Connect Incompatible Devices by Device Emulation at the Network Level; a Good Gateway is Invisible Just Like the Gateway Embedded in the Telephone Network
- A media gateway provides translation of protocols for call setup and release, conversion of media formats between different networks:
- Transfer of information between H.323 and SIP networks on an IP Network
- Translation between transmission formats and communication signals and procedures (e.g. between IP and PSTN)
- Passes call signaling not applicable to the media gateway through to the network endpoint (e.g. supplemental services such as call forwarding)
- Performs call setup and clearing on both sides
- Translates between encoding formats

Major System Components (Continued)

Gatekeepers

- An Optional Element of H.323
- Link Endpoints Directly to Gatekeepers
- Reside Anywhere in H.323 Network Entities or Run as Server Application
- Do Bandwidth and Resource Management, Access Control, Endpoint Registration, Zone Definition, Enhanced Call Control and Address Translation
- Platform Independent
- An Embedded Component in Hardware Building Block
- Gatekeeper "Engine" Software Development Application

Major System Components (Continued)

Multipoint Control Units

- Provide Audio Bridge With Value-Added Video Multipoint Capabilities, Unattended Operation, Full Audio Mixing With Advanced Techniques for High Quality Compressed Speech
- Support Multicast and Unicast Sessions

H.323, SIP, MGCP, H.248

- **H.323**
  - IP communications protocol for real-time voice and video over IP.
  - Includes core protocol and gatekeeper toolkits.
  - International Telecommunications Union (ITU) recommendation for audio, video, and data communications across IP-based networks.
- **SIP (Session Initiation Protocol)**
  - Signaling protocol for establishing real-time calls and conferences over IP networks.
  - SIP is an IETF (Internet Engineering Task Force) Protocol.
- **MGCP (Media Gateway Control Protocol)**
  - A complementary IETF protocol to H.323 and SIP
  - Defines the communication procedures for a Media Gateway Controller to provide instructions and to gather information from Media Gateways
- **Megaco/H.248 (Media Gateway Control)**
  - Similar to MGCP, jointly defined by the IETF and ITU-T SG-16
  - Gradually replacing MGCP
  - Megaco renamed GCP (Gateway Control Protocol) -- RFC 3525

RTP / RTCP

Real-Time Transport Protocol (RTP)

- Provides end-to-end delivery services of real-time Audio (G.711, G.723.1, G.728, etc.) and Video (H.261, H.263),
- Data is transported via the user datagram protocol (UDP).
- RTP provides payload-type identification, sequence numbering, time stamping, and delivery monitoring.
- UDP provides multiplexing and checksum services.
- RTP can be used with other transport protocols.

Real-Time Transport Control Protocol (RTCP)

- Counterpart of RTP that provides control services
- Primary function of RTCP is to provide feedback on the quality of the data distribution – RTCP-XR
- Carries transport-level identifier for an RTP source
  - Used by receivers to synchronize audio and video.

Quality of Services (QoS)

Technical Constraints

- Latency is the Most Technical Problem Over Internet Telephony: by Delay, Delay Variance (or Jitter), Asymmetrical Delay, and Unpredictable Delay
- Twenty (20) ms Coast-to-Coast Delay in the U.S. : Mostly Not Noticeable
- Fifty (50) ms Delay is Noticeable
- 250 ms Delay by the Satellites - Conversation Becomes Difficult
- 350 ms Delay Over the Public Internet From Encoding and Packetizing at Both Ends of the Call
- Standard Half-Duplex Sound Card: Amateur Radio Conversation Quality
- Latency is Dependent on Lost a Packet (30 ms) or Packets, Packet Size, Buffer Size, Speaker Behavior Parameter, Protocol Application, Frame Delay, Speech Process Delay, Bridging Delay, PC Too Overloaded to Run Vocoder, and Protocol Limitations

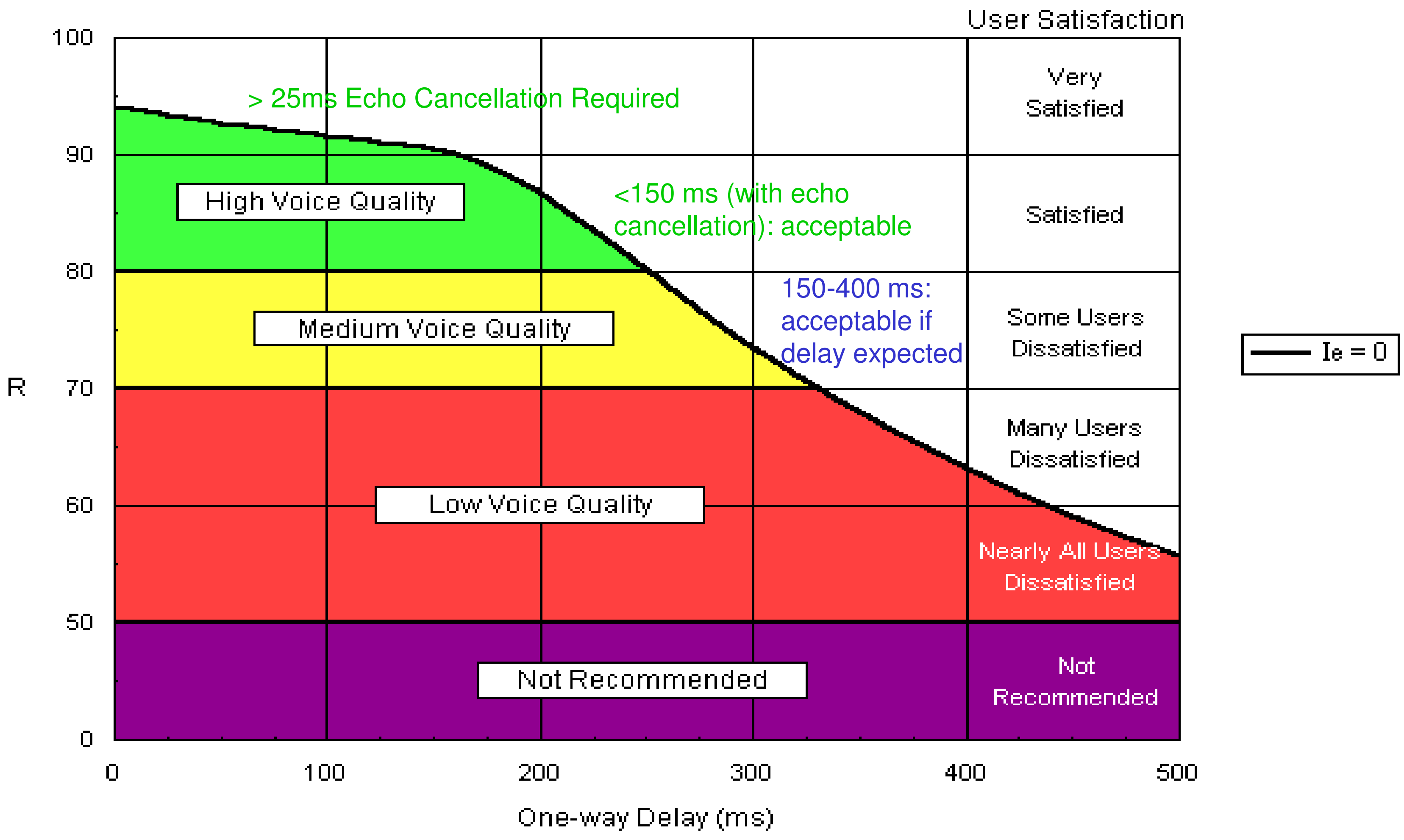
Quality of Services (Continued)

Performance Evaluations:

- Delay 200 Milliseconds From a Private IP Network With Good Encoding and Excellent DSP Technologies
- Laboratory Demonstrations to Analyze Voice Quality With 100 ms, 150 ms, 200 ms, and 250 ms Latency With the Following Setups:

1. Workstation-to-Workstation Using the Gatekeeper
2. Workstation-to-Phone Using the Cisco 3620 as a H.323 Gateway
3. Phone-to-Phone Using Netrix 2210 and Cisco 3620 for Calls Connections Through IP Network

Effect of Delay on Voice Quality



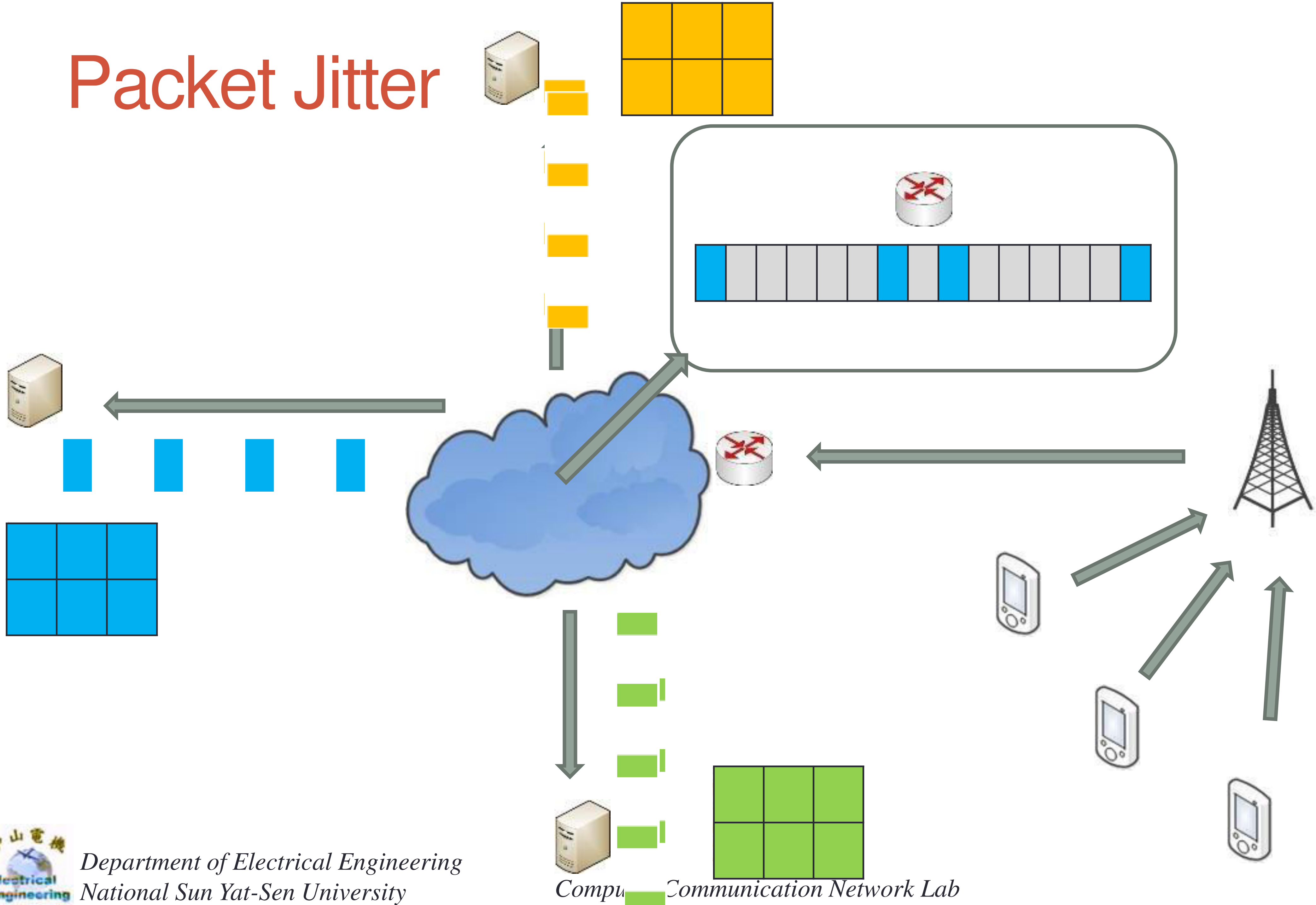
Technical Challenges in Multimedia Networks

- Resource Reservation - It is a Receiver-Driven and up to the Receiver to Select which Source to Receive and Amount of Bandwidth to be Reserved or Paid for
- Parallel IP Networks - Different Bandwidth Allocations for Data and Multimedia by Virtual or Physical network
- Voice Traffic on Circuit Switched Networks
- Parallel or Overlay Networks are Being Built to Support Real-time Multimedia Traffic
- Today’s DSP Delivers More Than 10 Times the Price/Performance of its Predecessors Five Years Ago, Providing 1000 MIPS for Voice Compression and Thus Reducing Latency
- SDN (Software Defined Network): Centralized routing using cloud

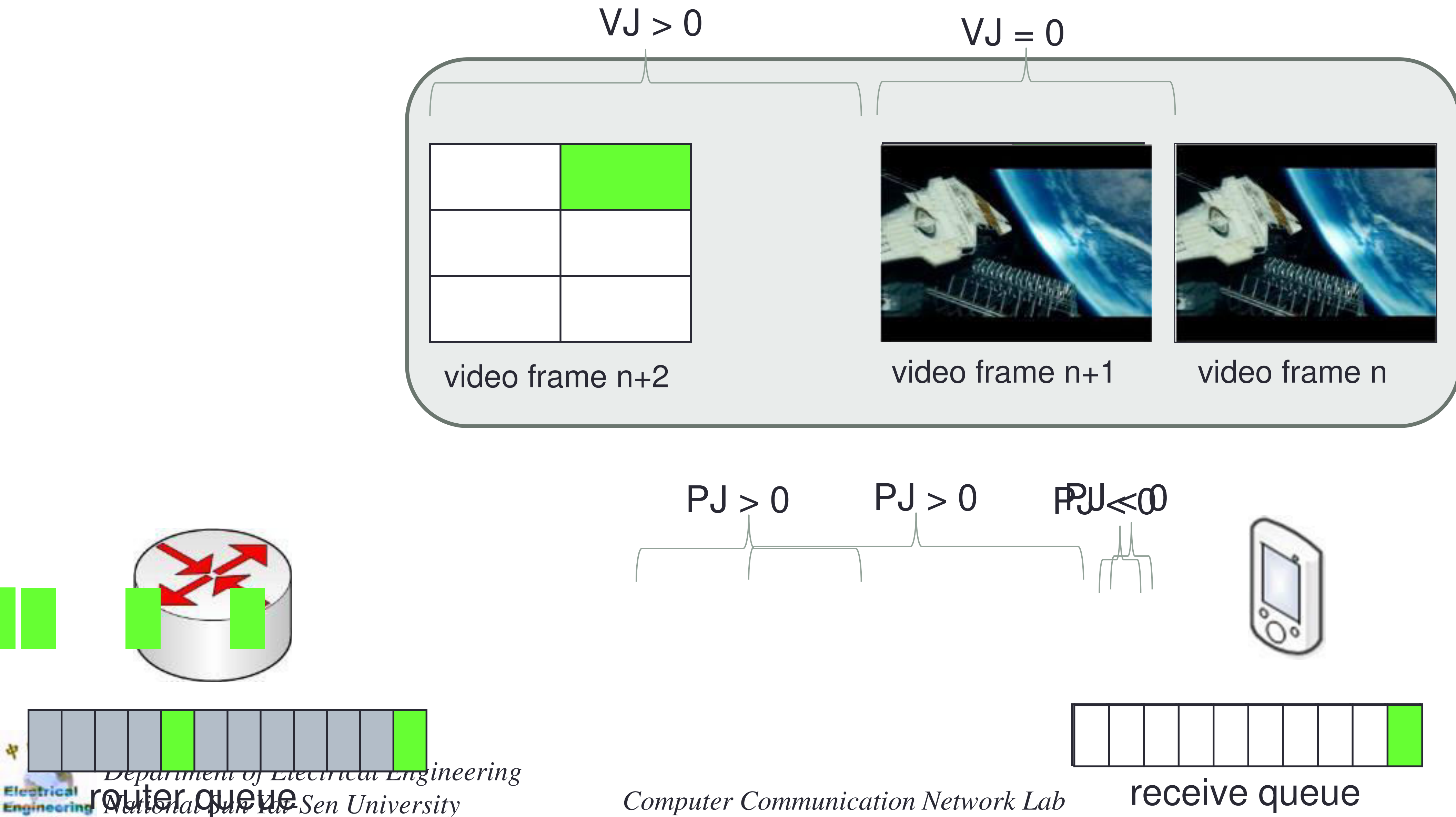
Researches on Multimedia Networks

- Inter-Frame De-Jittering (IFDJ)
  - Tsang-Ling Sheu and Po-Wen Lee, "An Inter-Frame De-Jittering Scheme for Video Streaming over Mobile Communication Networks ," WSEAS Conference, Salerno, Italy, Jun. 2015.
- ARQ Block Retransmission (ABR)
  - Tsang-Ling Sheu and Ching-Hua Li, “An ARQ Retransmission Scheme for Real-Time Video Multicasting over Mobile Communication Networks,” To be presented in this Multimedia Conf., Birmingham, UK, Aug. 2015.
- Off-loading in LTE-WiFi
  - Paper is being prepared

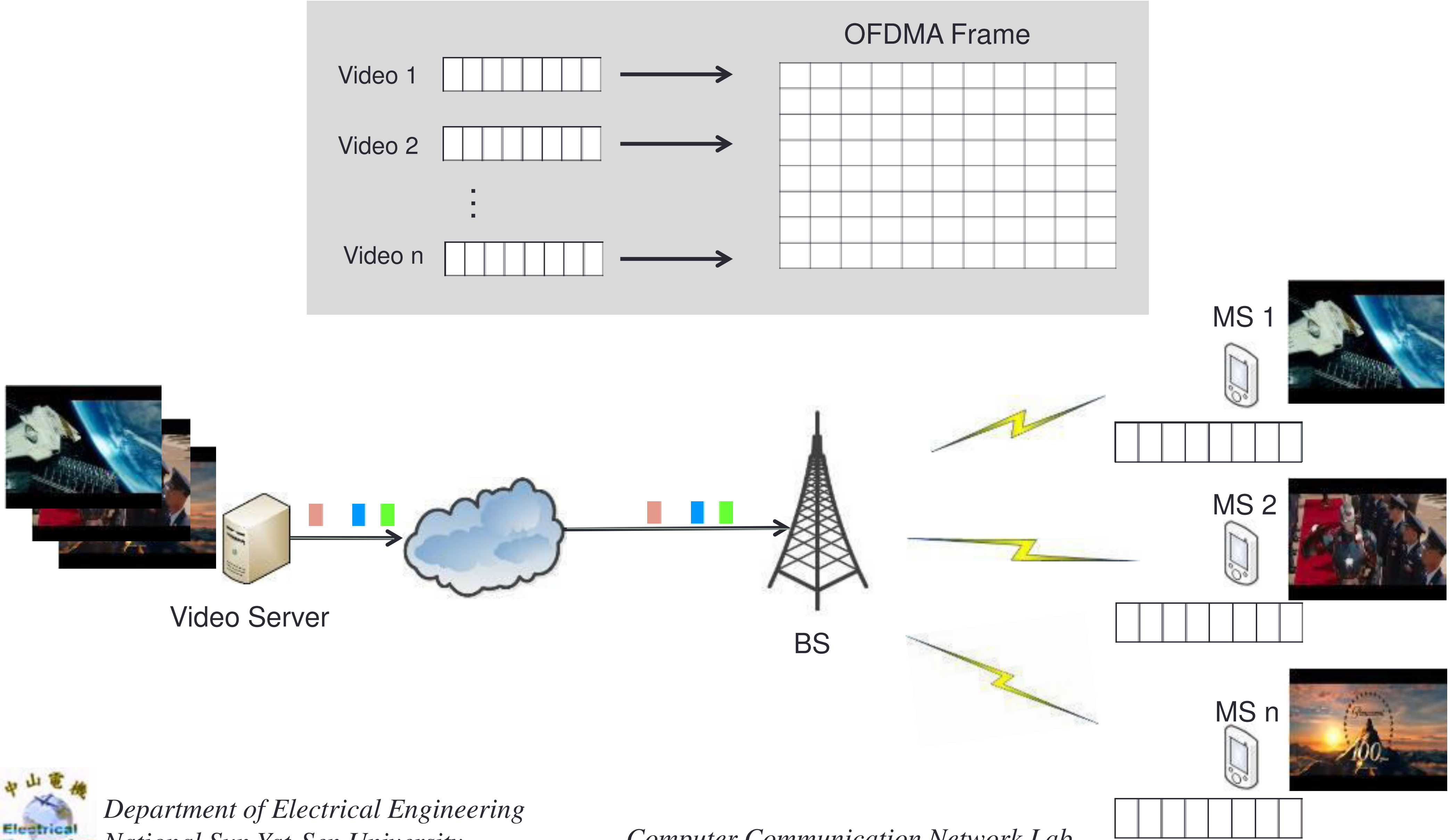
Packet Jitter



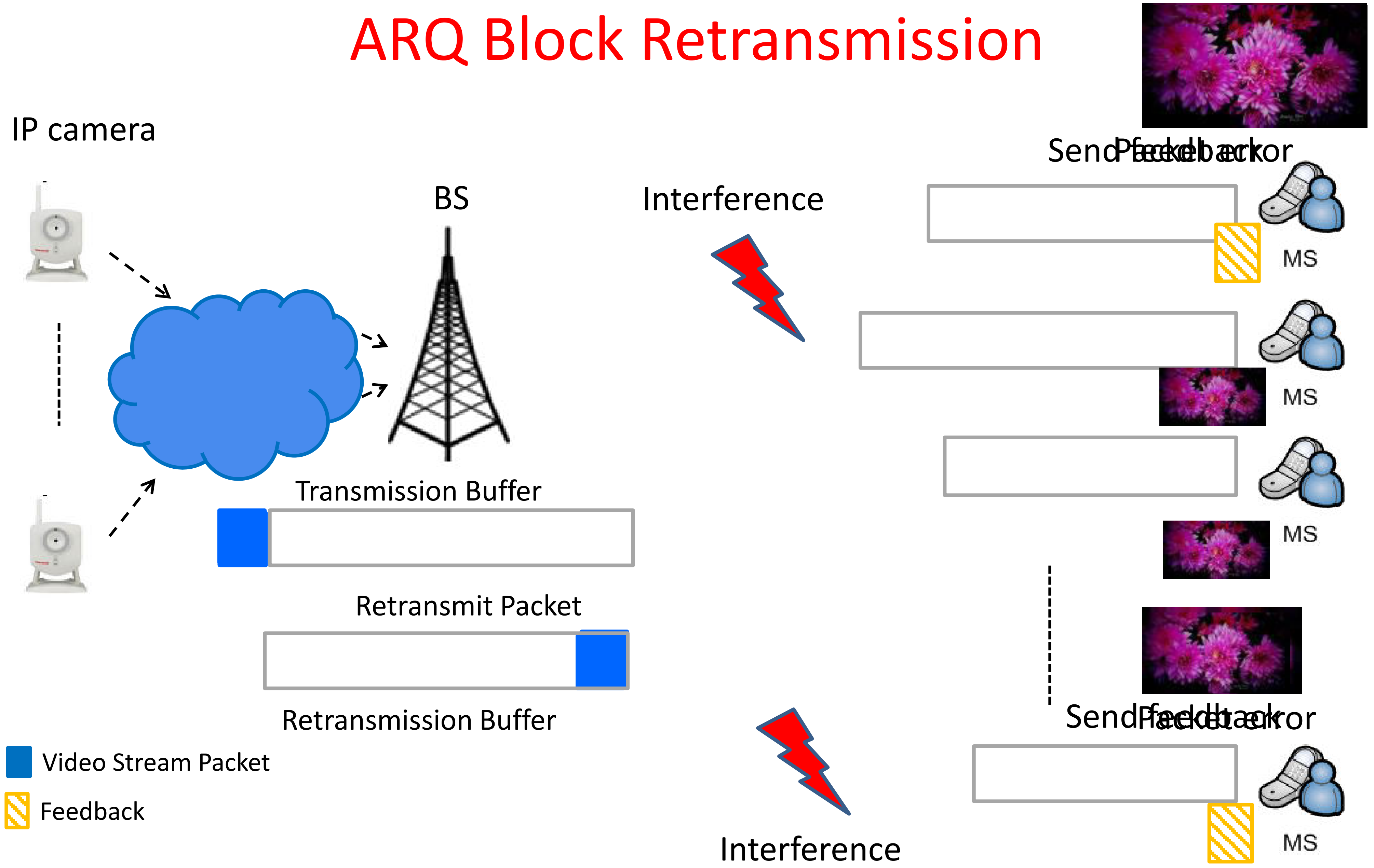
Video Frame Jitter



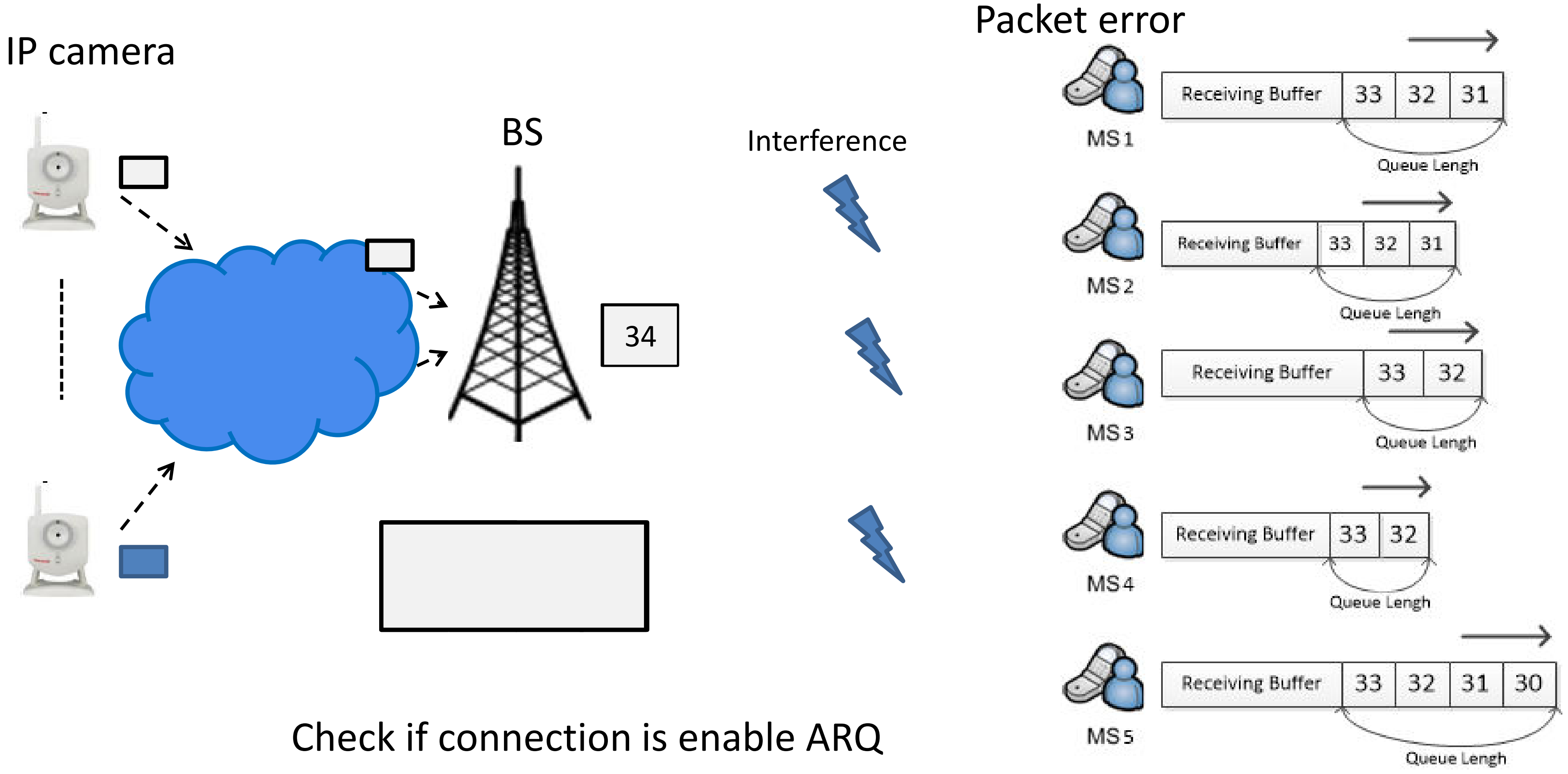
System Architecture



ARQ Block Retransmission



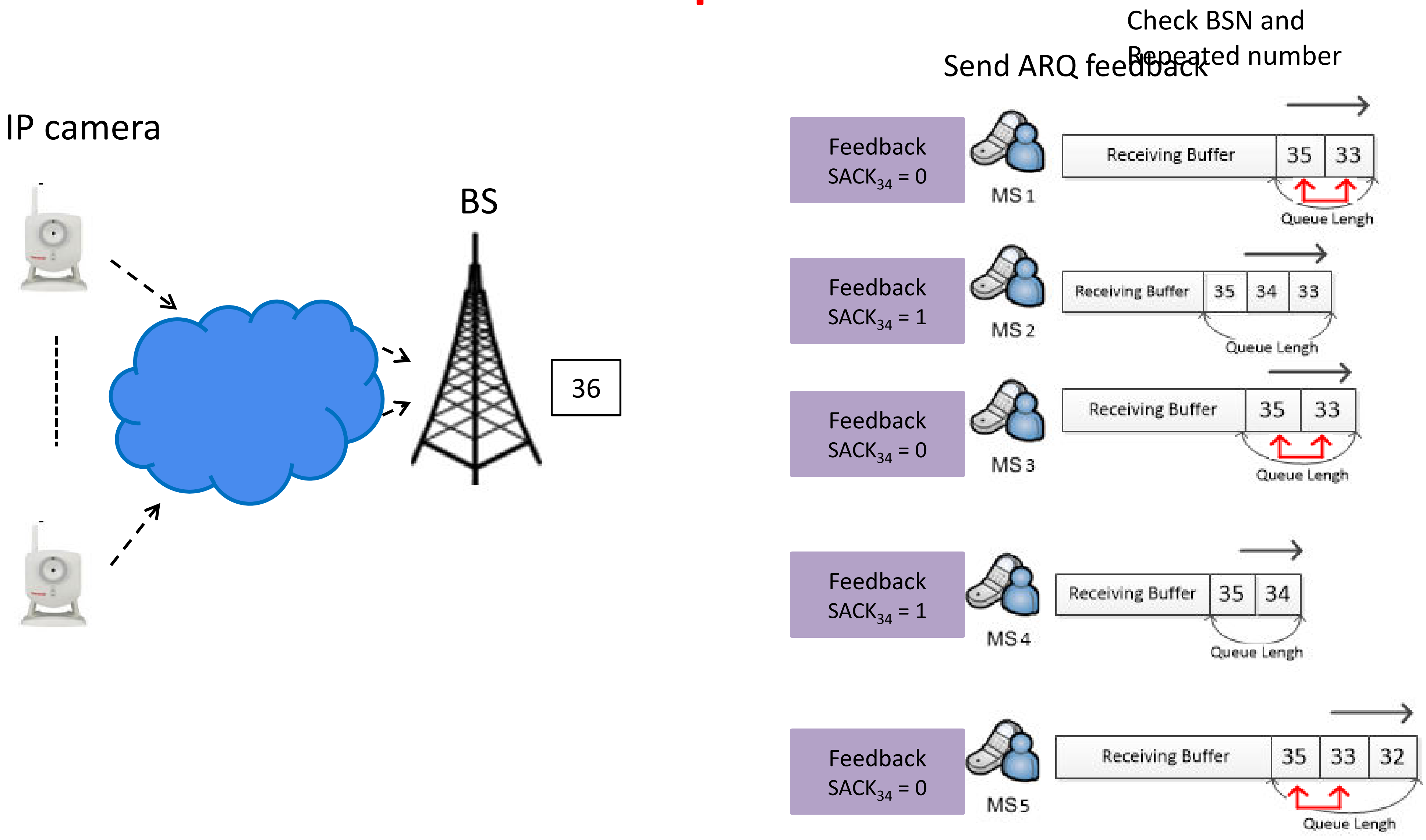
The Proposed ABR



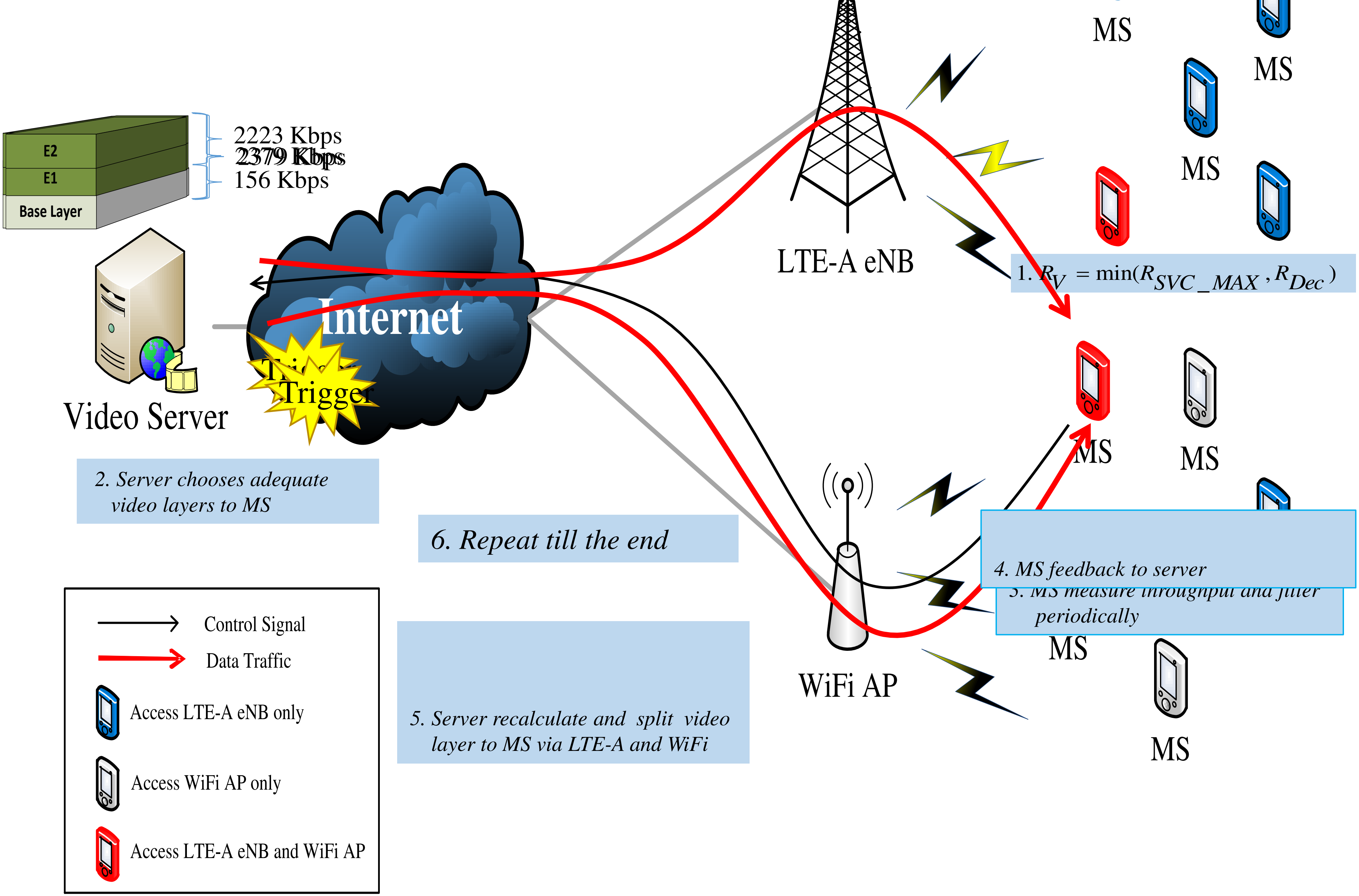
Check if connection is enable ARQ

Divide into ARQ blocks

The Proposed ABR



Off-loading in LTE-WiFi



Conclusions

- Wireless Communications and Technologies
  - WiFi vs LTE-A
  - First-hop vs Last-hop
- Challenges in Multimedia Networks
  - Compression, Multicasting, Separate Networks
  - QoS Guarantee: Delay, Jitter, Packet Loss Rate
- Researches
  - Inter-Frame De-Jittering
  - ARQ Block Retransmission
  - Off-loading in LTE-WiFi

Thank you

Q & A