Real-Time Communication
Real-Time Systems Design (CS 6414)

Sumanta Pyne

Assistant Professor
Computer Science and Engineering Department
National Institute of Technology Rourkela
pynes@nitrkl.ac.in

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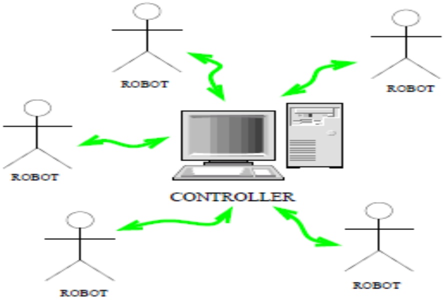
Real-time Communication

- A real-time application can make specific QoS demands from network
  - maximum permissible delay
  - maximum loss rate
- Network connection guarantees requested service quality
- Traditional network protocols designed for best-effort performance
  - Achieves good average performance
  - No attempt to meet individual QoS requirements
- Real-time network applications need
  - deterministic delay
  - predictable performance
- Examples of applications requiring real-time communication
  - Manufacturing Automation - robots communicate with controller
  - Automated Chemical Factory - sense, monitor, control, command
  - Internet-Based Banking - secure, reliable, fast financial transactions
  - Multimedia Multicast - many receivers receive audio and video
  - Internet Telephony - VoIP require firm real-time data transfer

Basic Concepts

- Types of Networks
  - Controller Area Network (CAN)
    - Connects different components of an embedded system
    - End-to-end length usually less than 50 meters
    - Vey low propagation time - like local computer bus
    - Industrial automation, ships, trains, elevators, house, office, agriculture
    - ISO11898 and ISO11519-2
  - Local Area Network (LAN)
    - Privately owned - deployed in building, campus
    - Connects a number of computers within an organization
    - Share data and resources like files, printers, FAX services, etc
    - Operates at 10 Mbps - 1 Gbps (gigabit Ethernet)
  - Packet Switched Networks - Wide Area Networks
    - Divides messages into small units of data called packets
    - Packets with destination address transmitted over the network
    - Destination receive and combine packets into original message
    - Connection oriented and connectionless services
    - Internet - TCP/IP
    - Protocols - ftp, http, VoIP, Telnet, SMTP, POP3

Robot Communication in an Automated Factory



Quality of Service (QoS)

- QoS parameters
  - bandwidth
  - maximum transmission delay
  - delay jitter
  - loss rate
  - blocking probability
- Delay
  - Packet delivery exceeds delay bound
    - Failure - hard real-time applications
    - Discarded by receiver - firm real-time applications
    - Degrades performance - soft real-time applications
  - Packet delay has three parts
    - Propagation delay - depends on distance packet travels and medium
    - Transmission delay - per-hop transmission delay, size/link bandwidth
    - Queuing delay - depends on packet waiting in queue before transmission
  - During congestion queuing delay far greater than other components

Delay Jitter

- Maximum variation in delay experienced by messages
- Ex: (min,max) end-to-end packet delay (1 ms, 10 ms), has 9 ms jitter
- LAN using collision-based protocol, jitter due to variation in n/w load
- Packet-switched n/w - jitter due tonqueuing delays at different nodes
- Unacceptable in hard real-time applications - system failure
- Undesirable in firm real-time applications
- In video conferencing - jitter show a picture frame for certain time
- Buffers at receiver-end used to control jitter in packet-switched n/w
- Packets buffered long enough, allow slowest packets to arrive in time
- Process packets in correct sequence
- *Buffer size* = *peak rate* x *delay jitter*
- A video source transmitting 30 frames per second to a receiver
- Each frame contains 2 mb of data, jitter of network is 1 second
- *Buffer size* = 2 mb/frame x 30 frames/sec x 1 sec = 60 mb
- A tight jitter bound reduces buffering requirement at receiver end

Bandwidth

- Rate at which a connection serviced by network
- Determines an application's maximum throughput
- Bounds on end-to-end delay
- Sufficient bandwidth to sustain required throughput of an application
- Transmission delay of message inversely proportional to bandwidth
- CAN and LAN - delay of message depends on bandwidth
- Packet-switched n/w - transmission delay $<<$ queuing delay

Loss Rate

- Percentage of all transmitted packet lost during transmission
  - Delay-bound violation
  - Delay jitter-bound violation
  - Buffer overflow
  - Data corruption
- Data corruption insignificant in fibre optic, large in wireless
- Different applications are sensitive to loss rate to different extents
- Process control applications require zero loss rate
- Multimedia applications can tolerate a certain amount of data loss

Blocking Probability

- Probability of a new connection being rejected
- By admission control mechanism of a network
- A call may be rejected under heavy load situations
- Rejection due to lack of resources such as bandwidth

QoS parameters in real-time systems

- Audio-video broadcasting - strict jitter bound, insensitive to delay
- Sensor data processing in fly-by-wire aircraft - delay sensitive sub msec
- File transfer, E-mail, remote login - non real-time applications
- Non real-time applications QoS - reliable, loss-free, no data corruption
- Multimedia applications not much affected by delay jitter and loss rate
- Video sensitive to packet losses, show glitches on display screen
- Voice quality degrades rapidly with loss rate and jitter
- Voice packets are deliberately kept small
    - minimize packetization delays
    - limit effect packet of losses
- 48-byte cell size for ATM n/w for benefit of voice applications

Traffic Categorization

- A network can guarantee required QoS
- CBR Traffic
  - Arises due to Constant Bit Rate generation by a source
  - Data generation and transmission for hard real-time applications
  - Periodic sensor data - fixed sized messages over fixed intervals
- VBR Traffic
  - Different rates of data transmission at different times
  - Fixed sized packets with deterministic spacing and idle Periodic
  - In compressed audio signals generated by speech signals
    - To reduce size of voice traffic, no data during silence periods
    - Silence suppression - significantly reduces bandwidth requirements
    - Achieved at cost of reconstruction of original traffic at receiver
  - Compressed video signals with periodic variable sized data
    - Digitized video data highly redundant
    - Compressed using algorithms like MPEG before transmission
    - Periodic submission of variable sized packets to network
  - VBR makes better use of available bandwidth for video-audio signals

Sporadic Traffic

- Special type of variable sized packet transmissions
- Packets generated in bursts followed by long silence periods
- A special type of VBR traffic, occurs in special circumstances
- Traffic - command, control, alarm messages at exception conditions
- A fire alarm
  - Fire condition is detected
  - Large number of alarm, command and response messages generated
  - Sudden peak load due to alarms - alarm avalanche

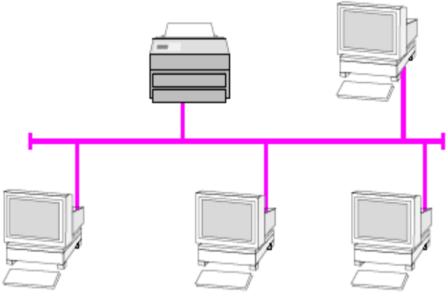
Real-time Communication in a LAN

- Hard real-time applications span small geographical areas
- Automated manufacturing, industrial process control
- High speed data acquisition
- Local Area Networks (LANs) preferred choice
- Circuit-switched network not suitable for bursty real-time traffic
- Circuit-switched application specific fixed bandwidth at set-up time
- LAN has single shared channel, only one node transmit at any time
- Exact transmission duration of node by MAC layer protocols
  - access arbitration - when a node can use channel
  - transmission control - how long a node can use channel

LAN Architectures

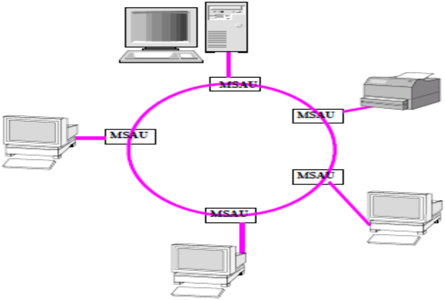
- Bus Architectures
  - Nodes are connected to network cable using T-shaped connectors
  - Terminating points at each end of network cable
  - Single shared channel (bus) for which transmitting nodes contend
  - Nodes communicate using broadcasting
  - Carrier Sense Multiple Access with Collision Detection (CSMA/CD)
    - When two or more transmit packets simultaneously
    - Transmissions overlap in time
    - Resulting signal gets garbled - collision occurs
    - Collision entails retransmission of corrupted data
    - Any node at any time sense whether channel is idle
    - A node transmits a packet only if it senses idle channel
    - If collision occurs, nodes immediately stop transmitting
    - Collision resolution protocol - Binary Exponential Back-Off (BEB)

Bus Architectures



- Propagation delay of network $\propto$ probability of packet collisions
- Commonly used Ethernet LAN - CSMA/CD with BEB
  - Ubiquity, high speed, simplicity, low cost
  - At high load situations #collisions per unit time increase rapidly
  - Throughput drops, delay rises - concern for real-time applications

Ring Architectures



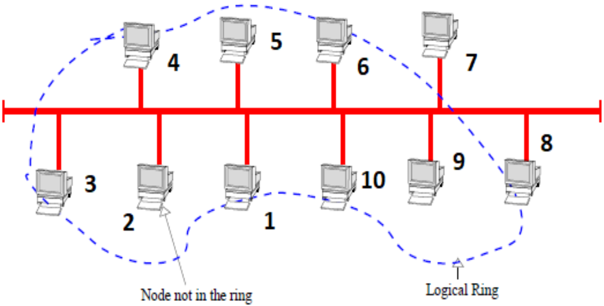
- Nodes are placed along ring
- Multistation Access Unit (MSAU) connects node to ring
- Nodes transmit in turn for a predetermined period of time
- More predictable packet transmission delay, made smaller
- Preferred in real-time applications

Ring Architectures

- Reliability issue - any break in ring brings down whole network
- Poor fit to linear topology found in applications like assembly lines
- Alternative with advantages of bus and ring architectures - token bus
  - Stations on bus arranged logically in ring
  - Each station knows address of stations to its left and right
  - Initialization - highest numbered station gets chance to transmit
  - Station transmits for a predetermined duration
  - Then permits immediate neighbour sending a control frame *token*
  - Token propagates around logical ring
  - At any time token holder is permitted to transmit packets
  - Only one station holds token at a time - collisions cannot occur

Logical Ring in a Token Bus

- Physical order of stations need not be same as order in logical ring
- A cable is a broadcast medium
- Each station receive every frame transmitted
- Discard frames not addressed to the node
- After time slot exhausts, a node hand overs token to logical neighbour



- Stations 2, 5, 7 and 9 not on ring
- MAC protocol support add/remove stations to/from logical ring

Soft Real-time Communication in LAN

- Not expected to provide any absolute QoS guarantee
- Ensure prioritized treatment for real-time messages
- Minimum message deadline miss ratio
- Statistical guarantee on delay bounds of soft real-time messages
- Both soft real-time and non real-time messages
- Soft real-time traffic generated by CBR and VBR sources
- Soft real-time message rates $<<$ channel capacity
- Soft real-time & non real-time messages arrive aperiodically in bursts
- Difficult to sustain guarantees to real-time traffic
- Burst smoothing to meet statistical guarantees on deadline bounds

A Fixed-Rate Traffic Smoothing Algorithm

- Kweon and Shin fixed-rate traffic smoothing
- Considering limits on transmission capacity of a network
- Input limit for each node derived from limits on transmission capacity
- Traffic smoother between MAC and TCP/IP layers
- Smooth non-real time stream, meet real-time message guarantees
- Use a leaky bucket algorithm - Credit Bucket Depth (CBD)
- Parameters - CBD and Refresh Period (RP), fixed statistically
- CBD - max #credits added to bucket at every refresh
- RP - period with which bucket filled up with new credits
- CBD/RP - average guaranteed throughput for non real-time messages
- Current Network Share (CNS)- #credits present in bucket at any time

A Fixed-Rate Traffic Smoothing Algorithm

- Smoothing mechanism when a non-real time message arrives
- If CNS$>$0 and CNS$<$#message_bytes
  - Borrow credits, CNS $=$ CNS - #message_bytes
  - Send message for transmission
- Otherwise, hold message in buffer until CNS > 0
- At every refresh, CNS $=$ min(CNS $+$ CBD, CBD)
- CBD credits replenished, limits CNS not to exceed CBD
- Not flexible - network-wide input rate fixed
- Non real-time source transmission rate reduced as #nodes increases
- Increase in #stations, decrease in non real-time traffic throughput
- Based on worst-case total real-time traffic arrival rate

Adaptive Traffic Smoothing

- Kweon and Shin adaptive traffic smoother
- Reasonable throughput for non real-time messages from VBR sources
- Transmission rate adapt itself to load conditions of network
- Nodes increase transmission limits if network utilization is low
- Nodes decrease transmission limits when network utilization is high
- Meet delay requirement of real-time packets
- Improve average throughput of real-time packets
- Network utilization measured as #collisions per unit time
- Collision - immediate empty of credit bucket
  - Suspension of non-real-time packets, except packets under transmission
- Increases real-time packet delivery of other nodes within delay bounds
- Prevents delays due to bursts of non real-time traffic from a node

Adaptive Traffic Smoothing

- Station i/p limit CBD/RP adapted by changing CBD or RP
- No recent collisions - periodic increase in CBD/RP, RP decreased
- Collision detected - all credit buckets completed, RP is doubled
- $RP\_min \leq RP \leq RP\_max$ is fixed and predefined
- Real-time message deadline miss ratios in acceptable range
- Achieved in arbitrary non real-time message arrival rates
- Fails to provide deterministic real-time communication
- Not suitable for hard real-time communication

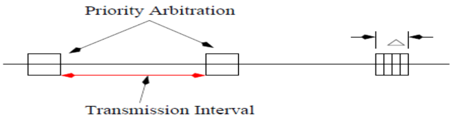
Hard Real-time Communication in LAN

- Involve transmission of CBR traffic - periodic sensor signals
- Network utilization kept low - predictibity more importance
- Soft and non real-time traffic, when no real-time messages
- Protocol classes - Global priority, Bounded access, Calendar based
- Global Priority Protocols
  - Each message in network assigned a priority value
  - MAC layer protocols serves highest priority message at any time
  - Use of RMA or EDF for scheduling messages
  - Unlike tasks packet transmission cannot be stopped halfway
  - Instantaneous determination highest priority message not possible
  - RMA/EDF restrict achievable schedulable channel utilization very low
- Bounded Access Scheduling
  - RT guaranteed messages, bounded channel access time for each node
  - Ensures bounded waiting time for a packet before transmission
  - Fix time for which a node is permitted to transmit message
  - Local schedulable algorithm - nodes determine order of queued packets

Hard Real-time Communication in LAN

- Calendar based Scheduling
  - A calendar indicates which node transmit during which time period
  - A copy of calendar maintained by every node
  - Traffic sources broadcast to reserve packet transmission time interval
  - When a node transmits for which there is no reservation
    - it finds free slot by consulting its local copy of calendar
    - reserves required time interval by broadcasting reservation information
  - Simple
  - Efficient when all messages in system are periodic and predictable

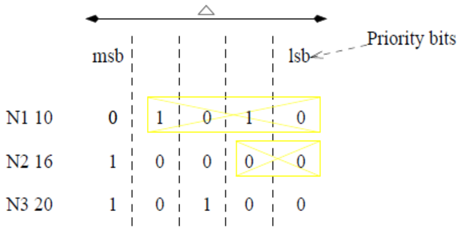
Global Priority-Based Scheduling - Countdown Protocol



- Time line divided into fixed sized intervals called slots
- At every slot, priority arbitration determines highes priority message
- Arbitration completes, allow highest priority message transmission
- Priority arbitration
  - Over each slot every has a pending message
  - Transmits priority value of its highest priority pending message as msb
  - Simultaneous transmission follows *or* logic
  - A node transmits 0 receives 1 - knows at least one with higher priority
  - Drops out of contention
  - Node transmits last, concludes no others have higher priority message

Countdown Protocol

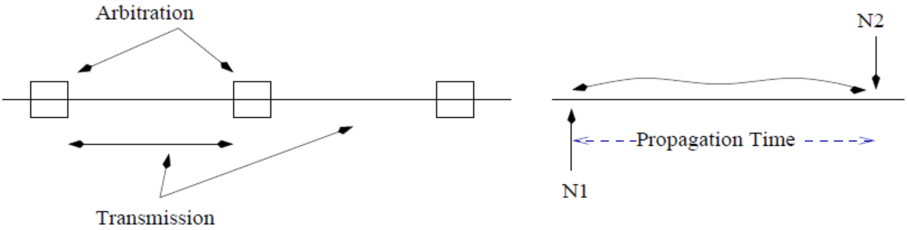
- Protocol efficiency depends on slot size
- Slot duration - end-to-end propagation delay of medium
- Smaller slot size - priority arbitration not work, cannot detect collision
- Larger slot size - increase in channel idle time during every slot
- Message priorites at nodes - N1 10, N2 16, N3 20
- N3 has highest priority message, begins transmission



Global Priority-Based Scheduling - Virtual Time Protocol

- A node uses channel state to reason about pending packets at others
- Higher priority of max_priority_message, lower is wait interval of node
- Assigns priority to nodes
- Node priority - highest priority message it has
- At expiration of wait time, node senses channel status
- If channel busy
  - imply higher priority message being transmitted
  - need to wait until an idle period
- Otherwise, it starts to transmit
- What difference in wait times if |mess_prior(Ni)-mess_prior(Nj)|=1 ?
- Due to propagation delay, no instant detection when another node starts transmit during arbitration

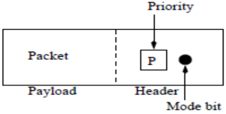
Virtual Time Protocol



- Let N1 and N2 have priorites differing by one
- After N1 starts transmitting
- If N2 waits for any time shorter than propagation time
- N2 cannot detect transmission of N1
- N2 starts transmission, collision occurs
- Results incorrect priority arbitration
- Propagation time bounds on wait time difference prevents this
- Longer wait time, channel idle times, channel underutilization

IEEE 802.5

- A priority-based token ring protocol
- Header - reservation and mode fields
- Alternative token modes - reservation and free modes



- Node determine token mode from mode bit in token header
- Messages transmitted split into frames - token payload
- Assign message priorites - recorded in reservation field
- Higher priority message than being transmitted, node registers priority
- Token returns sender obeserves reservation by other nodes, releases it
- Free token in ring seized by reserving node to start transmission

Theorem 1

The minimum time required to complete transmission of a frame using IEEE 802.5 protocol is max(F,θ), where F is the frame transmission time and θ is the propagation time.

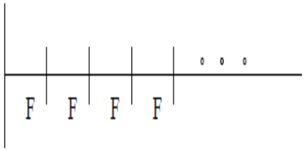
- PROOF: A node does not start transmitting the next frame until:
  - It has completed transmission of the last bit of the frame
  - It has received the header of the transmitted token back
- First activity takes F time units
- Second activity takes θ time units
- Transmission time is higher of two values, max(F,θ)
- Minimum time before which transmission of next packet cannot start
- Time bound - max(F,θ)

Theorem 2

A higher priority packet might undergo inversion for atmost $2\times$max(F,θ) time units.

- PROOF: A higher priority message undergoes inversion, until:
  - The reservation mode completes
  - The node receives the token in free mode
- Each step takes max(F,θ) time units to complete
- Total time a higher priority packet may have to wait is $2\times$max(F,θ)

Window-Based Protocol

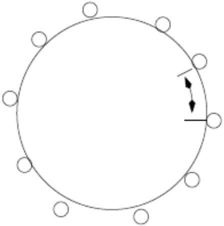


- Time is divided into frames
- Every node maintains current transmission window
- Defined priority values (low,high)
- A node having message in window with priority $\in$ (low,high), transmit
- At start of a frame more than one node eligible, start transmitting
- Would result collision
- On a collision every node does low=low+1
- On a free frame every node does low=low-1

Calendar-Based Protocol

- Example - dynamic reservation technique
- Each node maintains calendar with access time reservations for all
- Reservations of guaranteed messages
- A message with no reservation arrives a node
- Node first determines a free slot consulting local calendar
- Attempts to reserve a suitable free future time interval
- Broadcasts a control message, other nodes update local calendar
- Efficient for deterministic periodic messages
- No overhead in priority arbitration
- Difficult for aperiodic and sporadic messages
- Used in very simple networks - CANs

Bounded Access Protocols for LANs - IEEE 802.4



- Timed token protocol, used in token ring and token bus
- A node can transmit when it holds token
- Real-time guarantee to messages by bounded token hold time
- Incorporated in Fibre Distributed Data Interface (FDDI)
- Design parameter - Target Token Rotation Time (TTRT)
- TTRT - time between two consecutive visits to a node
- At initialization - TTRT specified on message characteristics
- RT messages - periodic, synchronous; non-RT - asynchronous
- Each node allocated a portion of TTRT - synchronous bandwidth
- Depends on timing characteristics sychronous messages

IEEE 802.4

- Let sychronous bandwidth of a node Ni be Hi time units
- Ni can transmit its synchronous message for atmost Hi duration
- SN be set of all nodes in the network
- TTRT $= \theta + \sum$ Hi, $\forall$ Ni $\in$ SN
- θ - propagation delay, Hi - token holding time at Ni
- Ni receives token, transmits all synchronous traffic in bounded time
- Transmit asynchronous traffic if Ni's token departure time $<$ TTRT
- Transmit non-RT messages only when token arrives earlier than TTRT
- Asynchronous frame transmission duration - asynchronous overrun
- Asynch. overrun reduces available bandwidth, delays token arrival
- No asynch. overrun - expected interval successive token visits TTRT
- Asynch. overrun - worst case time successive token visits 2$\times$TTRT

IEEE 802.4

- Ni has no message to transmit, token arrives Ni+1 TTRT time early
- Ni transmits asynchronous messages for TTRT time units
- Ni & all subsequent nodes fully use assigned sync. message time slots
- Token arrive Ni after 2×TTRT time units since last visit
- N/W synch. mode - consecutive token arrival interval limit TTRT
- Δ - shortest deadline of Ni's message
- Worst case successive token arrival interval for Ni is 2×TTRT
- To meet Δ, during n/w initialization TTRT set to value $< \frac{\Delta}{2}$
- TTRT $< \frac{\Delta}{2}$ - increases overhead, low network utilization
- TTRT $> \frac{\Delta}{2}$ - real-time messages miss deadline

IEEE 802.4

- Token holding time of a node is synchronous bandwidth alloted to it
- Ni receives token, starts timer set to synchronous bandwidth (Hi)
- Releases token upon timer expiry
- Hi $=$ (TTRT-θ)$\times$(Ci/Ti)/($\sum$Ci/Ti)
- Ci - message size in bits
- Ti - Ni's transmission interval
- Ci/Ti - channel utilization of Ni

Example 1

- Suppose a network designed using 802.4 protocol has three nodes. Node N_1 needs to transmit 1 MB of data every 300 msec. Node N_2 needs to transmit 1.2 MB of data every 500 msec. Node N_3 needs to transmit 2 MB of data every 200 msec. Select a suitable TTRT for the network and compute the token holding time for each node. Ignore the propagation time.
- Shortest deadline among all messages, $\Delta = 200$ msec
- TTRT$= \frac{\Delta}{2} = \frac{200}{2} = 100$ msec
- Channel utilization - $C_1/T_1 = \frac{1\times 8}{300}$ Mb/msec
- $C_2/T_2 = \frac{1.2\times 8}{500}$ Mb/msec
- $C_3/T_3 = \frac{2\times 8}{200}$ Mb/msec
- $C_1/T_1 + C_2/T_2 + C_3/T_3 = \frac{1\times 8}{300} + \frac{1.2\times 8}{500} + \frac{2\times 8}{200} = \frac{377.6}{3000}$ Mb/msec
- Token holding time - $H_1 = 100 \times \frac{8}{300} \times \frac{3000}{377.6} = 21.18$ msec
- $H_2 = 100 \times \frac{9.6}{500} \times \frac{3000}{377.6} = 15.25$ msec
- $H_3 = 100 \times \frac{16}{200} \times \frac{3000}{377.6} = 63.56$ msec

RETHER

- Real-time ETHERnet enhances TCP/IP for performance guarantees
- Without modifying existing Ethernet hardware
- N/W transmission modes - CSMA/CD mode and RETHER mode
- Real-time transmissions in RETHER mode - token passing scheme
- CSMA/CD mode when all real-time sessions terminate
- In absence of RT messages nodes compete usign CSMA/CD
- A node receives RT request from local application
- Broadcasts a Switch-to-RETHER message, if not in RETHER mode
- Every node responds sets to RETHER mode, acknowledges initiator
- Transmitting node waits for ongoing packet transmission to complete
- Acknowledges initiator, to RETHER mode and no back-off data
- Initiator receives all acknowledgements, creates and circulates token
- Completes switch to RETHER mode

RETHER

- If more than one initiator tries RETHER-mode at same time
- Each node acknowledges switch messages to min(ID) initiator
- An initiator Y acknowledges another initiator X if X.ID $<$ Y.ID
- Acknowledge loss - fixed retries, concludes node dead, convey others
- RETHER mode uses IEEE 802.4 for bandwidth guarantees
- Any time one real-time request per node, reqd. bandwidth in TTRT
- Maximum token holding time based on amount of data and TTRT
- This information for bandwidth reservation in each session
- Token circulates among Real-time Set (RTS) and NRTS nodes
- Nodes with bandwidth reservation belongs to RTS, rest NRTS
- Each rotation, token visits all nodes in RTS in order
- RTS node receives token, sends RT data, passes token to neighbour
- Last node in RTS passes token to NRTS, if no reservations

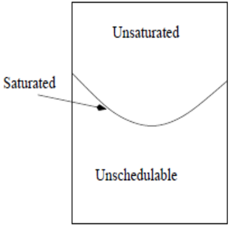
RETHER

- TTRT - mean token rotation time
- MTHTi - mean token holding time for node Ni
- Token field TimeToDeadline = TTRT - $\sum$ MTHTi, i$\in$RTS
- TimeToDeadline>0, some NTR messages transmitted after all RT
- An NRT receives token, determines sufficient time send RT message
- If sufficient time, sends packet, decrements TimeToDeadline
- Then passes token to next node in NRTS
- If no time to send packet, informs last RTS node
- To be first NRTS node to get token in next round
- Passes token to first node in RTSD
- New RT request accept - admission control procedure in each node
- Request admitted if, $\sum$MTHTi+MTHT_new+TBNRT$\leq$TTRT,i$\in$RTS
- TBNRT - bandwidth reserved for NRTS
- MTHT_new - bandwidth required by new node in RTSD
- Node terminate RT connection, removes RTS information from token

Switched Real-Time Ethernet

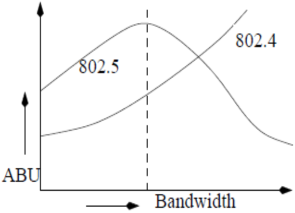
- Ethernet networks use more switches instead of hubs or coaxial cables
- Hubs and switches form star-like network
- Hub passes incoming traffic from an port to all other ports
- Switches send through specific port based on n/w topology knowledge
- In star topology, a dedicated cable in each direction, no collisions
- Switches - no guarantees to which one will be sent first
- Switch deploy bandwidth reservation to solve this

Performance Comparision of IEEE 802.5 and IEEE 802.4



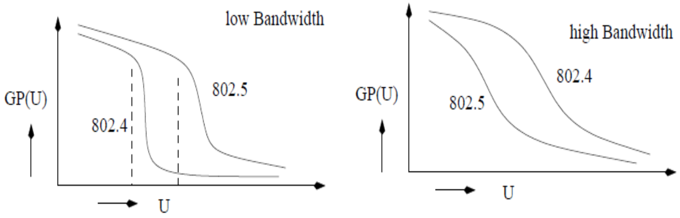
- Schedulabilty classes of real-time messages
- Unsaturated Schedulable
  - Message sets schedulable even when a message size slightly increased
  - Result low channel utilization
- Saturated Schedulable
  - Message sets unschedulable for any increase in a message size
  - Messages miss deadlines
- Unschedulable
  - Message sets unschedulable for at least some messages miss deadline

Utilization metrics - Absolute Breakdown Utilization (ABU)



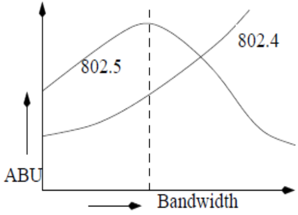
- Indicates expected value of utilization of a message set S, U(S)
- Average utilization of all messages in saturated set
- $U(S)=\sum Ci/Ti,\ i \in S$
- Ci is size of message i, Ti is period of message i
- $ABU=\sum U(S)/|Sat|$
- Sat is set of all saturated message sets
- Average case traffic accommodation with no message missing deadline

Guarantee Probability (GP(U))



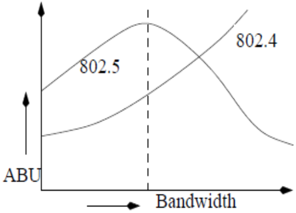
- Probability of meeting all deadlines of a message set with utilization U
- GP(U) $\rightarrow$ 1 for U $<$ ABU, GP(U) $\rightarrow$ 0 for U $>$ ABU
- Priority-based protocols work better in low bandwidth networks
- Priority inversion by RR in bounded access/timed token protocol
- Impact messages with short deadlines
- At high bandwidth, priority-driven protocol, low U, arbitration
- Bounded access protocols work better in higher bandwidth networks

Anomaly in priority-based IEEE 802.5 protocol



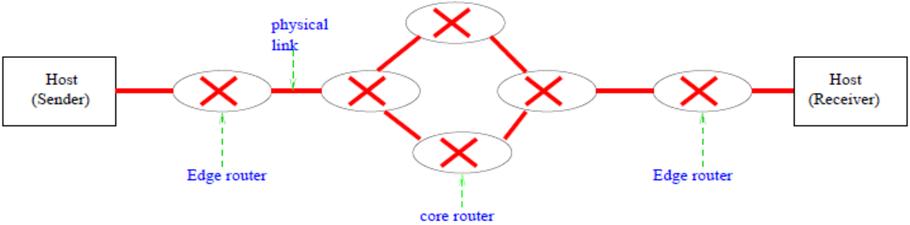
- Performance of priority-driven IEEE 802.5 protocol initially improves
- Starts to drop off beyond certain point
- Decrease in transmission time causes F$< \theta$
- Before releasing a new token, node waits for token to returns
- Even after node has completed transmission of a frame
- Effective frame transmission time is θ
- Fraction of wasted bandwidth is (θ-F)/F

Anomaly in priority-based IEEE 802.5 protocol



- θ is independent of bandwidth, and is constant
- Bandwidth wastage $\propto$ bandwidth
- Performance deteriorates after a certain bandwidth at θ=F
- Timed token protocol does not exhibit this anomaly
- Since a node transmit continuously during token hold duration

Real-time Communication in Packet Switched Networks



- Packet switching - wide area networks
- Internet provides best effort service to applications
- No guarantee of timeliness or actual delivery
- QoS - routing, resource reservation, traffic shaping, policing
- RT communication requires contract between certain senders and ISP
- Data packets must be classified for RT services before transmission
- At connection set-up sender provides traffic and QoS specifications
- ISP uses specifications to accept or refuse client request

Real-time Communication in Packet Switched Networks

- ISP checks availability of resources to satisfy QoS on possible routes
- Accept communication request, guarantees required QoS will be met
- Guarantee valid if client-generated traffic is within specificied bounds
- Traffic bound violated - network drop packets, reshape traffic
- Policing - dropping packets of misbehaving traffic sources
- Need to perform within specifications, irrespective of other clients
- Performance not guaranteed, if not protected from malicious sources
- Avoid this - control i/p rates per connection or scheduling at switches
- Scheduling algorithms - Fair Queuing, Weighted Fair Queuing, etc.
- Exists as part of Qos architectures for internet - IntServ, DiffServ

Traffic Characterization

- Sender provides traffic and QoS specifications
- Network checks for adequate resources to meed required QoS
- Part of an admission control procedure
- A model of characteristics of data generated by a source to transmit
- Traffic characterized by bounding data volume generated per unit time
- Accurate traffic bound, bounds n/w resources reserved to meet QoS
- At connection set-up, traffic sources specify traffic characteristics
- Avoid this - control i/p rates per connection or scheduling at switches
- Resources reserved on traffic characteristics & service requirements
- Traffic characterization models

(Xmin, Smax) model

- Bounds a traffic source with a peak rate
- A connection satisfies this model
- If inter-arrival times between two packets is always $<$ Xmin
- and, size of largest packet $\leq$ Smax
- Identical peak and average traffic rates, Smax/Xmin
- Provides tight bound for CBR traffic
- Cannot accurately specify bursty traffic, models worst-case traffic
- For bursty traffic, leads to very conservative resource reservation
- Low utilization of reserved resources
- Inefficient functioning of network

(r, T) model

- Time axis is divided into intervals of length T
- Each interval is called a frame
- Source satisfies model, if it generates no more than traffic rate r
- T bits of traffic in any interval T
- r is upper bound on average rate at which traffic
- generated over averaging interval T
- Many ways similar to (Xmin, Smax) model
- Provide tighter bound for CBR traffic
- Bursty traffic, very conservative resource reservation
- Low utilization of reserved resources

(Xmin, Xavg, Smax, I) model

- Xmin - minimum inter-arrival time between two packets
- Smax - maximum packet size
- I - an interval over which observations are valid
- Xavg - average inter-arrival time of packets over an interval size I
- A connection satisfies this model
- if it satisfies (Xmin, Smax) model, and
- during any interval length I avg inter-arrival time of packets is Xavg
- Peak traffic rate - Xmin/Smax, average traffic rate - Smax/Xavg
- Bounds peak and average traffic rates
- More accurate for VBR traffic than (Xmin, Smax) or (r, T) model

(σ, ρ) model

- σ - maximum burst size
- ρ - long term average rate of traffic source
- $\rho =$ number packets generated over a large duration / size of duration
- A connection satisfies this model
- if during any interval of length t, number of bits generated $< \sigma + \rho \times t$
- Satisfactory for bursty traffic sources

Multiple rate bounding

- Characterize bursty traffic sources with sufficient accuracy
- Bounding traffic over multiple averaging intervals
- A traffic satisfy $\{(r_1, T_1), (r_2, T_2), \ldots\}$,
- if $T_1 < T_2 < T_3 < \ldots$, and over any interval I,
- number of bits generated is bounded by $ri \times Ti$ if $Ti\text{-}1 < I < Ti$
- As averaging Ti interval gets longer, a source bounded by rate$<$peak
- closer to its long term average rate

Thank you