Video Communications Over Mobile IP Networks

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- Universal Mobile Telecommunications Systems (UMTS)
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- Quality Control
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Aim of 3G Mobile Network

- Combination
  - Multimedia Service of Internet
  - Digital Cellular Concept of Mobile Network
- Real-Time Mobile Multimedia Service
  - Over Mobile Wireless Platforms
  - Combinations
    - Video/Speech/Audio
    - Data Text
  - High Data Rates

3G Mobile System (IMT-2000)

- Better Voice Communication Service than 2G
- Connection to Internet
  - E-mail and Web Service
- Conversational Video Communication
  - Multi-User Capability
  - Multi-Party Video Conferencing
  - Among various Fixed and Mobile Users
- Internet Connection (Ubiquitous Connection)
  - E-Commerce/E-Business
- Selective and On-Demand Coverage of Live Events
  - Breaking News, Sports in the Form of Streaming Audiovisual Content
  - High Quality Highlight of TV Scene/Remote Audiovisual Clips

End-to-End Mobile Network

- QoS: Function of Different Connection Parameters
  - End-to-End delay, Error Rate, Frame Dropping Rate
- Each Mobile Terminal
  - Access to A Number of Channels
  - Each Channel Offering Different QoS to Various Services
- Internet Protocol
  - Widespread Success
- Hypertext Transfer Protocol (HTTP)
- Combination and Interoperability
  - Universally Accepted Application
  - Network-Layer Standards
- Real Time Transmission of Compressed Video Data
  - Encapsulated in IP Packets over the Future Mobile Networks

Video Communication over Network

- Video Communication Networks
  - Video Telephony and Conference
  - Video On-Demand Service
  - Distance Learning Application
- Requirements
  - Networks for Routing of Multimedia Traffic
  - More than Two Users at the Same Time
- Markets of Multimedia Services
  - from PC-based Applications
  - to Multi-Sharing Services on Worldwide Basis

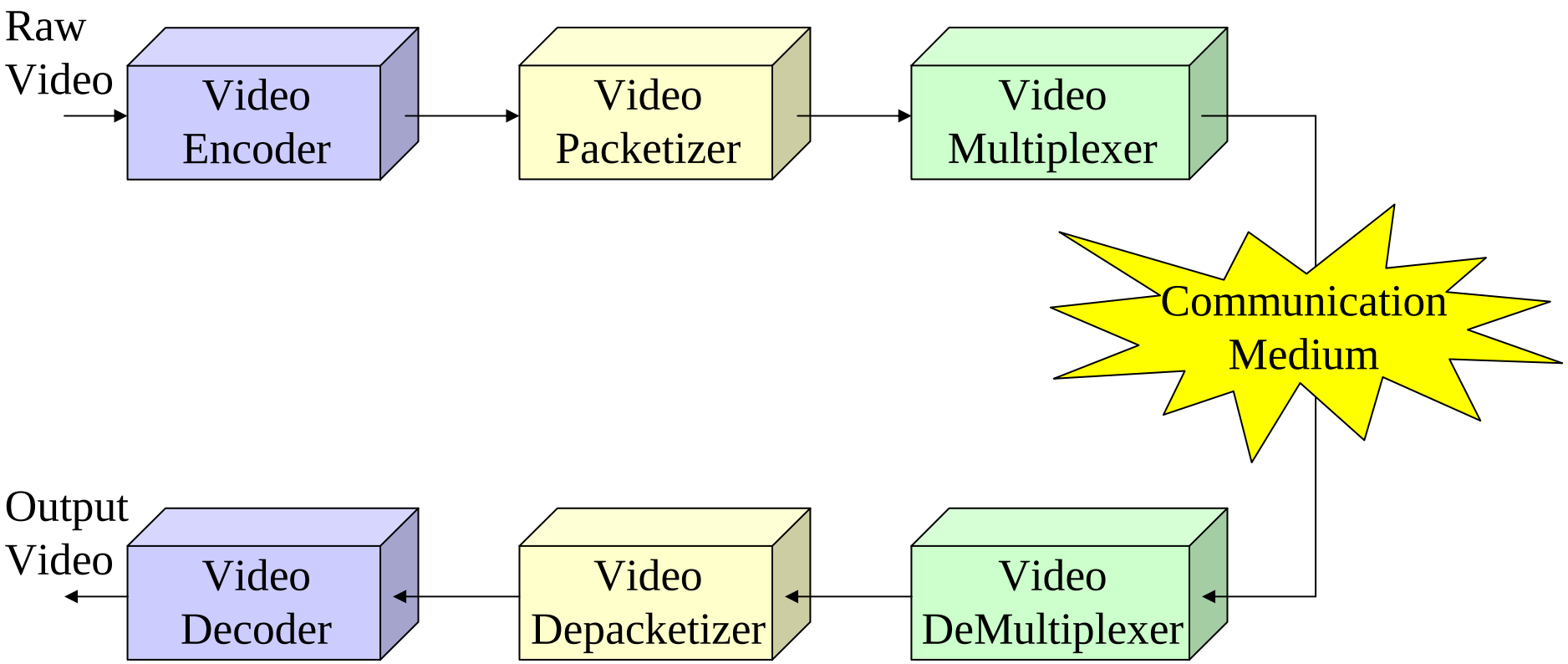
Synchronization

- Time Synchronization Between Sender and Receiver
- Asynchronous Communication
  - Data Stream in the Form of Symbols
  - Symbol
    - Predefined Number of Bits
    - Start Bit and Parity Bit
- Synchronous Transmission
  - Without Any Start and End Indicators
  - Packet
    - Indication of Beginning and End of a Block Data
    - Preamble Bit Pattern and Post-amble Bit Pattern
    - Fixed Length (ATM Cell 53Bytes)
    - Variable Length (IP Packet)

Packets

- Compressed Video Stream
  - Low Tolerance to Delay
  - No Retransmission
  - Possibility of Error Control about Losses
    - Special Data Structure -> Packet
- Packetization
  - Video Payload & Protocol Header

Packetization/Depacketization



Multiplexing by Packet Header

- Traffic Flow between Two End Points
  - Various Traffic Types
- Type Filed in Each Packet Header
  - To Identify Traffic Type of Payload
  - To Multiplex Various Streams onto the Same Channel
- Multiplexing of Various Streams
  - Efficient Sharing of Available Bandwidth
  - Statistical Multiplexing

Resynchronization of Packet

- Information Loss and Bit Errors
  - from Excessive Delays and Interference
  - Serious Degradation of the Decoded Video Quality
- Packetization
  - Effects of Bit Error & Information Loss
    - Affect Only One Single Packet
    - Resynchronization at Following Error-Free Packet
  - MBs in Video Packet
    - Predicted Independently of MBs in Other Packets
    - Independent Segment Decoding (Annex R of H.263+)

Connectionless Service

- High Flexibility
  - In Selection of Path between Two End Points
- High Channel Utilization
  - No Prior Bandwidth Allocation
- Out of Sequence Arrival of Packets
  - Multipath Fading and Varying Network Conditions
  - Reordering Received Packets by Depacketizer

Checking Network Status

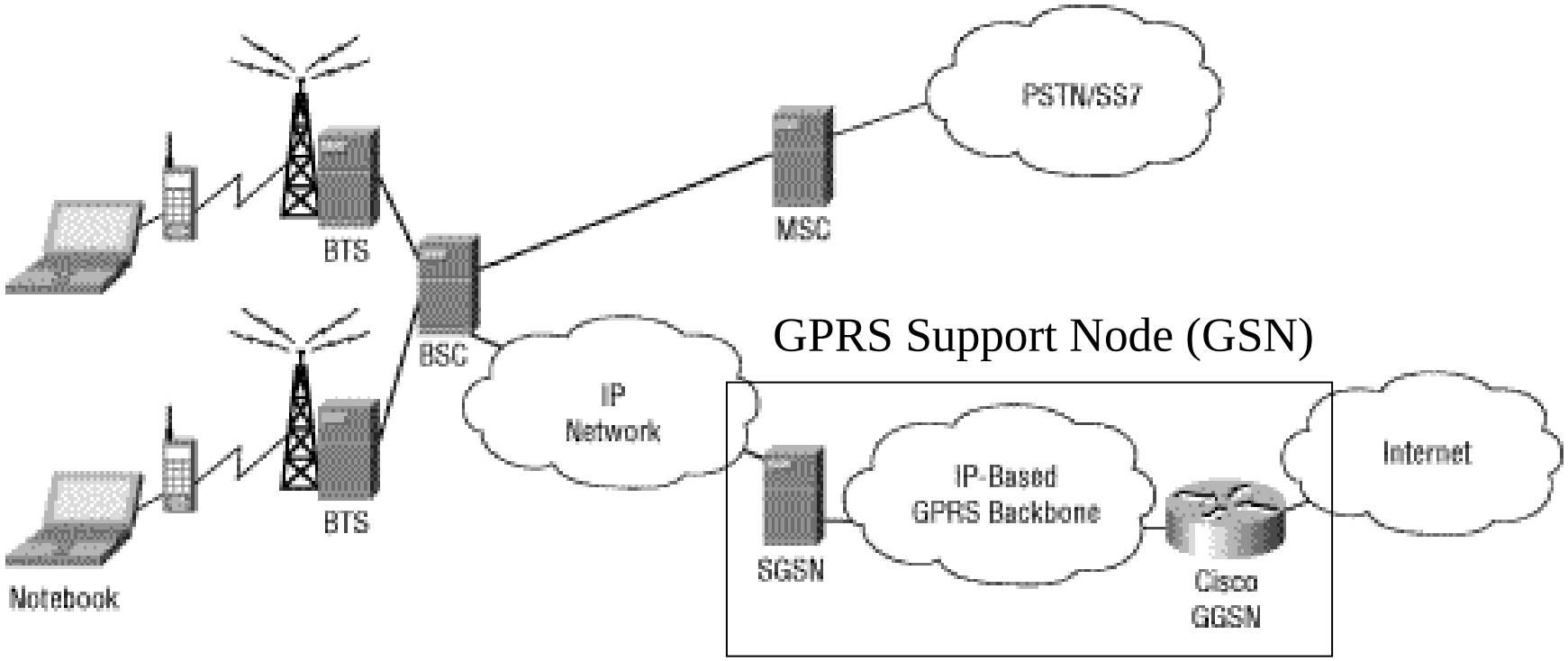
- Network Check by Decoder
  - Acknowledgement of Correct Delivery
  - Periodically Report to Encoder
  - Controls by Encoder
    - Output Rate, Error Protection Mechanism
- Encoder Adaptation
  - on the Latest Status of the Network
  - Flow Control, Error Resilience
  - Reference Picture Selection Techniques (H.263+)
  - Priority Control
    - Dropping Low-Priority Packets in Case of Network Congestion
    - Reducing Output Rate for Graceful Quality Degradation

General Packet Radio Service

- GSM
  - Insufficient Capabilities for the Routing of Packet Data
- GPRS
  - Packet Data Transmission
  - Efficient Radio Access to External IP-based Networks
  - End-to-End Mobile Packet Radio Communication System
  - Same Radio Architecture as GSM
  - Packet Mode Data Transmission/Reception without Circuit Switched Resources
  - Accommodation to Video Communication
    - Multi-Slotting Feature
      - More Time Slot or PDTCH (Packet Data Traffic Channel)
    - IP Support

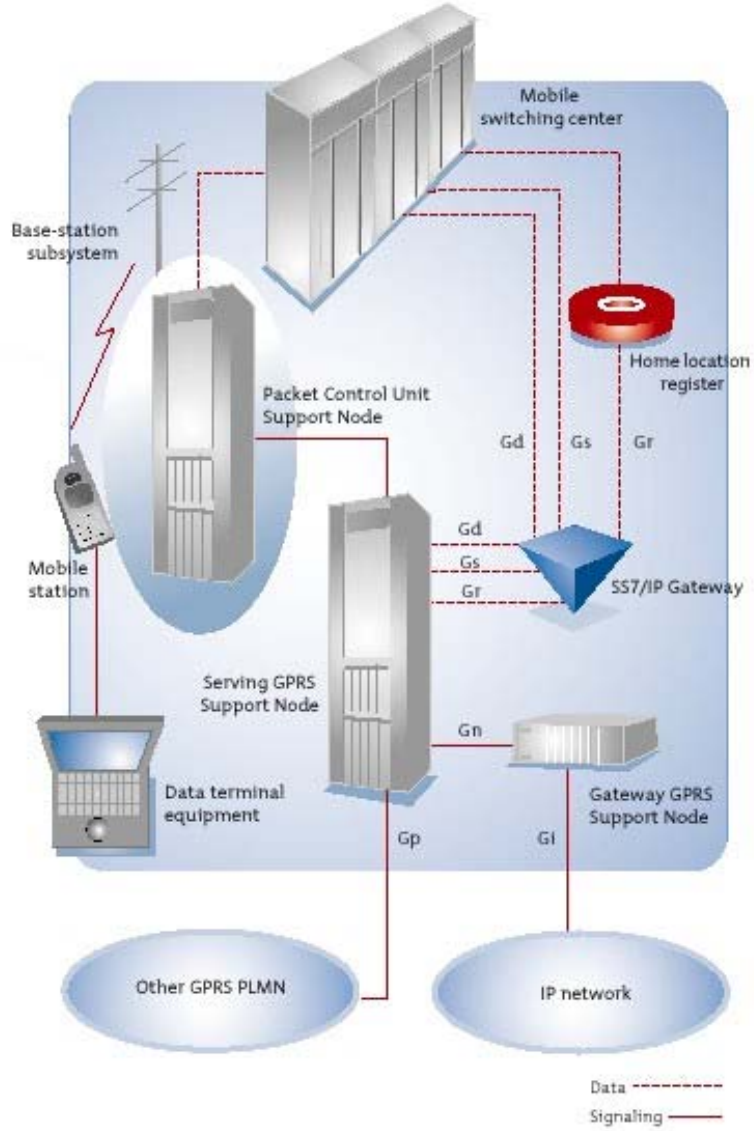
Network Infrastructure for GPRS

- IP-Based Network
- Static or Dynamic IP are Assigned to MS



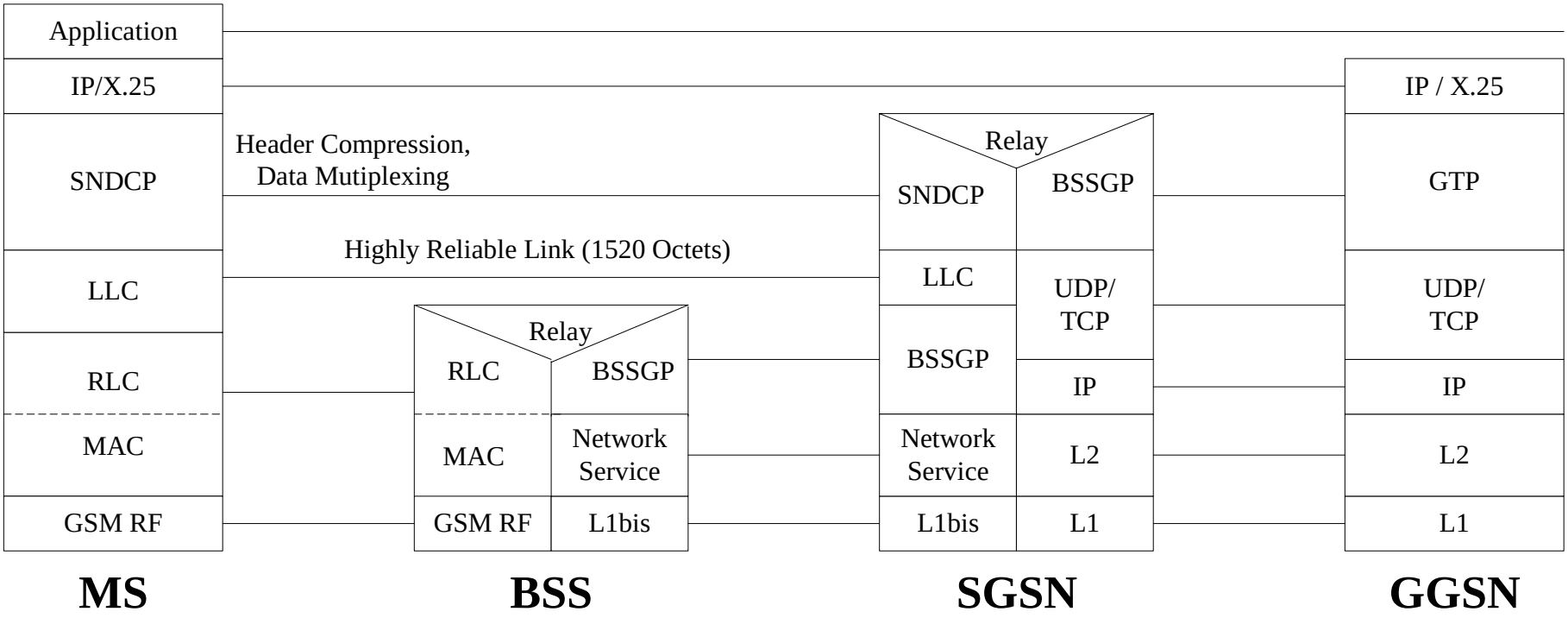
GPRS Support Node (GSN)

- SGSN (Serving GSN)
  - Access Network
  - Location Information
  - Security & Access Control
- GGSN (Gateway GSN)
  - Interworking Unit
- SGSN & GGSN
  - Connected by IP-Based Transport Network



GPRS Logical Protocol Architecture

- MAC (Medium Access Control)
  - Multiple MSs Share a Common Transmission Medium
  - Each Time Slot to be Multiplexed between up to 8 Users
  - Each User to Use up to 8 Time slots
- RLC (Radio Link Control)
- Physical Link Layer
  - Interleaving of Radio Blocks
  - Method to Detect Link Congestion



Enhanced GPRS(EGPRS)

- GPRS in GSM
  - Intermediate Step towards 3rd-Generation UMTS Network
- EGPRS
  - Enough Traffic Availability for Single User
  - Benign Interface Conditions
  - Higher Data Rate than GPRS
  - Same Protocol as GPRS
  - Improvements of Modulation Scheme in Radio Interface
    - Increase in Throughput Availability

UMTS

- Universal Mobile Telecommunications System
- UMTS Infrastructure is Integrated with GSM
- Core Network
  - Circuit Switching Function & Packet Switching Function
- Major Innovation
  - Packet Switched IP Nodes
    - Similar to GPRS
    - BSS Access Segment is Replaced by UTRAN
- Universal Terrestrial Radio Access Network
  - W-CDMA (Wideband Code Division Multiple Network)
  - Interface I_u: Connection between UMTS and UTRAN
    - Specialized in managing both Packet-Switched & Circuit-Switched Components

Corruption in Packet Network

- Three Possible kinds of Errors
- Packet Sequence Number is Affected
  - Unable to Figure Out Correct Order of Packet Transmission
  - Depacketizer Fails to Merge Consecutive Video Packets
- Bit Errors in Payload Generate Packet Delimiter
  - Incorrect Split of Video Data
  - Loss of Synchronization
  - Many Subsequent False Merges/Splits of Video Packets
- Errors of Payload
  - More Frequent Because of Higher Proportion of Packet Length

Other Network Problems

- Network Congestion & Link Overflow
- Completely Discarding the Video Packet
  - Excessive Amounts of Delay
- Some Intelligent Content-Based Packetization!
  - To Mitigate Effect of Packet Loss

Packetization

- Structure of Packet
  - Layer at which the Packet is Defined
  - Networking Platform
- MPEG-4
  - Defines Application Layer Packet Structure
  - Application Layer Packet Differs from Transport Layer Packet
- Transport Layer Packet
  - Additional Protocol Headers
    - Reduce Overall Throughput
    - Packetization Overhead Depends on Transport Mechanism
  - Packetization in RTP over IP Network is different from that in ATM over B-ISDN

Layer for Video Coding

- Frame Level
  - Temporal Reference, Picture Header
- GOB Level
  - GOB Number, Quantizer Level for Entire GOB
- MB Level
  - Coded or Non-coded Indication, Optional Quantizer
  - Information about Coded Blocks such as MVs
- Need Frame Header to Decode GOBs
- Need GOB Header to Decode MBs

Video Packetizing

- MBs as the Unit of Fragmentation
  - Not to be Split across Multiple Packets
  - # of Packets could be Packed into a Single Packet
  - Video Packet Loss ➔ Damage of the Corresponding Frame

- To Limit the Propagation of Errors
  - Independent Segment Coding Mode (Annex R of H.263+)
  - Resynchronization on the Occurrence of a Packet Loss
    - Each Packet Includes Picture Header & GOB Information

Packetizing for ATM Cell

- Fixed Size Packet:
  - 5 Byte Header + 48 Byte Payload = 53 Bytes
- Close Packing Scheme
  - Data is Packed in Payload Field Until ATM Cell is Completely Full
  - Some MBs can be Split between Two Adjacent Cells
  - Unique Bit Pattern (11 Bits)
    - To Designate the End of the Variable-Length Section of Data belonging to the Previous Cell
    - Different from GSC (GOB Start Code)
- Loose Packing Scheme
  - ATM Cell Contains Integral Number of MBs

Close & Loose Packing

- Both Scheme
  - Cell Sequence Number & Picture Number
  - Absolute Address for First MBs inside ATM Cell
    - Eliminating Effect of Cell Loss Propagation
- Overhead
  - Close Packing Scheme: 4.125 Bytes
  - Loose Packing Scheme: 2.75 Bytes
- Loose Packing Scheme
  - Less Efficient Use of Bandwidth when ATM Cells Carry the Traffic of Multiple Video Sources
  - Better Error Performance than Close Packing Scheme

Effect of Packet Losses

- Variable Size Packet
  - Longer Packets Lead to Improved Throughput
  - Lower Tolerance to Loss
- Predictive Video Coding Techniques
  - Temporal and Spatial Dependencies of Video Data Contained in Different Packets
  - Disastrous Damage to the Forthcoming Video Data Predicted from the Lost Information in both Time and Space

Overheads of GRPS

- QoS for Video Communications
  - Available Throughput & Channel Coding Scheme
- GPRS
  - Packet Data Traffic Channel(PDTCH) after CS-1/2/3/4
  - Multi-slotting Provides Multiple Data Rate of Single Slotting
- Reducing True Available Throughput at App. Layer
  - Overheads of LLC, SNDC, IP, UDP RTP Layer
  - Overhead : 10%~15% at the RLC Layer

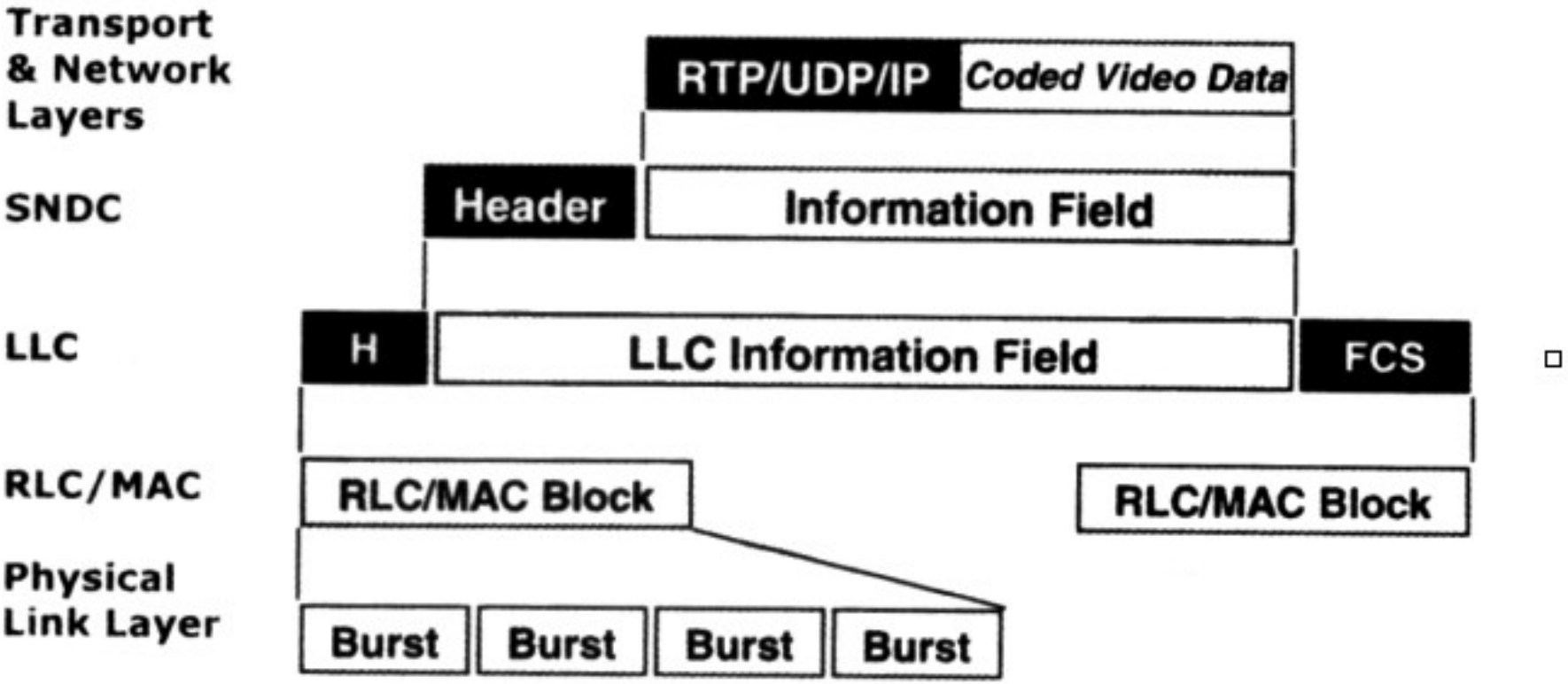
Overheads of EGPRS

- Joint Modulation Coding Scheme
  - MCS-1~MCS-9
  - Radio Block Headers are Encoded Separately from the Data Payload
  - MCS-7,MCS-8,MCS-9
    - Insertion of two RLC/MAC Blocks into a Single Block
- Better Protocol Efficiency than GPRS
- Wide Range in Value of Available Throughput for Video Service
- Choice of CS-TS
  - Activity of Video Source
  - Error Characteristics of Radio Network

Real-Time Transmissions

- Not Exceeding 200 ms
- No Retransmission
- User Datagram Protocol (UDP) instead of TCP
- IP Protocol
  - Jittering Effect
    - Fluctuation of Inter-Arrival time of Packet
  - Packet could be Delivered Out of Sequence
- Real Time Protocol (RTP) based on UDP
  - Payload Type Identification, Time Stamping
  - Sequence Numbering, Delivery Monitoring

Protocol Architecture



Packetization for MPEG-4

- Optimal Trade-off
  - Channel Utilization & Error Robustness
- Synchronization
  - MPEG-4 Stream & Other RTP Payloads
- Monitoring of MPEG-4 Delivery Performance
  - RTCP (Real-Time Control Protocol)
- Packetization
  - Time Varying Nature of the Mobile Channel
  - To Maintain an Optimal Trade-off between Throughput and Error Resilience at Any Instant of Time

Adaptation of RTP Packet Size

- Time Varying Mobile Channel
  - Error Conditions of Network
- Motion Activity of Video Content
  - Lager Packet in Lower Motion Activity
  - Smaller Packet Size in Higher Motion Activity
- MPEG-4 Video Data
  - First Partition: Header and Motion Data
    - Loss of the whole MPEG-4 Packet
  - Second Partition: Texture Data
    - Video Data following the Position of Errors
- Selecting Long Packet Size
  - Less Robust
  - High Channel Utilization

RTP Packet for MPEG-4

- First Packetization
  - Larger Overheads
    - Packet Header
    - First MPEG Partitions
- Second Packetization
  - Higher PSNR
  - Corruption of First Partition
    - Only Loss of the Video Packet
    - Not the Whole RTP Packet

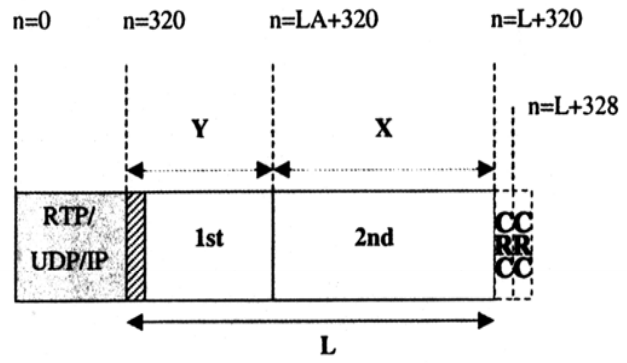


Figure 5.6 One MPEG-4 packet per RTP packet

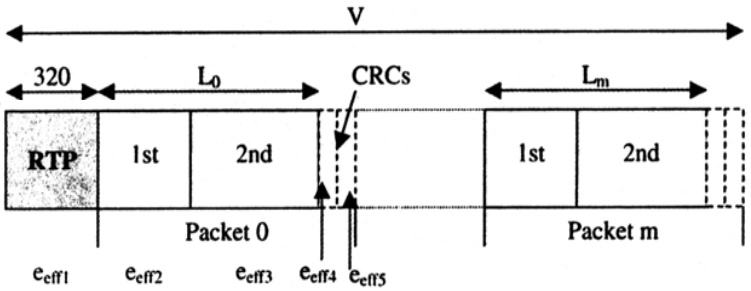


Figure 5.7 Several MPEG-4 packets per RTP packet

Adaptive Quality Control

- Time-Varying Nature of Channel
- Quality Control Tools
  - Mobile Network Condition
  - Type of Video Application
- Various Profiles & Levels
  - Annex X of H.263++
  - Baseline Mode (Profile 0)
  - Error Protection
  - Content Based Adaptive Quality Control

Profile 3

- Profile 0
- Intra Coding (Annex I)
- De-blocking Filter (Annex J)
- Modified Quantization (Annex T)
- Slice Structure Mode (Annex K)
  - Arbitrary Slice Ordering (ASO)
  - Error Resilience Capability
  - Resynchronization Points

Profile 4

- Profile 3
- Data Partitioned Slice Mode (Annex V)
  - Separating MV Data from DCT Coefficient Data within the Same Slice
  - RVLC for MV Information
- Annex W of H.263++
  - Header Repetition
  - Supplemental Enhancement Information
  - Decoder to Recover the Header Information
    - From a Previous Frame in Case of Data Loss or Corruption

Profile 6

- Conversational Video Services for IP Network
  - Real Time Operation: Delay Factor is Important
  - Drop Excessively Delayed Packets
- Optional Modes Enabled by Profile 5
- Several Enhancements for Coding Efficiency

Content-based Packetization

- Higher Motion Activity
  - Motion Data Increase
  - Highly Sensitive to Errors
  - Set Smaller RTP Packet Size
  - Overheads Increase
  - Robustness Improves
- Combined with Application-Layer Resilience Tools

Content-based Resilience Tool

- Adaptive INTRA Refresh (AIR)
  - Annex H of MPEG-4
- Estimation of the Amount of Motion in Video Scene
- # of AIR MBs per Frame
  - Adapted to the Amount of Motion in Video Scene
  - AIR MBs are Intra-coded
- High Motion Activity → Large # of AIR MBs

Prioritized Transport

- Scalability
  - Base Layer: High Priority
  - Enhancement Layer: Low Priority
- Data Partition
  - Motion & Header: High Priority
  - Texture: Low Priority
- Limitation at Application Layer
  - Different GRPS Channel Protection Schemes

Prioritized Transport over Mobile Access Network