Freie Universität Berlin

Embedded Internet and the Internet of Things WS 12/13

7. Transport Layer

Prof. Dr. Mesut Güneş
Distributed, embedded Systems (DES)
Institute of Computer Science
Freie Universität Berlin

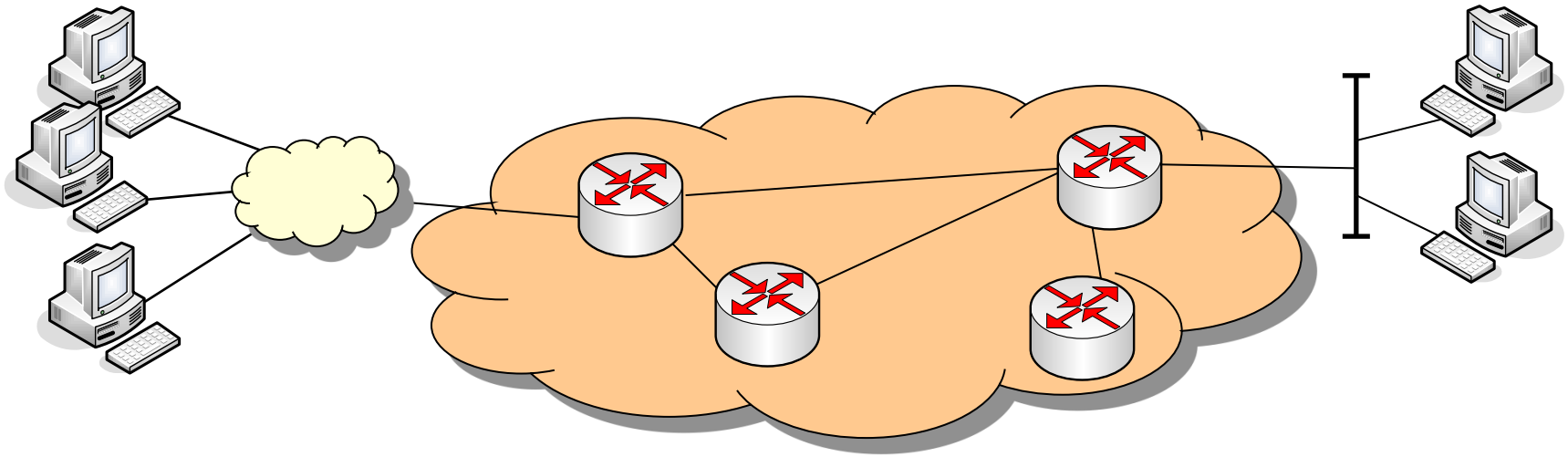
Overview

- Introduction
- Refresher: TCP
- RMST
- PSFQ
- ESRT
- STCP
- Summary

Introduction

Introduction

- Transport protocols are used to …
  - orderly transmission
  - eliminate or mitigate congestion
  - reduce packet loss
  - provide fairness in bandwidth allocation
  - guarantee end-to-end reliability



Introduction

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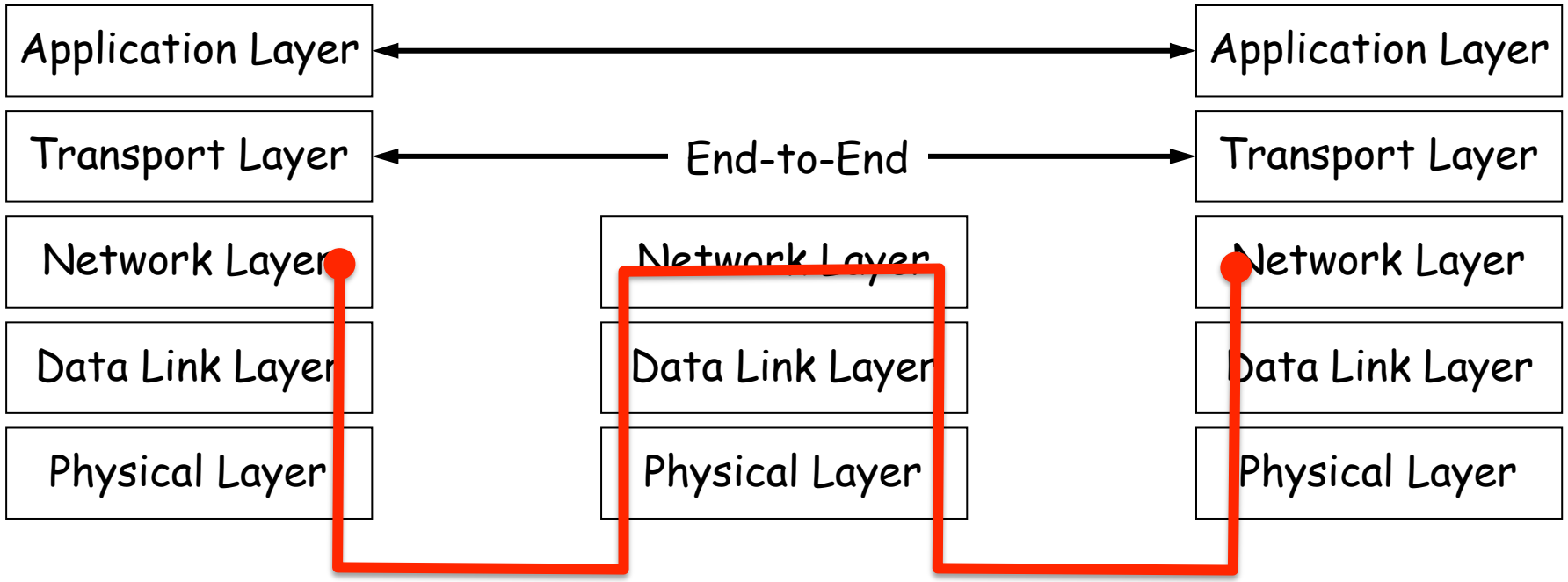
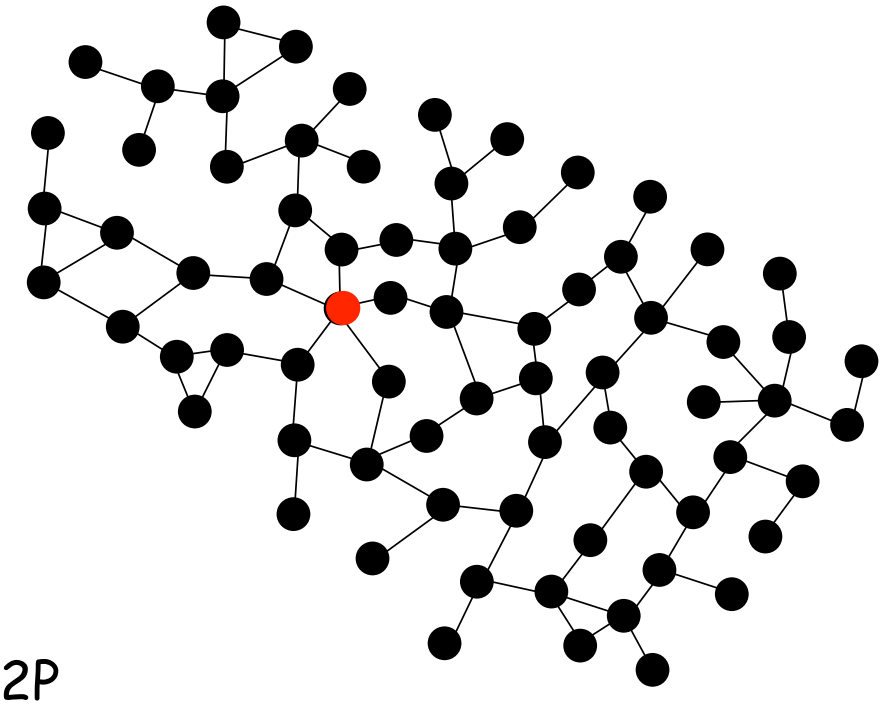


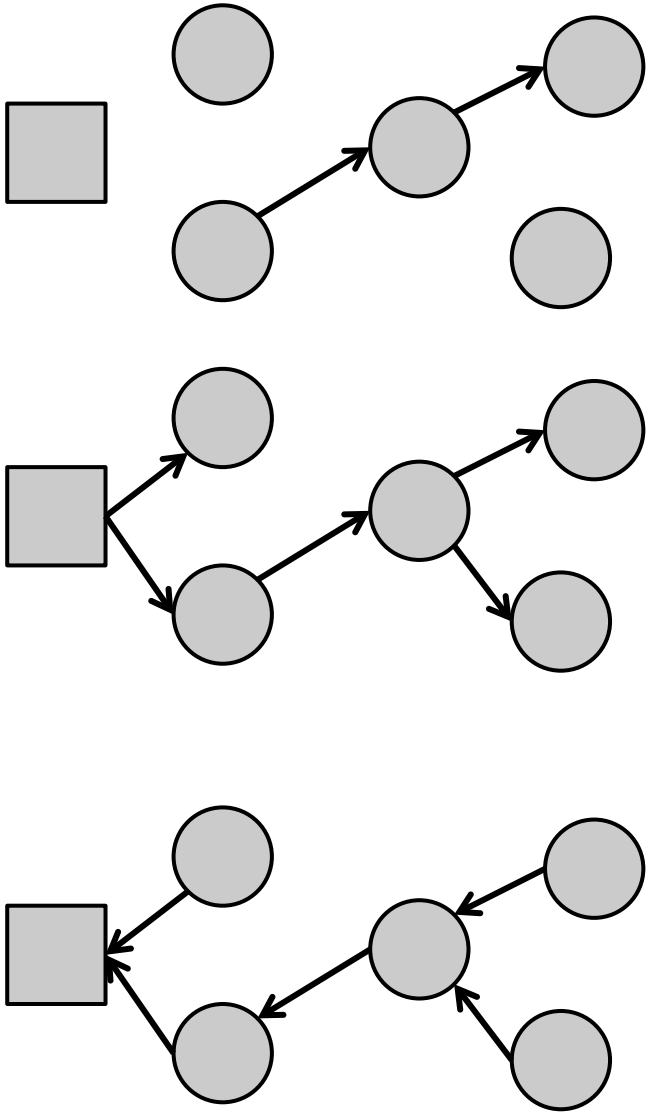
Distinct features of WSNs

- Topology
  - multi-hop star topology
- Diverse applications
  - event-driven
  - continuous delivery
  - query-driven delivery
  - hybrid delivery
- Traffic characteristic
  - Mainly from sensors to sink -> MP2P
- Resource constraint
  - Limited resources (CPU, Memory, Bandwidth, …)
- Small message size
  - No segmentation of messages



Typical traffic structure

- Sensor -> Actor
  - Unicast
  - Challenge: Reliability
- Sink -> Sensor/Actor
  - Data
    - Control and signaling
    - Code update -> reprogramming
  - Multicast/Broadcast
  - Challenge: Reliability
- Sensor -> Sink
  - Data
    - Sensor measurements/events
  - Concast
  - Challenge: Reliability + Congestions



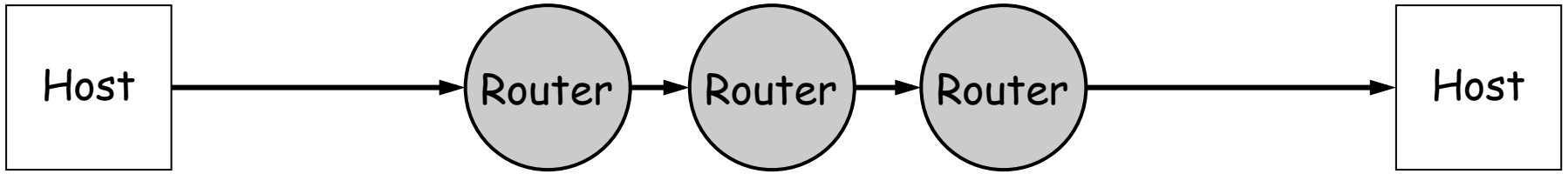
Design criteria for transport protocols

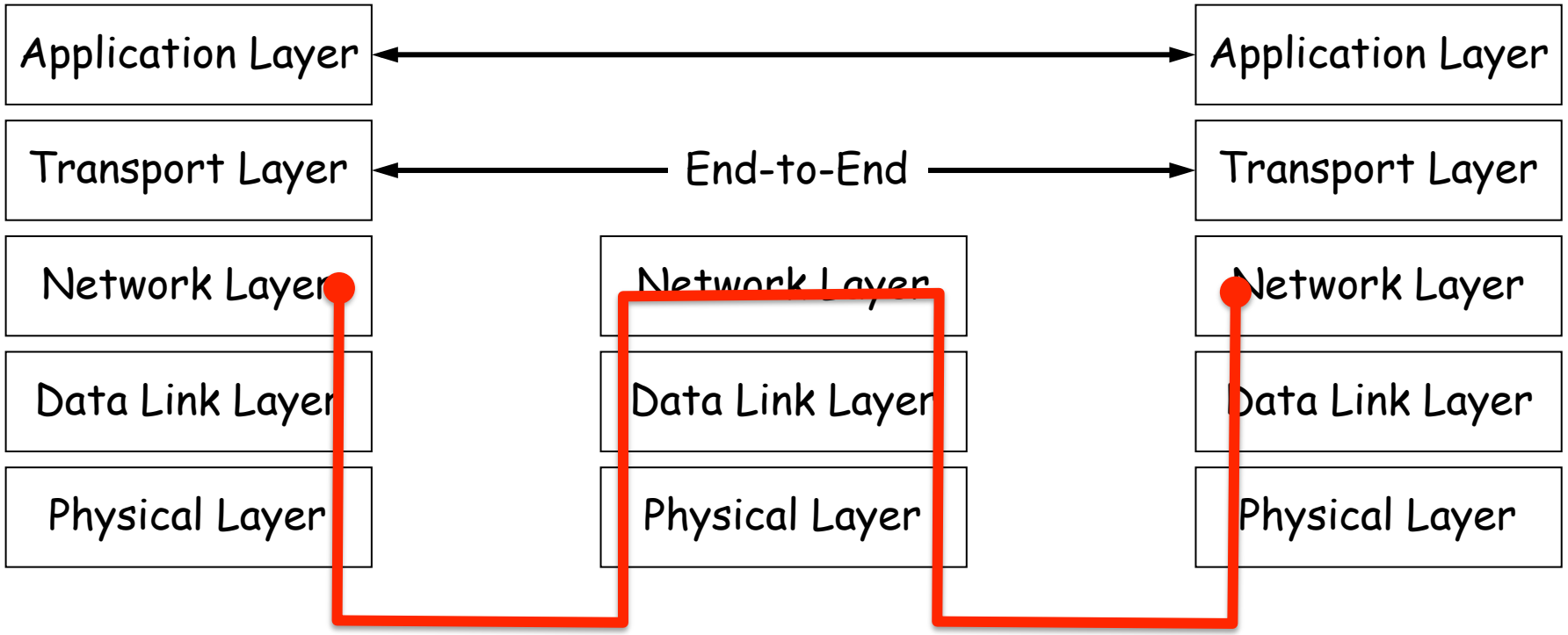

[Wang]

- Performance metrics
- Congestion control
- Loss recovery

Design criteria for transport protocols

[Wang]





Performance metrics

- Energy efficiency -> maximize system lifetime
- Reliability
  - Packet reliability -> consider every packet
  - Event reliability -> not all packets need reliability, but the event
- QoS metrics -> as in traditional networks
  - Bandwidth
  - Delay
  - Packet loss rate
- Fairness -> distance of the nodes to sink

Congestion control

- Two main causes for congestion
  - Packet arrival rate exceeding packet service rate
    - More likely at nodes close to the sink
  - Link level performance such as contention, interference and bit error

- Three methods involved
  - Congestion detection
  - Congestion notification
  - Rate adjustment

Congestion control

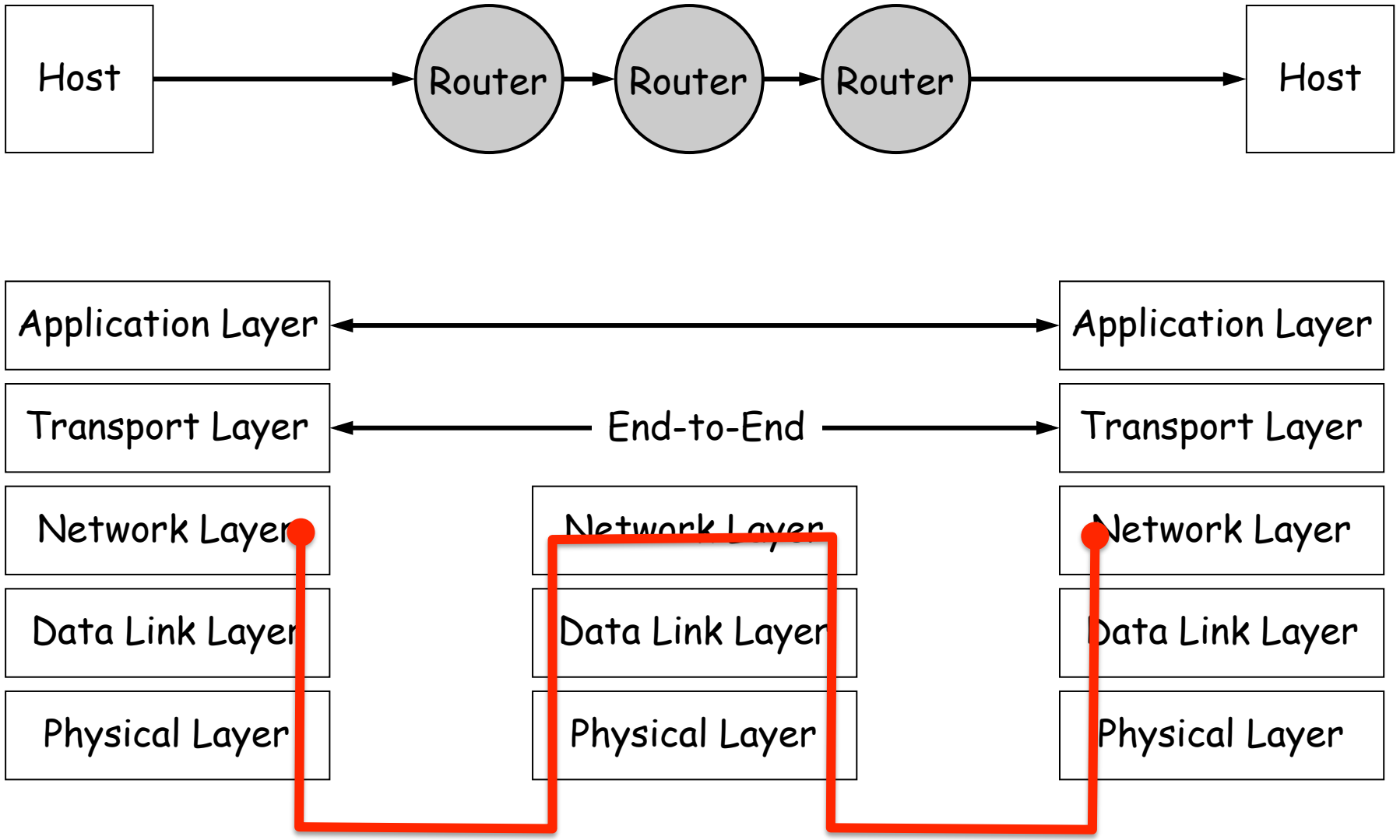
- Congestion detection
  - queue length
  - packet service time
  - ratio between packet service time and packet inter-arrival time at an intermediate node
  - channel loading
- Congestion notification -> inform the nodes
  - explicit notification
  - implicit notification
- Rate adjustment
  - Nodes adjust their traffic sending rate
  - AIMD (Additive Increase Multiplicative Decrease) schemes

Loss recovery

- Loss detection and notification
  - Usual approach -> sequence numbers
  - End-to-end -> like in TCP
  - Hop-by-hop -> receiver based or sender based

- Retransmission-based loss recovery
  - End-to-end -> like in TCP, expensive
  - Hop-by-hop -> nodes need to cache
    - Energy efficient

Design criteria for transport protocols



Refresher: TCP

TCP basics

- TCP applies a sliding window flow control
- Congestion window (cwnd)
  - Represents the maximum amount of outstanding data that have not been ACKed
  - The TCP sender maintains an updated value of the window
  - Its value is set by the congestion control algorithm
- Receiver window (rwnd)
  - cwnd ≤ rwnd
    - The amount of data which are not ACKed, cannot be higher than the receiver window (rwnd, set by receiver)
- When an ACK is received, the window slides forward

TCP Congestion Control

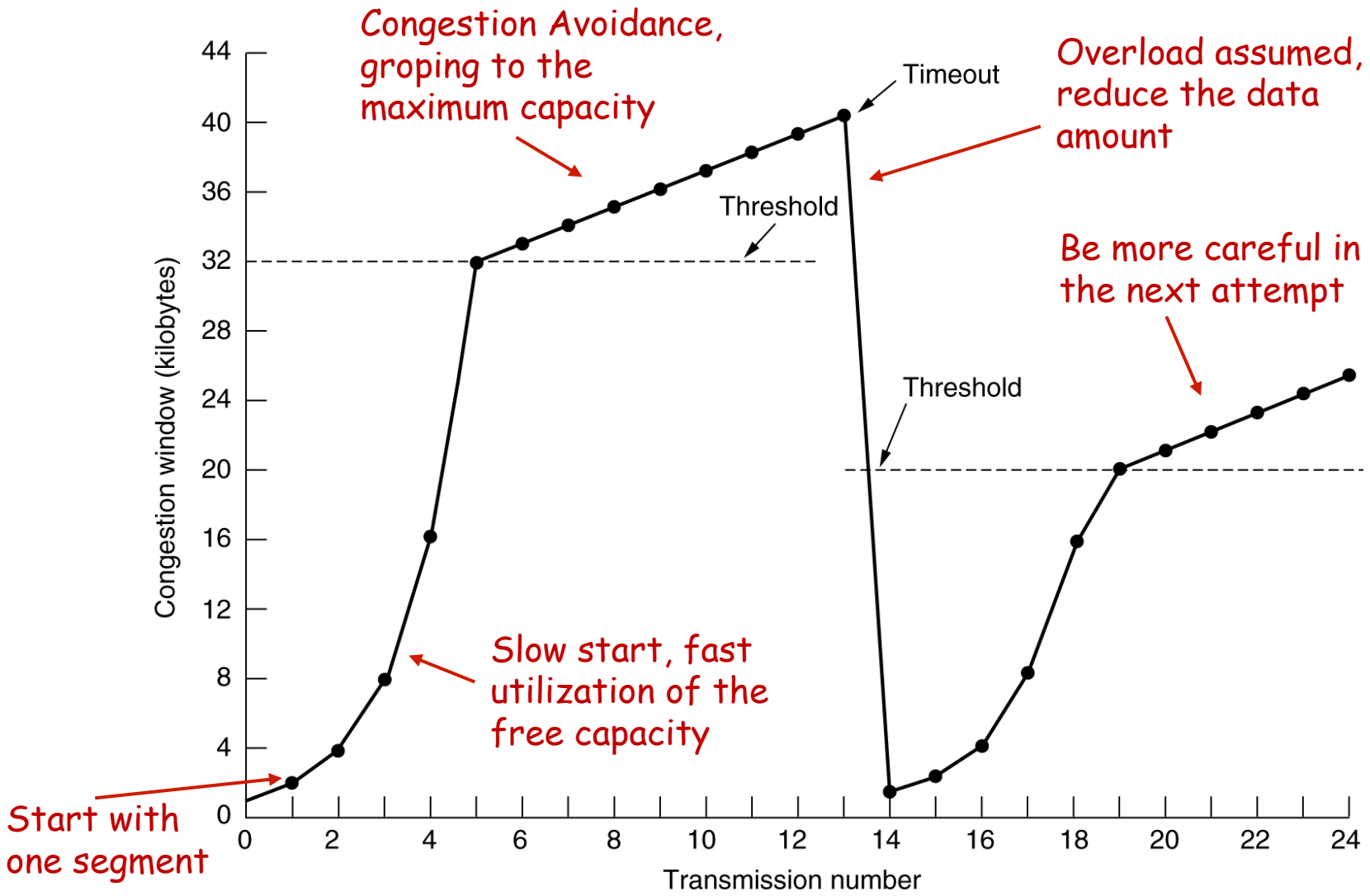
- Executed by the TCP sender
- Avoids the collapse of the network due to an overload by the traffic sources
- TCP congestion control is composed by the following algorithms:
  - Slow Start
  - Congestion Avoidance
  - Fast Retransmit
  - Fast Recovery

Tahoe

Reno, NewReno

- All based on the concept of "congestion event"

TCP Algorithm Stages



Congestion Events

- Loss of a segment = congestion
- A segment loss is detected by means of duplicate ACKs (generally ndup = 3)
- A timeout (equal to a certain value RTO) elapses without receiving any ACK for a segment

Slow Start

- Slow start is triggered
  - At the beginning of a new connection
  - When a timeout elapses
- The value of the cwnd is set equal to 1; then it is increased by 1 each time an ACK is received
- Slow Start is terminated when
  - A congestion event occurs
  - The congestion window cwnd reaches a threshold value ssthresh
- During the Slow Start the congestion window cwnd increases exponentially (it doubles at each round trip time)

Congestion Avoidance

- The congestion window cwnd is increased by 1/cwnd each time an ACK is received
- The procedure is terminated when a congestion event is detected
- During congestion avoidance, the congestion window increases linearly (it increases by 1 at each round trip time)

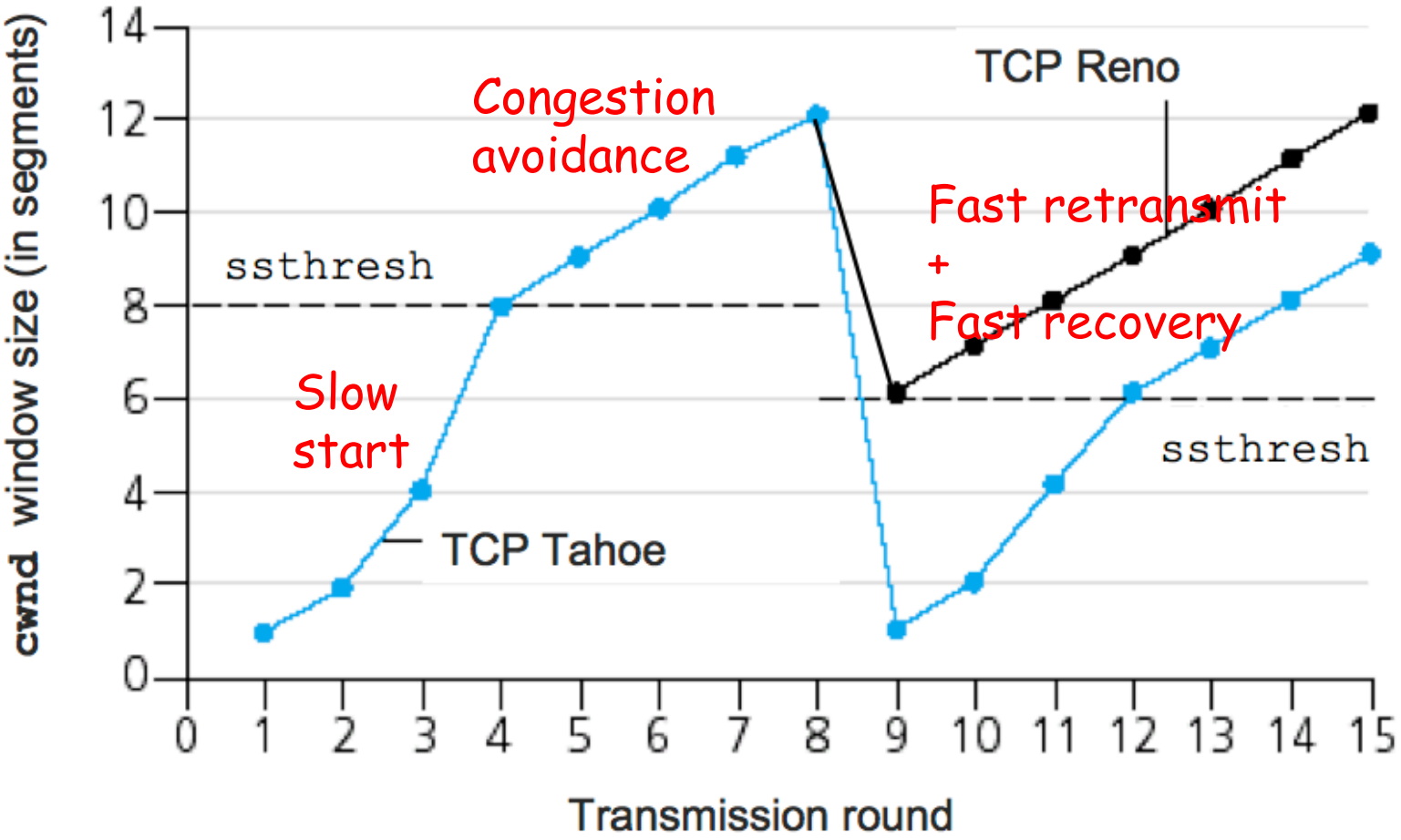
Fast Retransmit

- It is executed when ndup duplicate ACKs are received
- The sender sends again the segment for which several ACKs have been received
- The Fast Retransmit is terminated immediately and Fast Recovery is initiated

Fast Recovery

- Follows the Fast Retransmit
- The congestion window cwnd is halved
- The congestion window cwnd is increased by 1 each time an ACK is received
- When the missing segment is received, the congestion window cwnd is set equal to the value it had at the beginning of the Fast Recovery (after it has been halved)

TCP Congestion Control Recap



Summary: TCP

- Why not TCP?
  - TCP is unable to distinguish between congestion losses and transmission losses -> poor performance
  - TCP is not suitable for wireless multi-hop
  - TCP provides 100% reliability -> not required by many applications
  - TCP is connection-oriented -> expensive and not necessary
  - TCP depends on network-wide unique addresses of nodes
  - Preprocessing or aggregation of data in intermediate nodes is desirable and often necessary
    - Packets can be combined or changed before they reach destination
  - TCP is not light-weight -> too much overhead

- Why not UDP?
  - No reliability at all

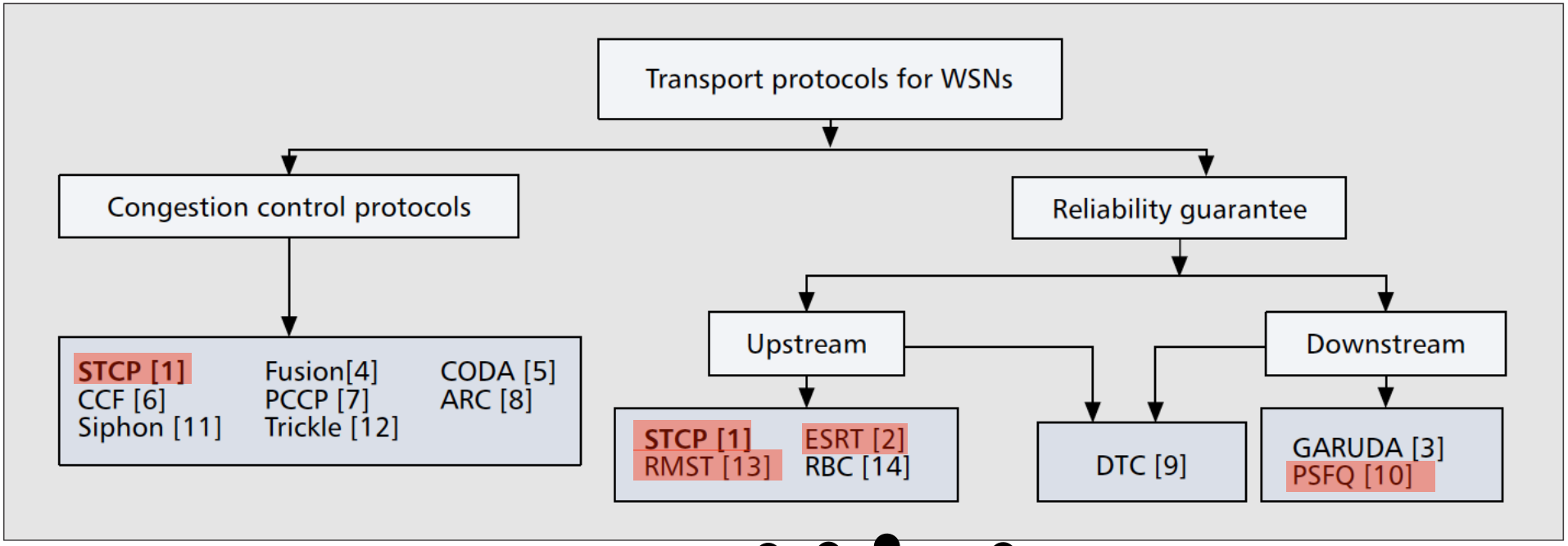
Challenges for the Transport Layer in WSNs

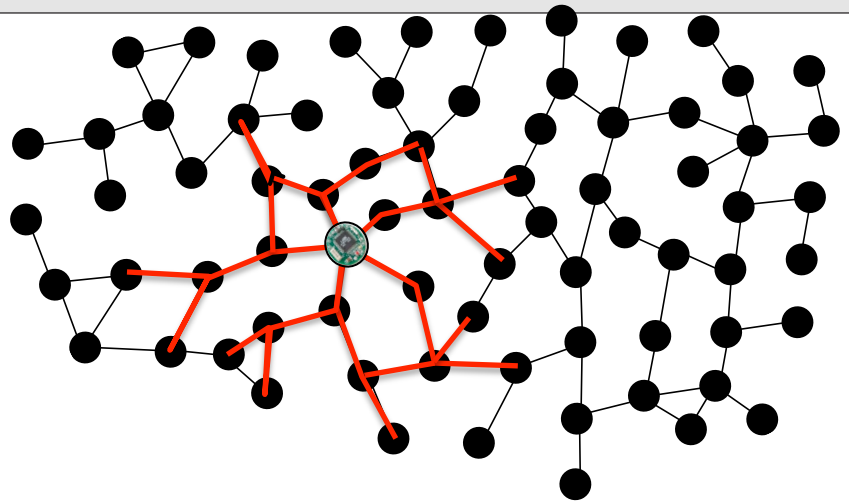
- Reliability
- Congestion Control
- Self-configuration
  - Adaptive to dynamic topologies caused by node mobility/failure/ temporary power-down, and random node deployment
- Energy Awareness
- Biased Implementation
  - Mainly run on the sink with minimum functionalities at sensor nodes
- Constrained Routing/Addressing
  - No assumption of the existence of an end-to-end global addressing

WSN transport protocols

WSN transport protocols

[Wang]





WSN transport protocols

- RMST (Reliable Multi-Segment Transport)
  - Reliability
- PSFQ (Pump Slowly Fetch Quickly)
  - Reliability
- ESRT (Event-to-Sink-Reliable Transport)
  - Reliability + Congestion

Effect of retransmissions

Effect of retransmissions

- Effect of the number of retries on the resultant probability that a packet will arrive between end points
- Independent loss probability p of transmissions
- Single-hop transmission with up to R retries
- Probability of success p_R

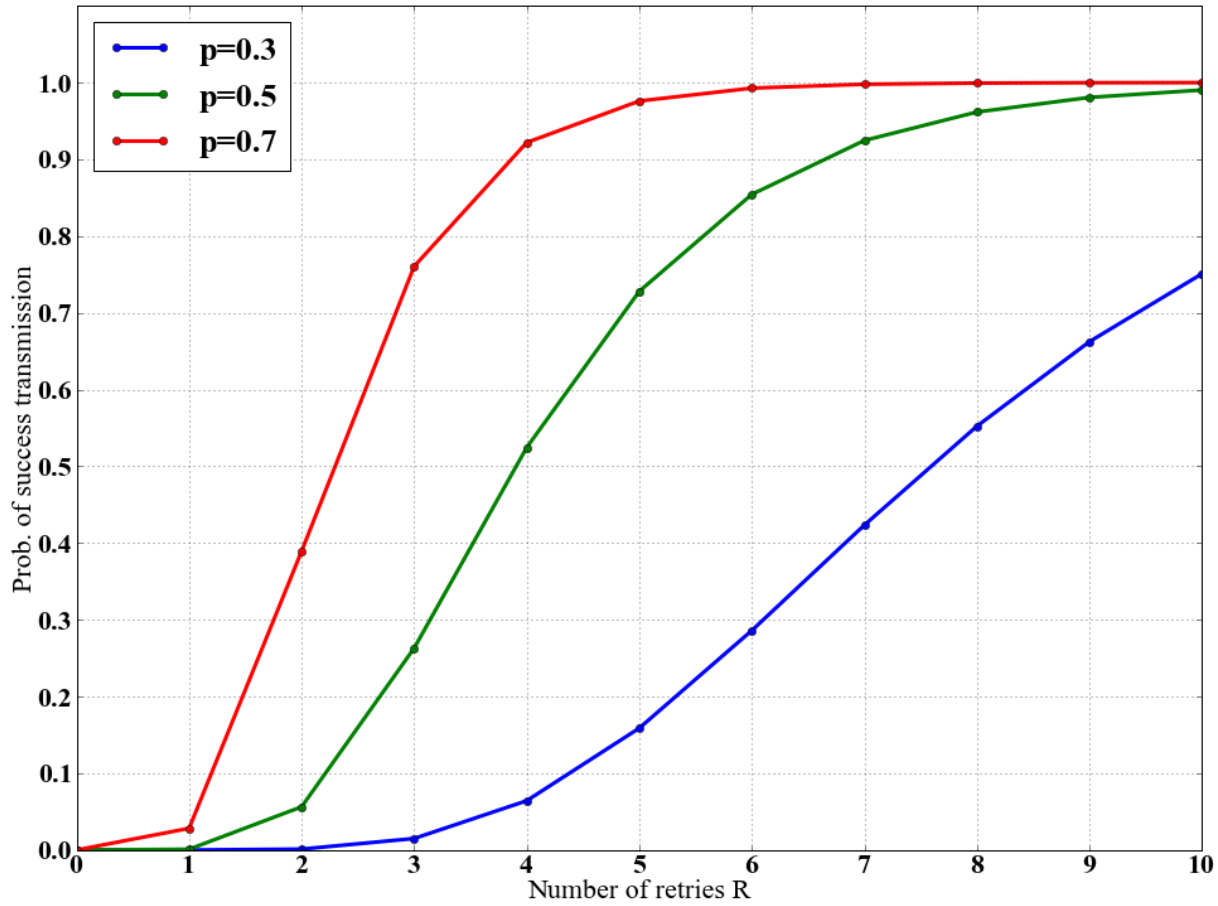
$$p_R = \sum_{i=0}^{R-1} p(1-p)^i$$

$$= 1-(1-p)^R$$

- Consider h-hop end-to-end transmission probability p_e with each hop success probability p_R

$$p_e = p_R^h$$

Effect of retransmissions



Effect of retransmissions

- Where to realize retransmissions?
  - Transport Layer -> End-to-end
  - MAC Layer -> Hopy-by-hop

Effect of retransmissions

[Stann]

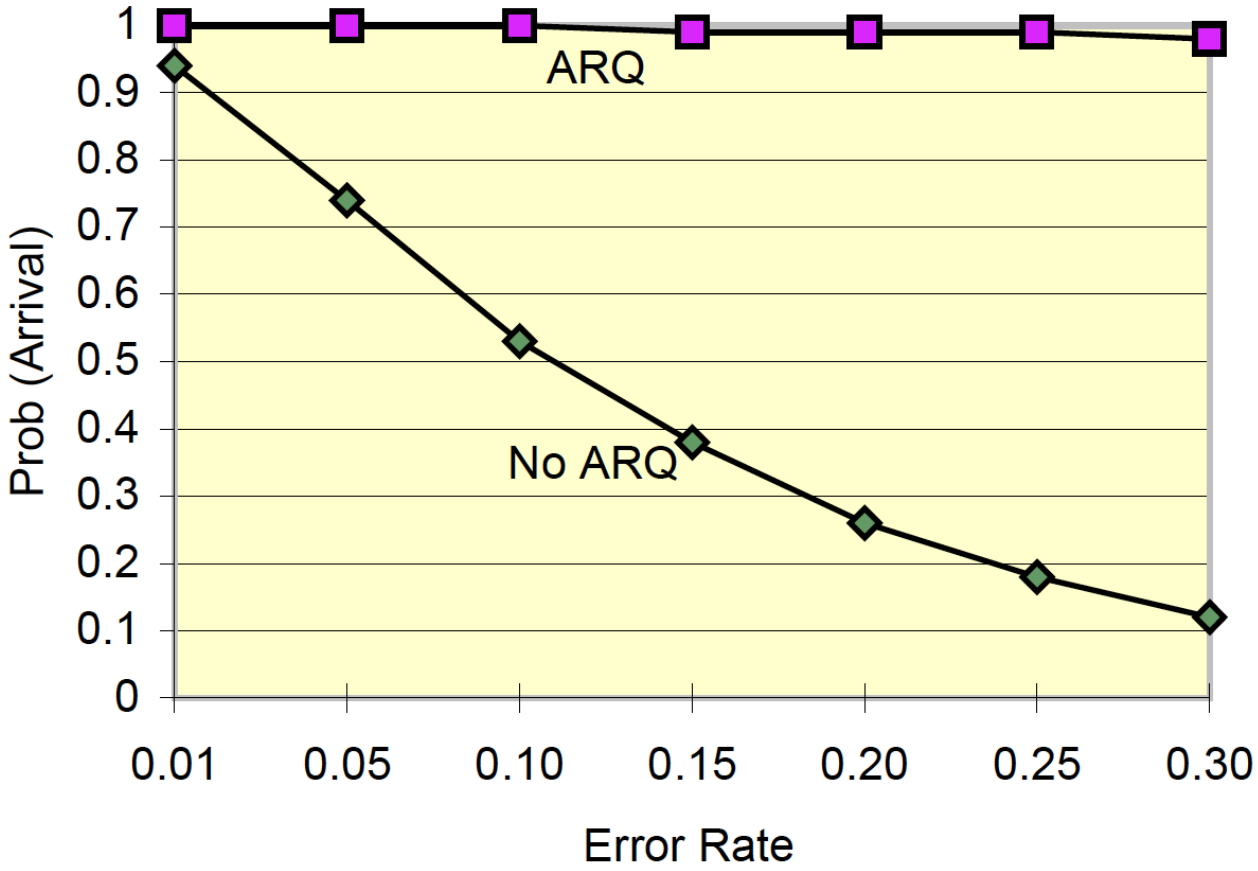


Figure 2: Probability of arrival across 6 hops

Effect of retransmissions

[Stann]

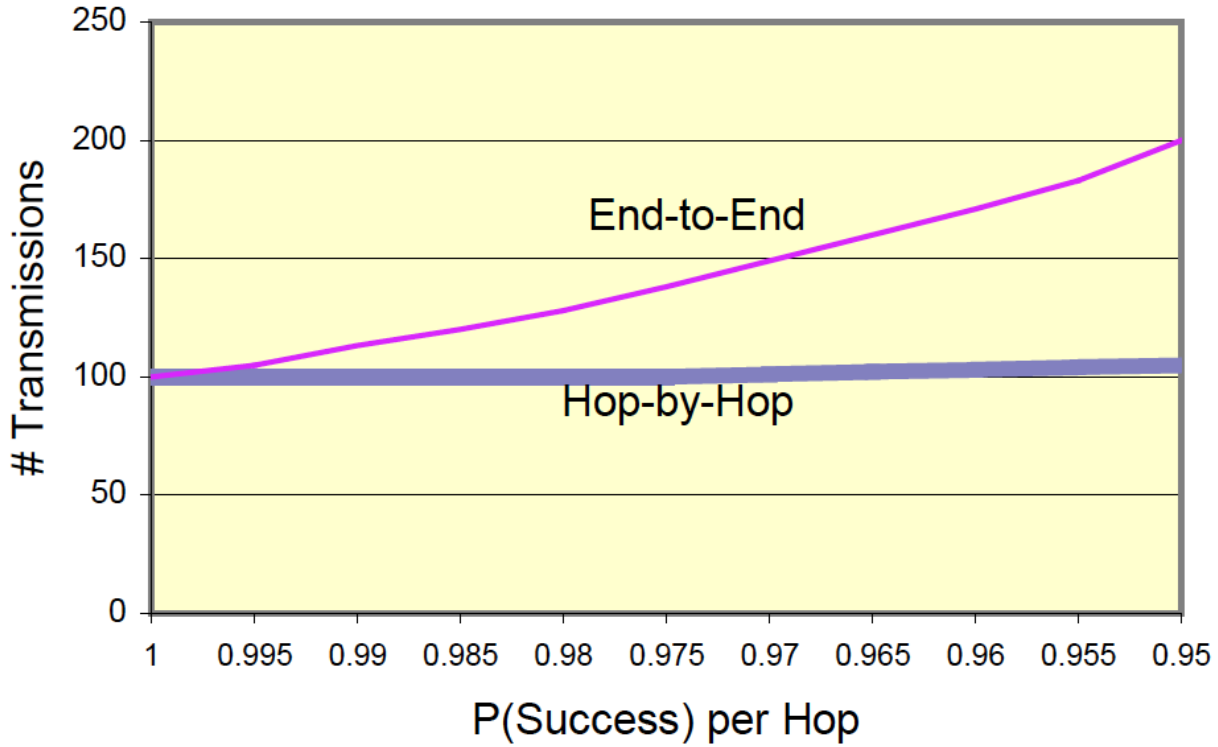
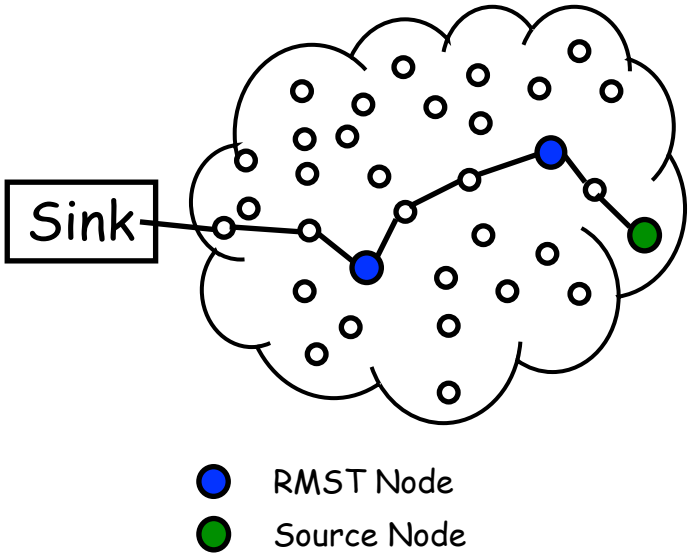


Figure 3: Number of transmissions required to send 10 fragments across 10 hops. Hop-by-hop vs. End-to-End repair.

Reliable Multi-Segment Transport (RMST)

Reliable Multi-Segment Transport (RMST)

- End-to-end data-packet transfer reliability
- Each RMST node caches the packets
- When a packet is not received before the so-called WATCHDOG timer expires, a NACK is sent backward
- The first RMST node that has the required packet along the path retransmits the packet
- In-network caching brings significant overhead in terms of power and processing
- Relies on Directed Diffusion



RMST

- Placement of reliability for data transport at different layers of the protocol stack

- Reliability in the
  - MAC layer
  - Transport layer
  - Application layer
  - Combinations of above

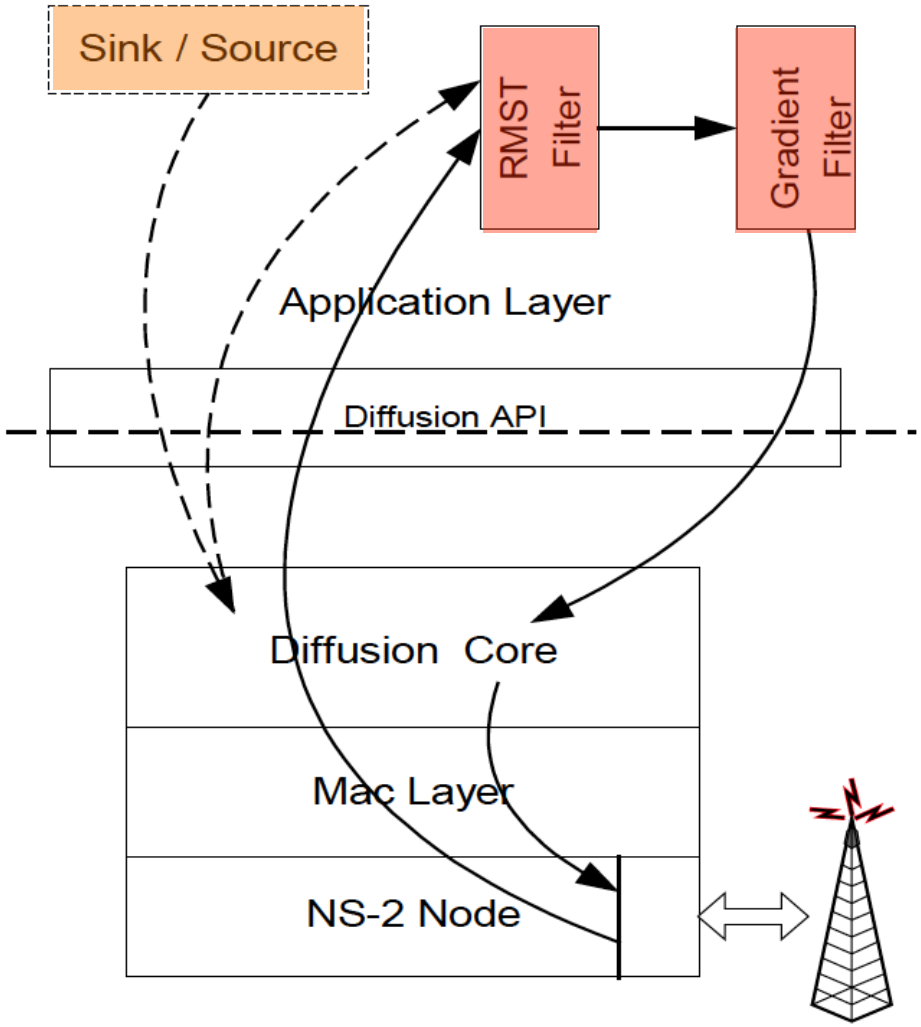
- Trade-off between implementation on the MAC vs Transport layer

RMST

- Implemented as a filter for Directed Diffusion
- Takes advantage of Directed Diffusion for
  - Routing
  - Path recovery and repair
- Adds to Directed Diffusion
  - Fragmentation/reassembly management
  - Guaranteed delivery

RMST

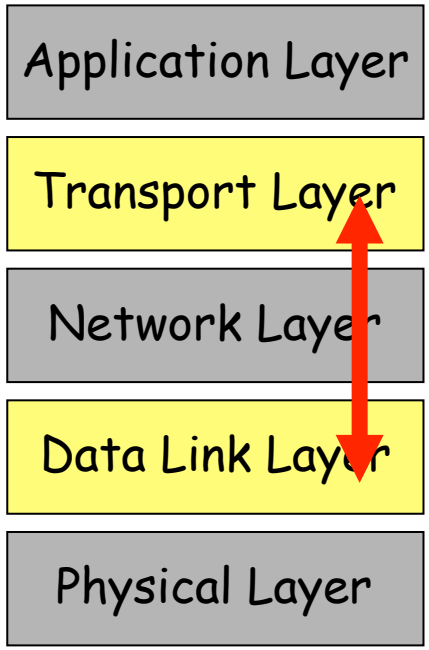
- RMST as a filter
- Transport layer pushed onto the diffusion stack
- Above the gradient filter



Placement of reliability for data transport

- Receivers responsible for fragment retransmissions
  - Receivers are not necessarily end points
  - Caching or non-caching mode determines classification of node
- Two types of loss detected by a "receiver"
  - A "hole" in a sequence of fragments
  - A truncated sequence
- RMST considers 3 layers
  - MAC
  - Transport
  - Application

Focus



MAC Layer Choices

- No ARQ
  - All transmissions are broadcast
  - No RTS/CTS or ACK
  - Reliability deferred to upper layers
  - Benefits: no control overhead, no erroneous path selection
- ARQ always
  - All transmissions are unicast
  - RTS/CTS and ACKs used
  - One-to-many communication done via multiple unicasts
  - Benefits: packets traveling on established paths have high probability of delivery
- Selective ARQ
  - Use broadcast for one-to-many and unicast for one-to-one
  - Data and control packets traveling on established paths are unicast
  - Route discovery uses broadcast

Transport Layer Choices

- End-to-End Selective Request NACK
  - Loss detection happens only at sinks (endpoints)
  - Repair requests travel on reverse (multi-hop) path from sinks to sources

- Hop-by-Hop Selective Request NACK
  - Each node along the path caches data
  - Loss detection happens at each node along the path
  - Repair requests sent to immediate neighbors
  - If data is not found in the caches, NACKs are forwarded to next hop towards source

Application Layer Choices

- End-to-End Positive ACK
  - Sink requests a large data entity
  - Source fragments data
  - Sink keeps sending interests until all fragments have been received
  - Used only as a baseline

RMST

- NACKs triggered by
  - Sequence number gaps
    - Watchdog timer inspects fragment map periodically for holes that have aged for too long
  - Transmission timeouts
    - "Last fragment" problem

- NACKs propagate from sinks to sources
  - Unicast transmission
  - NACK is forwarded only if segment not found in local cache
  - Back-channel required to deliver NACKs to upstream neighbors

RMST: Summary

- ARQ helps with unicast control and data packets
  - In high error-rate environments, routes cannot be established without ARQ
- Route discovery packets should not use ARQ
  - Erroneous path selection can occur
- RMST combines a NACK-based transport layer protocol with S-ARQ to achieve the best results

Pump Slowly, Fetch Quickly (PSFQ)

Pump Slowly, Fetch Quickly (PSFQ)

- Pace the data from a source node at a relatively low speed to allow intermediate nodes to fetch missing data segments from their neighbors
  - Assumption: no congestion, losses due only to poor link quality
- Hop-by-hop recovery
- Goals
  - Recover from losses locally
  - Ensure data delivery with minimum support from transport infrastructure
  - Minimize signaling overhead for detection/recovery operations
  - Operate correctly in poor link quality environments
  - Provide loose delay bounds for data delivery to all intended receivers

PSFQ

- Three basic operations
  - pump
  - fetch
  - report

- Alternate between multi-hop forwarding when low error rates and store-and-forward when error rates are higher

PSFQ

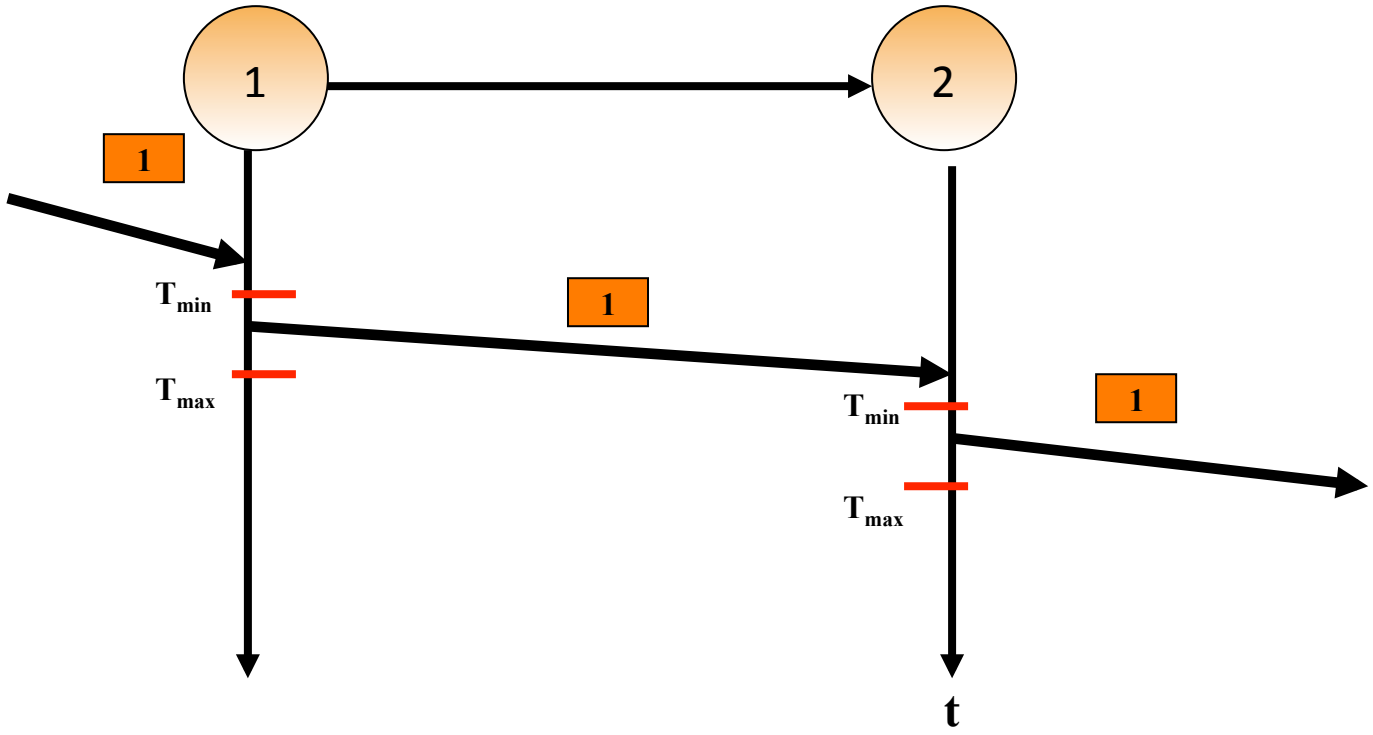
- Pump
  - Node broadcasts a packet to its neighbors every T_{min}
  - Neighbors who receive the packet check against their local cache discarding any duplicates
  - If it is just a new message the packet is buffered and the Time-To-Live (TTL) field in the header is decreased by 1
  - If TTL is not zero and there is no gap in the sequence number the packet is scheduled for transmission within a random time T_{tx}

$$T_{min} < T_{tx} < T_{max}$$

  - The random delay allows a downstream node to recover missing segments before the next segment arrives from an upstream node
  - It also allows reducing the number of redundant broadcasts of the same packet by neighbors

PSFQ: Pump operation

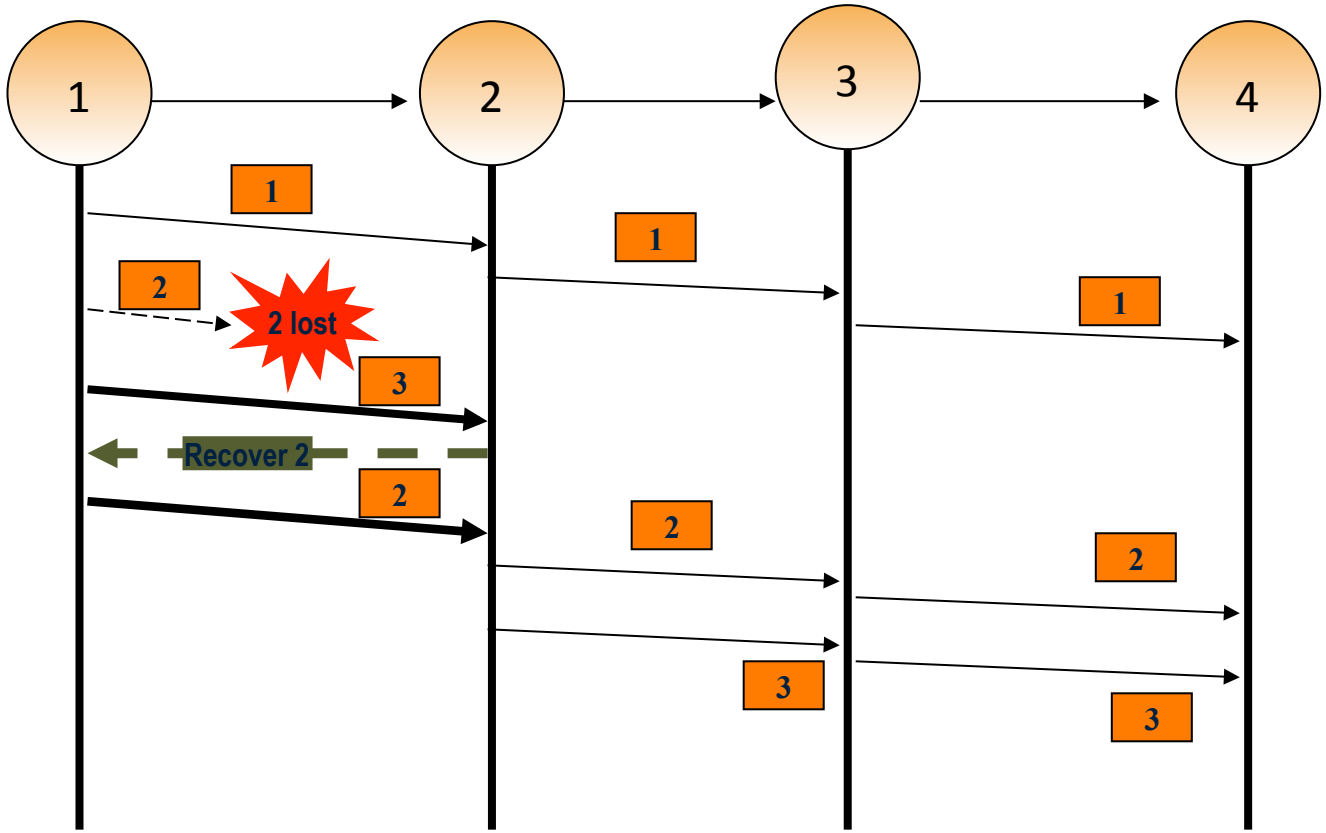
If not duplicate and in-order and TTL not 0
Cache and schedule for forwarding at time t (T_{min}< t < T_{max})



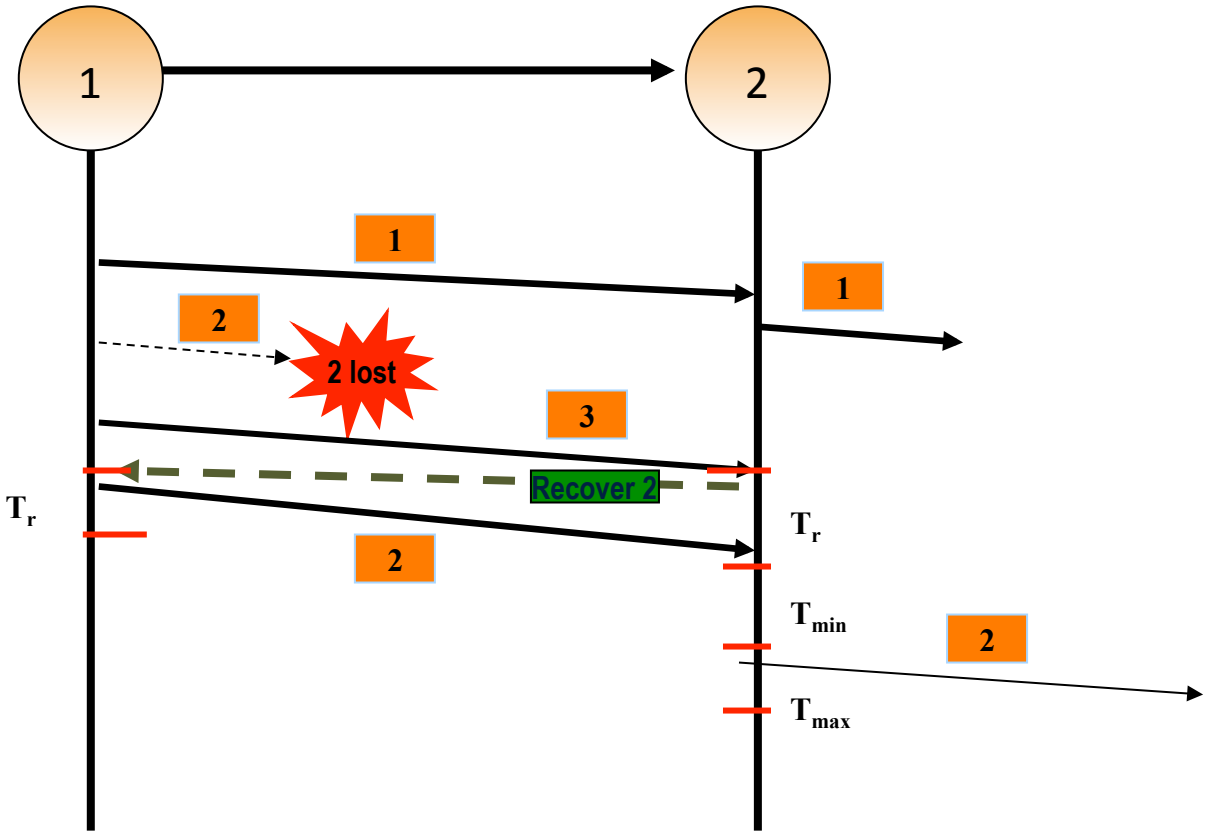
PSFQ: Fetch operation

- Fetch
  - A node goes into fetch mode when a sequence number gap is detected
  - In fetch mode a node aggressively sends out NACK messages to its immediate neighbors to request missing segments
  - It is very likely that consecutive packets are lost -> a "window" is used to specify the range of missing packets
  - A node that receives a NACK message checks the loss window field against its cache. If found the packet is scheduled for transmission at a random time in (0, T_r)
  - Neighbors cancel a retransmission when a reply for the same segment is overheard
  - NACK messages are not propagated to avoid message implosion
  - "proactive fetch" mode to take care of situations, where the last segment of a message is lost.
    - Node sends a NACK for the remaining segments when they have not been received after a time period T_{pro}

PSFQ: Fetch mode (Recovery from Errors)



PSFQ: Fetch quickly



PSFQ: Report

- Used to provide feedback data of delivery status to source nodes
- To minimize the number of messages, a report message travels back from a destination node to the source nodes
  - Intermediate nodes can piggyback their report messages in an aggregated manner

PSFQ: Results

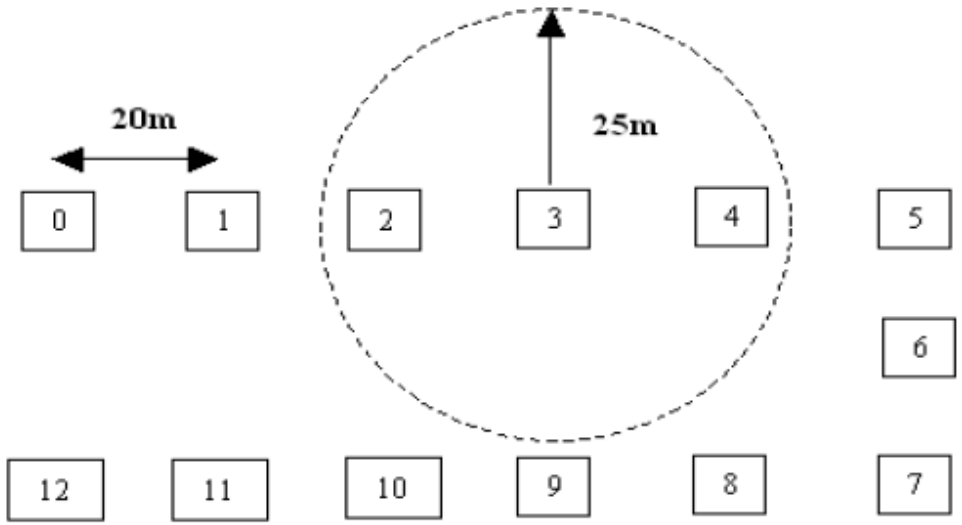
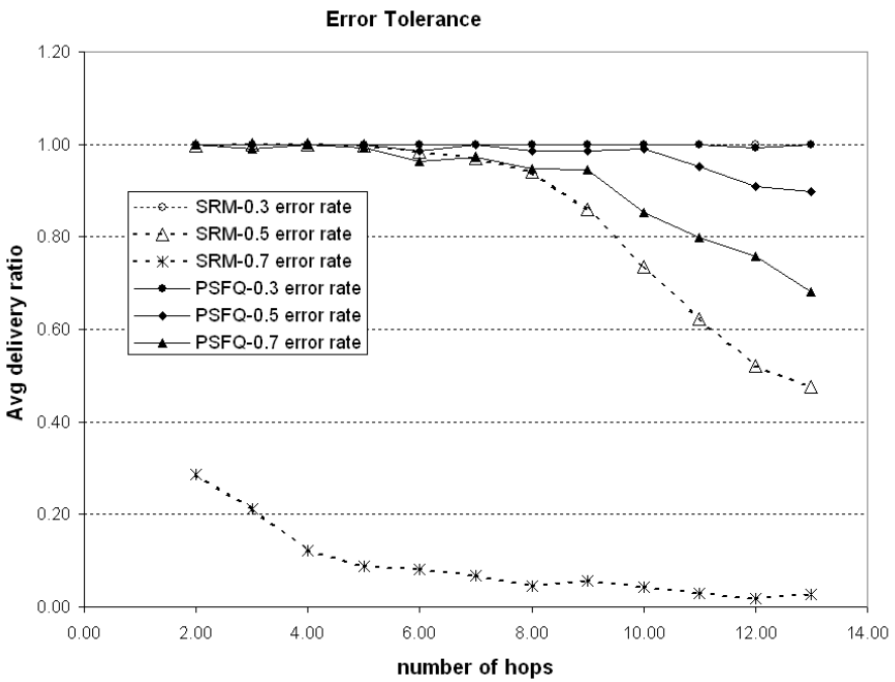
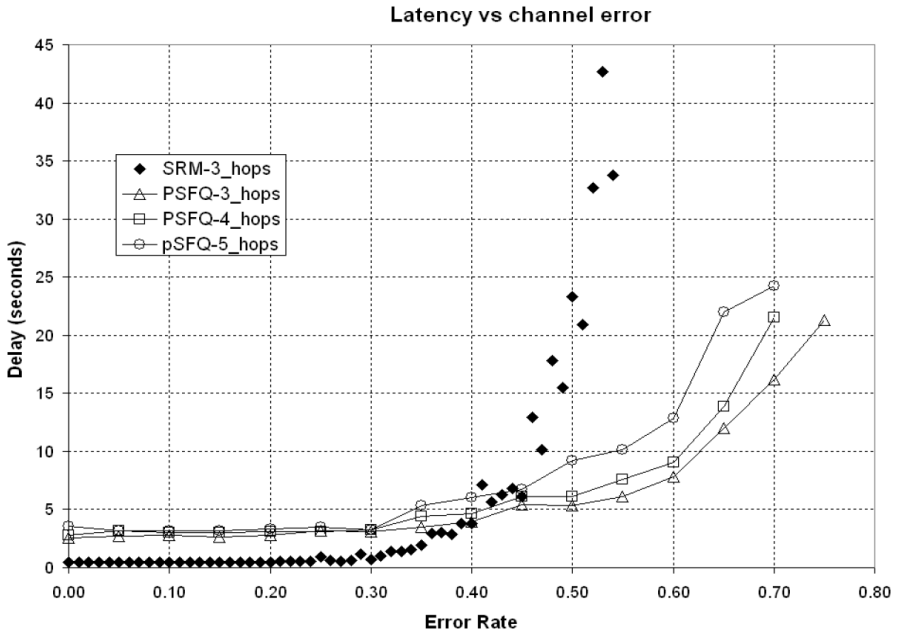


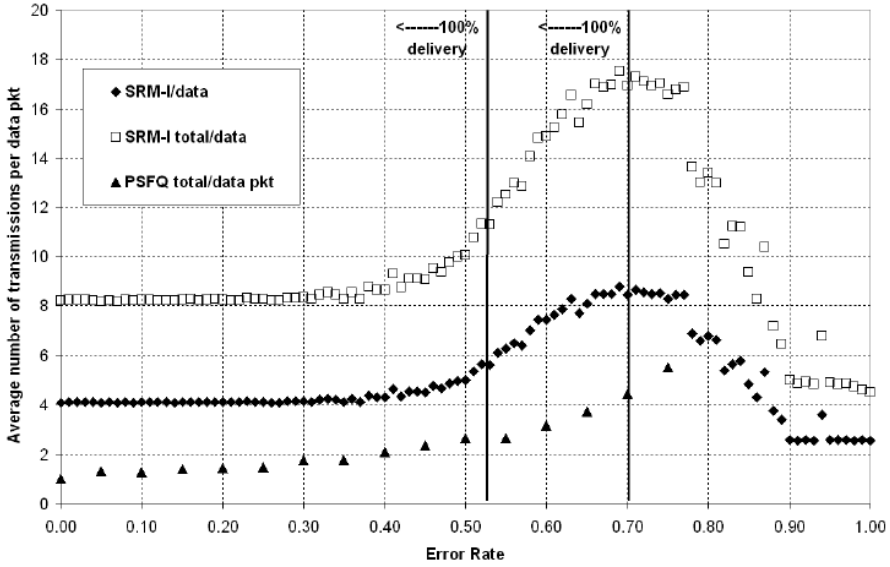
Figure 4. Sensor network in a building. A user node at location 0 injects 50 packets into the network within 0.5 seconds.

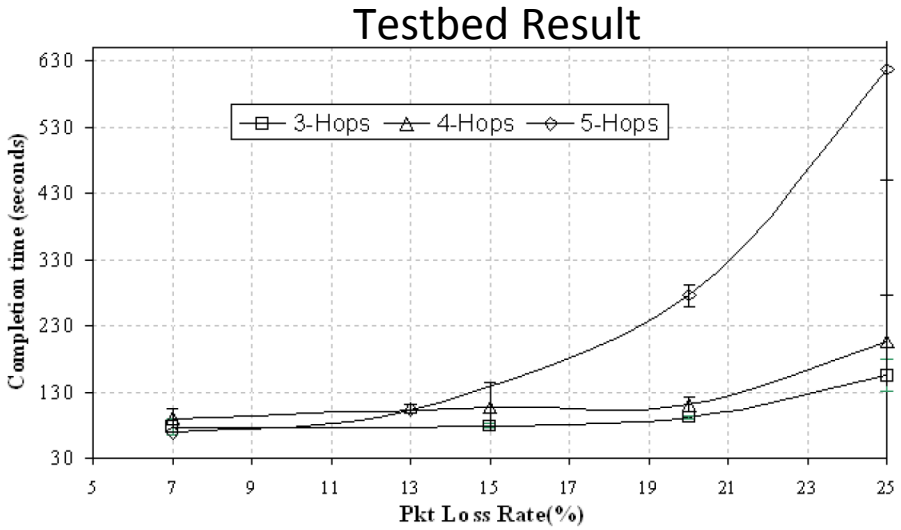
PSFQ: Results





PSFQ: Results





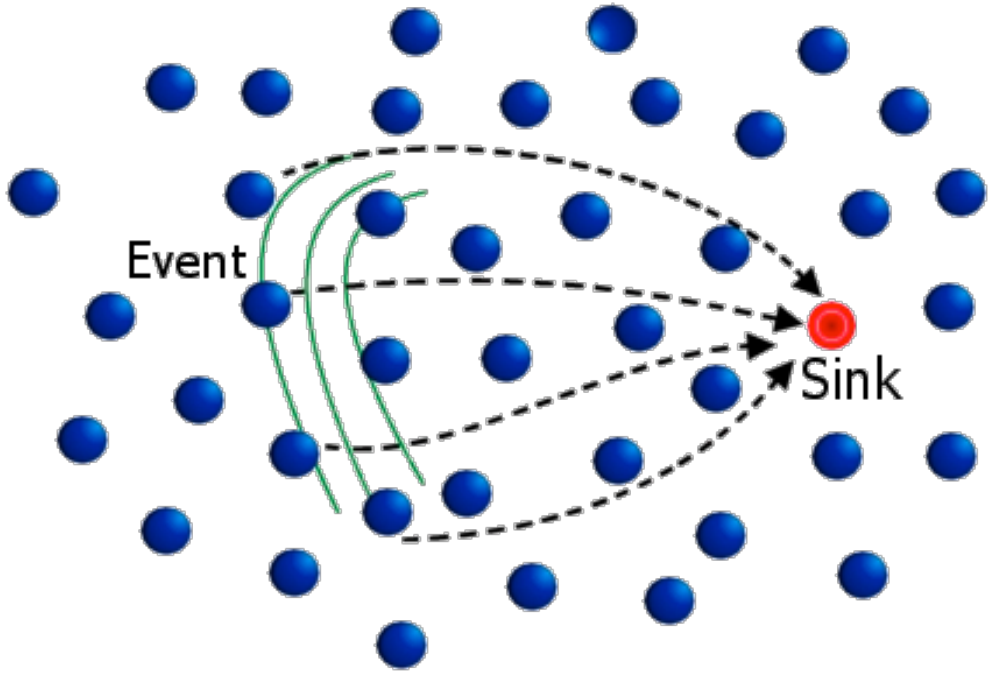
PSFQ: Summary

- Lightweight and energy efficient
- Simple mechanism
- Scalable and robust
- Need to be tested for high bandwidth applications
- Cache size limitation
- Does not address congestion control

Event-to-Sink Reliable Transport (ESRT)

Event-to-Sink Reliable Transport (ESRT)

- In a typical sensor network application the sink node is only interested in the collective information of the sensor nodes within the region of an event and not in any individual sensor data
- What is needed is a measure of the accuracy of the information received at the sink, i.e., and event-to-sink reliability



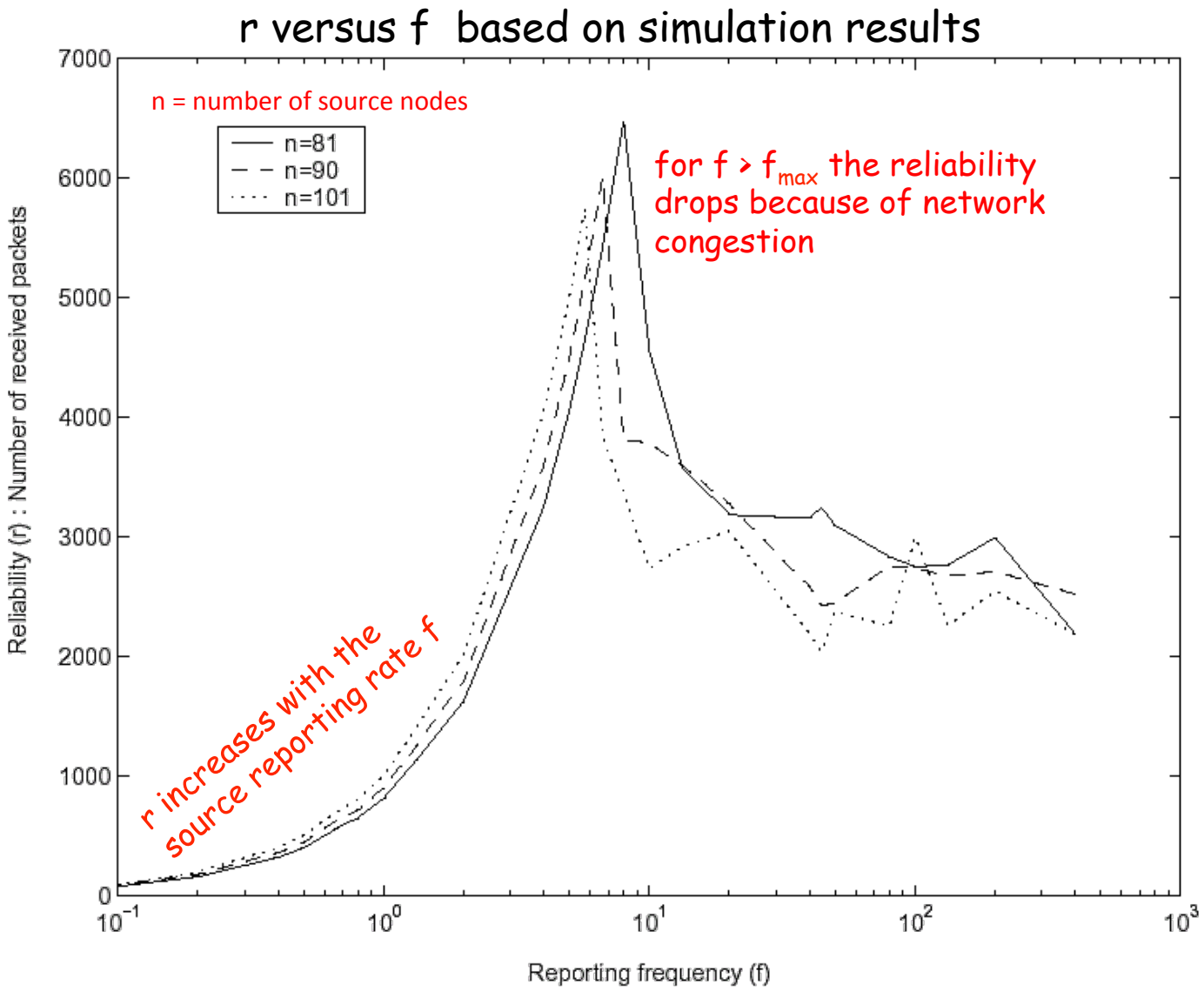
ESRT

- The basic assumption is that the sink does all the reliability evaluation using parameters that are application dependent
- One such parameter is the decision time interval τ
- At the end of the decision interval the sink derives a reliability indicator $\mathbf{r_i}$ based on the reports received from the sensor nodes
- r_i is the number of packets received in the decision interval
- If **R** is the number of packets required for reliable event detection then

$$\mathbf{r_i > R}$$

  is needed for reliable event detection
- There is no need to identify individual sensor nodes but instead there is the need to have an **event ID**
- The **reporting rate**, **f**, of a sensor node is the number of packets sent out per unit time by that node
- The ESRT protocol aims to **dynamically adjust** the reporting rate to achieve the required detection reliability R at the sink

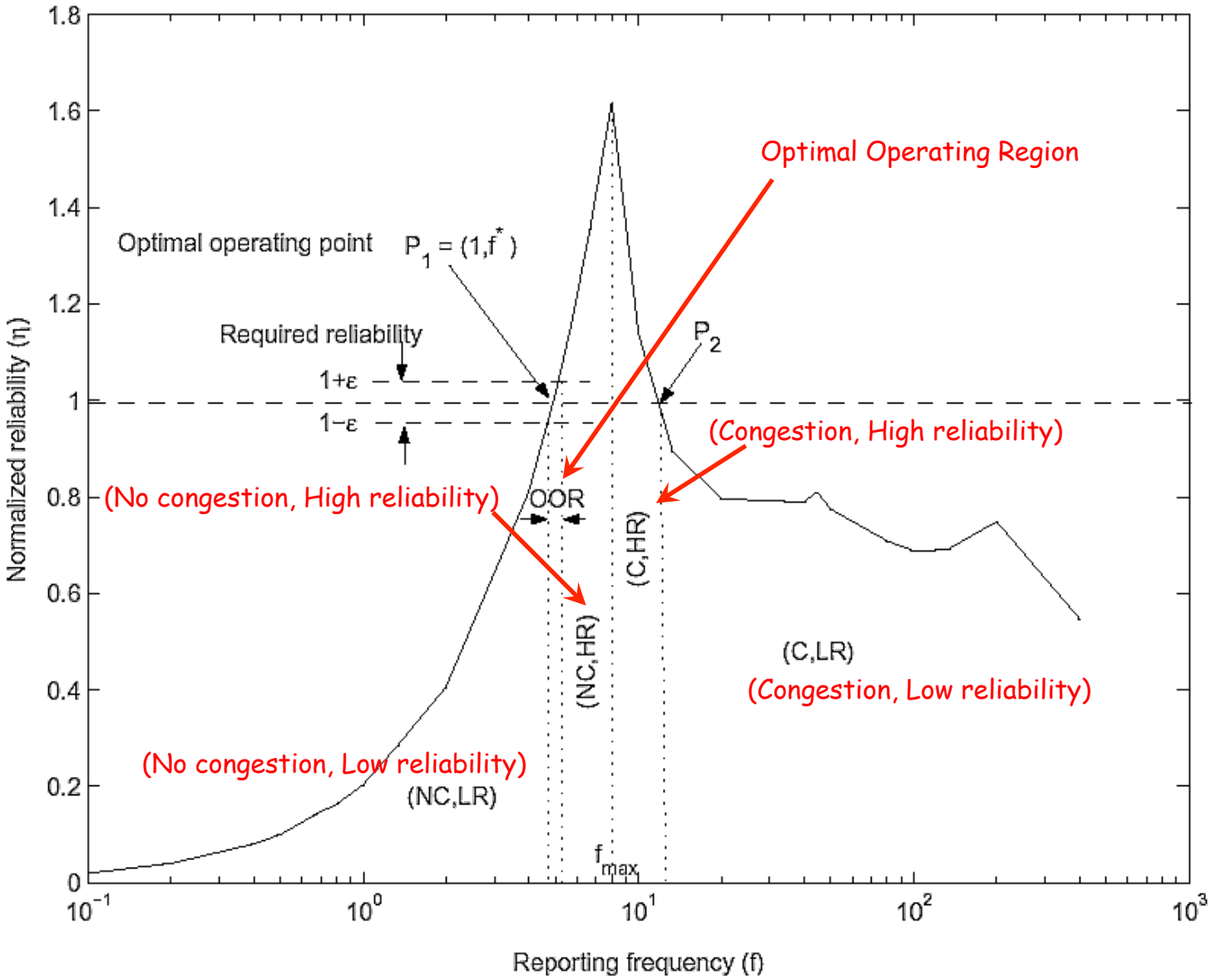
ESRT


r versus f based on simulation results
n = number of source nodes
n=81
n=90
n=101
for f > f_max the reliability drops because of network congestion
r increases with the source reporting rate f
Reliability (r) : Number of received packets
Reporting frequency (f)
7000
6000
5000
4000
3000
2000
1000
0
10^-1
10^0
10^1
10^2
10^3

ESRT: Protocol overview

- The algorithm runs on the sink
- Sensor nodes:
  - Listen to sink broadcasts and update their reporting rates accordingly
  - Have a simple congestion detection mechanism and report to the sink
- The sink:
  - Computes a normalized reliability measure $\eta_i = r_i / R$
  - Updates f based on η_i and if $f > f_{max}$ or $f < f_{max}$ in order to achieve the desired reliability
  - Performs congestion decisions based on feedback reports from the source nodes
- Congestion detection:
  - Uses local buffer level monitoring in sensor nodes
  - When a routing buffer overflows the node informs the sink by setting the congestion notification bit in the header packets traveling downstream

ESRT: Network states



ESRT: Frequency Update

State	f update	Action
(NC, LR)	$f_{i+1} = \frac{f_i}{\eta_i}$	Multiplicative increase f to achieve required reliability as soon as possible
(NC, HR)	$f_{i+1} = \frac{f_i}{2}\left(1 + \frac{1}{\eta_i}\right)$	Decrease f conservatively, reduce energy consumption and not lose reliability
(C, HR)	$f_{i+1} = \frac{f_i}{\eta_i}$	Aggressively decrease f to relieve congestion as soon as possible
(C, LR)	$f_{i+1} = f_i^{(\eta_i/k)}$	Exponential decrease. k is the number of successive decision intervals spent in state (C, LR)
OOR	$f_{i+1} = f_i$	Unchanged

ESRT: Summary

- Focus on event-to-sink reliability instead on individual end-to-end reliability
- Congestion control mechanism results in energy savings
- Self-configuration property of the protocol is very valuable for random and dynamic topologies

STCP

Sensor Transmission Control Protocol (STCP)

- STCP provides a
  - generic
  - scalable
  - reliable transport
- Majority of STCP functionalities are implemented at the base station
- Each node might be the source of multiple data flows
  - Different flow type, transmission rate, and required reliability

- Before transmitting packets, sensor nodes establish an association with the base station via a Session Initiation Packet

Sequence Number (16)		Flows (8)	Options (8)
Clock (32)			
Flow Id #1 (8)	Flow Bit (8)	Trans. Rate (8)	Reliability (8)
Flow Id #2 (8)	Flow Bit (8)	Trans. Rate (8)	Reliability (8)
⋮	⋮	⋮	⋮
Flow Id #N (8)	Flow Bit (8)	Trans. Rate (8)	Reliability (8)

- Sequence number: zero for the session initiation packet
- Flows: number of flows originating at the node
- Clock: Clock value at the time of transmission
- Flow Id: Identify packets of a flow
- Flow Bit: Specifies whether the flow is continuous or event-driven
- Transmission rate: For continuous flows indicates the rate at which a packet will be transmitted by the source node
- Reliability: Expected reliability required by the flow

- STCP Data Packet Header

Sequence Number (16)	Flow Id (8)	CN (1)	Options (7)	Clock (32)

  - CN: Congestion notification

- STCP Acknowledgement Packet

Sequence Number (16)	Flow Id (8)	CN (1)	ACK / NACK (1)	Options (6)

Continuous Flows

- Sink knows the transmission rate -> can compute the expected arrival time for the next packet
- Sink sends NACK if packet is not received
- Estimated trip time (ETT) based on received packets

- Expected time calculation
  - Timeout is calculated by the expression (T + α*ETT), where T is the time between successive transmissions and α is a positive integer that varies with ETT
  - The second approach is Jacobson/Karels algorithm which considers the variance of the trip time. ETT is used instead of Round trip time.

Event-driven Flows

- The source node buffers each transmitted packet till an ACK is received.
- When an ACK is received, the corresponding packet is deleted from the buffer.
- The nodes maintain a buffer timer that fires periodically.
- When the timer fires, packets in the buffer are assumed to be lost and are retransmitted.

Congestion Detection and Avoidance

- STCP adopts RED with explicit congestion notification with some modification
- Each STCP data packet has a congestion notification bit in its header.
- Every sensor node maintains two thresholds in its buffer: t_{lower}, t_{higher}
- When the buffer reaches t_{lower}, the congestion bit is set with a certain probability.
- When the buffer reaches t_{higher}, the node will set the congestion notification bit in every packet it forwards.
- On receiving this packet, the base station informs the source of the congested path by setting the congestion bit in the acknowledgement packet

Results

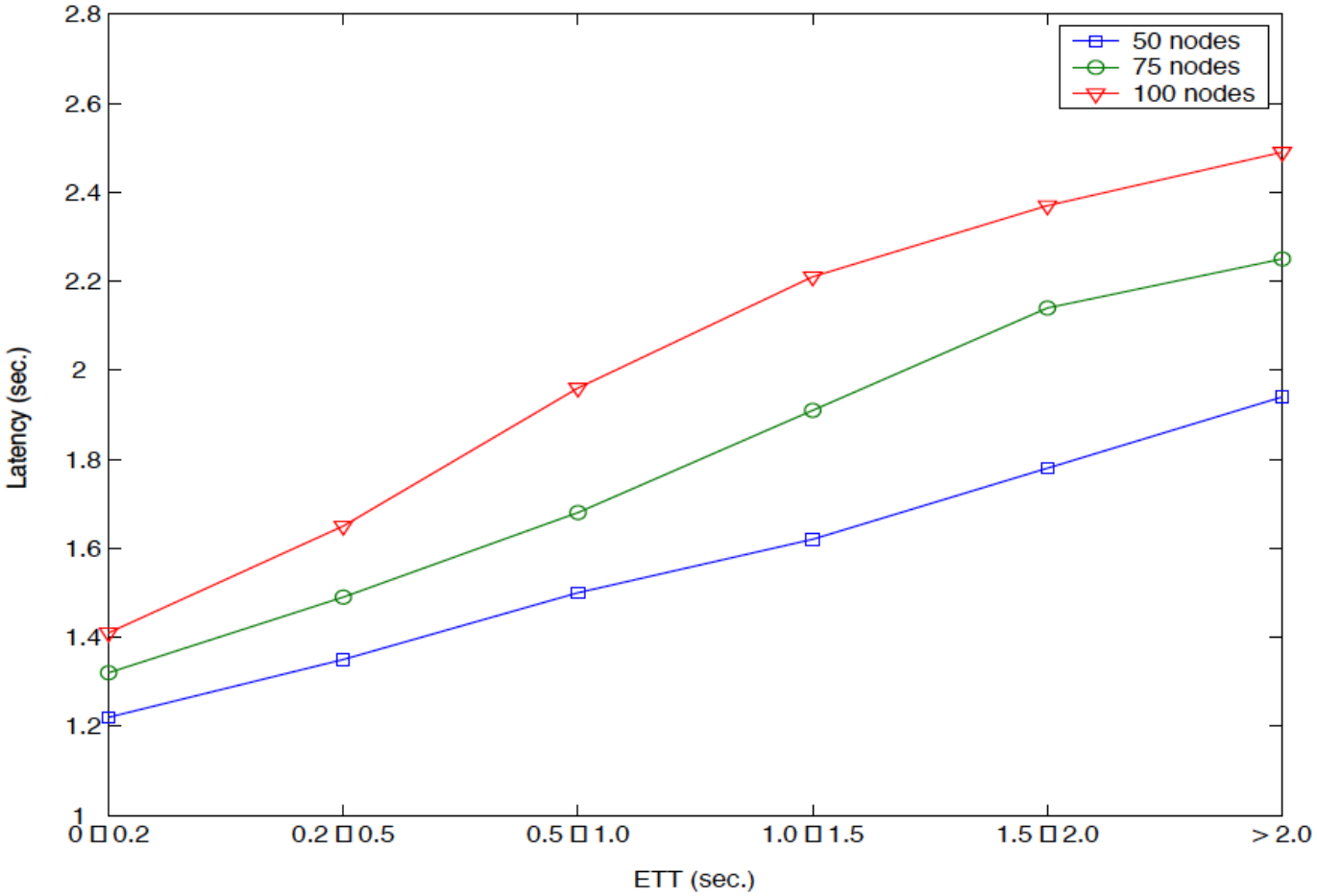


Fig. 7. Average packet latency

Results

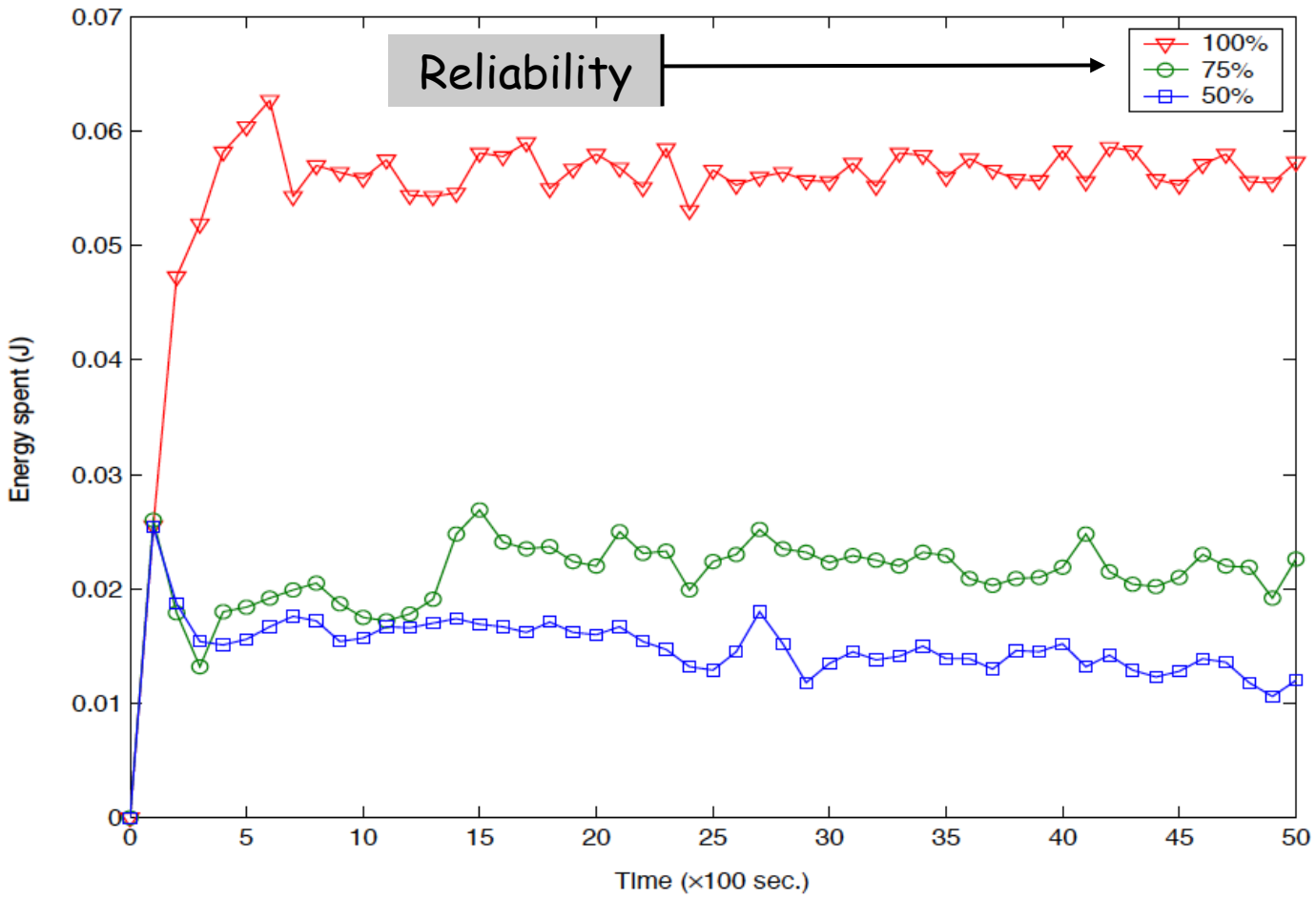


Fig. 8. Energy consumption for a 100 node network

Summary

Protocol	Error Recovery	Reliability Direction	Congestion Ctrl	MAC/ Routing Requirement	Sim	OS	Purpose	Metrics
RMST	h-h/e-e	event to sink	--	DF	ns2	--	evaluate tradeoffs	total bytes
PSFQ	h-h	sink to sensors	--	Broadcast	ns2	tos	compare SRM	avg delivery ratio/overhead/ latency
ESRT	--	collective event to sink	event report frequency	CSMA/CA	ns2	--	paramete r tuning	convergence to OOR
GARUDA	h-h/e-e	sink to sensors	--	Broadcast	ns2	--	loss recovery	Latency/ energy consumption
MOAP	h-h	sink to sensors	--	broadcast/ uni-cast	Em Star	tos	Code Updates	Latency/ energy consumption
CODA	--	collective event to sink	event report frequency	CSMA	ns2	tos	paramete r tuning	energy tax fidelity penalty

h-h = hop by hop
e-e = end to end
DF = Directed Diffusion
tos = Tiny OS

Summary

- Wireless TCP variants are NOT suitable for WSN
  - Resource constraints
- Different notion of reliability
  - End-to-end
  - Hop-by-hop
- Huge buffering requirements
- ACKing is energy draining

Literature

Literature

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