IP SHOWCASE THEATER™

CURATED BY VSF VIDEO SERVICES FORUM

Error Resilient Internet Video Transmission

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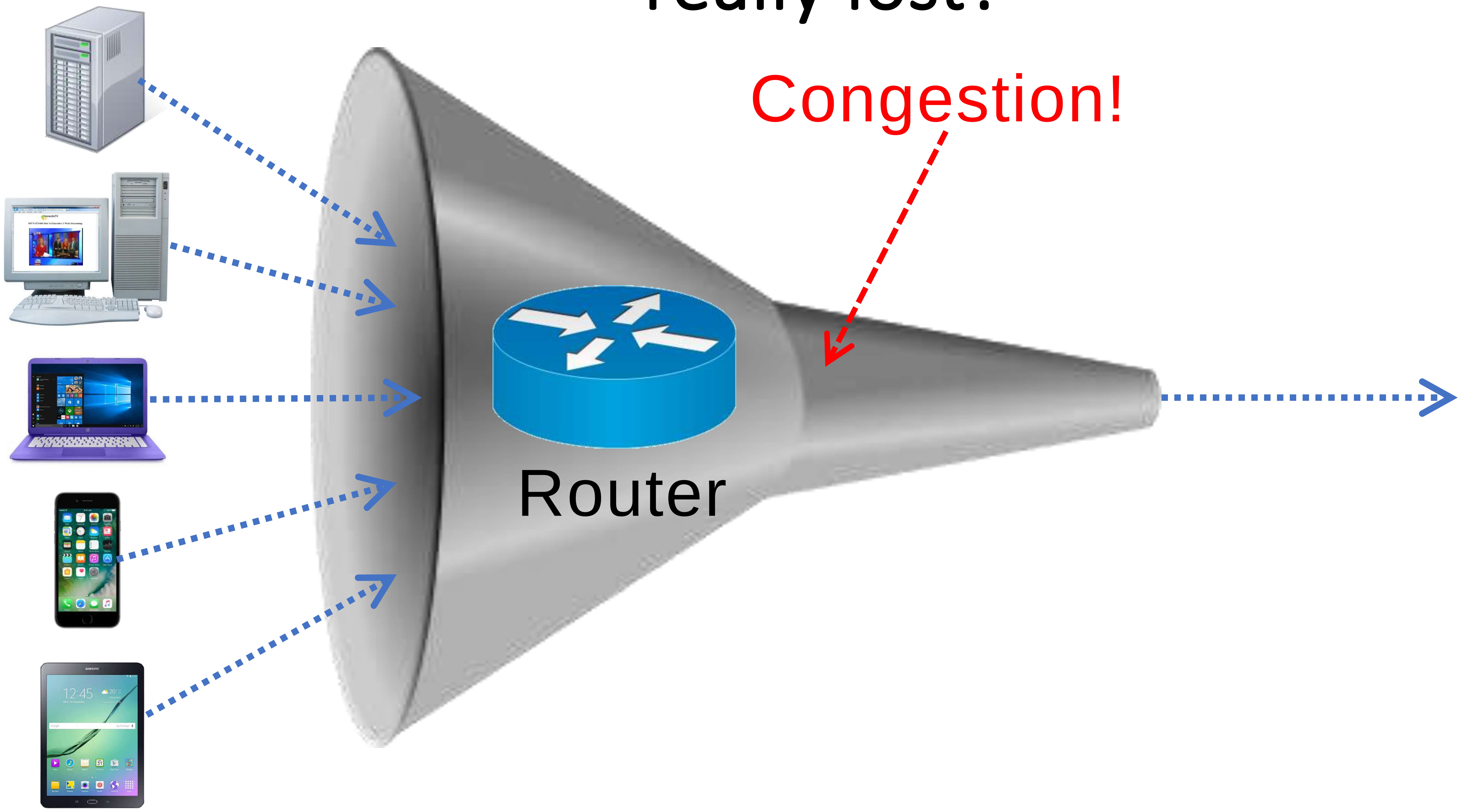
Motivation

- There are a number of protocols in use today to transport Video over IP.
- Since the “I” in IP stands for “Internet”, the Internet can (potentially) be used to transport Video over IP.
  - Low-cost contribution links!!
- However, not all Video over IP protocols are suitable for transporting Video on the Internet because:
  - **The Internet drops packets**
  - Video over IP is compressed and needs every bit
  - **Video over IP cannot take packet drops**
  - The Video over IP **protocol** has to handle this issue

So, where are packets really lost?



What is an "acceptable" packet loss?

- Video compression works by removing redundancy from the content
  - Every bit of compressed video is very important
- There is a simple way to look at the effect of packet loss:
  - Assume that every packet that is dropped by the network causes a noticeable glitch in the video
    - A block of packets dropped together causes one glitch
  - Decide how many glitches per (day/hour/minute) is acceptable to you

Some numbers

Assume a 4 Mb/s stream, with 1316-byte packets

Dropping one packet in	Produces a glitch every
1,000	2.6 seconds
10,000	26 seconds
100,000	4 minutes 23 seconds
1,000,000	44 minutes
10,000,000	7 hours 19 minutes

In order to achieve reliable operation on the Internet, a network protocol is needed to "recover" in some way the packets that have been lost.

Protocols Considered

- SMPTE-2022 FEC
  - Transmit redundant information with the packets
  - Losses may be recovered from received packets and redundant information
- Retransmission (ARQ)
  - If a packet is lost, receiver will request a retransmission

RTP plus SMPTE-2022 FEC

- Basic idea:
  - Transmit the video using RTP
    - That gets you timestamps and sequence numbers
    - Sequence numbers let you know when packets were dropped
  - Transmit "extra" FEC packets
  - If packets are lost in the network, it ***may*** be possible to rebuild them from the received packets and FEC packets:
    - For each N packets send 1 FEC packets
    - If there is one loss in this set of N+1 packets, it can be corrected
  - Use a matrix arrangement to deal with burst losses

Some FEC Numbers

Columns	Rows	Recovery Capability	Overhead	Latency @ 2 Mb/s	Latency @ 10 Mb/s
5	5	5 pkts every 25	20%	263 ms	53 ms
10	5	10 pkts every 50	20%	526 ms	105 ms
20	5	20 pkts every 100	20%	1052 ms	211 ms
10	10	10 pkts every 100	10%	1052 ms	211 ms

ARQ

- ARQ stands for:
  - **A**utomatic **R**epeat re**Q**uest
  - **A**utomatic **R**epeat **Q**uery
- This is the generic name for a number of retransmission strategies in the face of packet loss
  - Standard TCP uses a couple of ARQ variants
- In video transmission, the most useful variant is "Selective Retransmission" (NACK-based)
  - If you don't hear from me, everything is OK
  - If I miss anything, I let you know and you resend just that

ARQ Illustration

Comparison of FEC and ARQ

- FEC and ARQ have “decent” latency (typically 1 second or less)
  - May be acceptable for some forms of live contribution
- How do these two protocols compare?
  - Statistical models
  - Testing on a simulated network
  - Measurement data

A little probability and statistics…

- Assume independent loss probability for each transmitted packet (binomial distribution)
- Calculate the rate of packets still lost *after* correction with statistical analysis
- This allows us to theoretically compare the performance of the various protocols and settings
- Our variables are:

R = number of requests (ARQ)
N = number of packets per row (FEC)
M = number of packets per column (FEC)


Percentage Packet Loss (p_c) for ARQ and Column FEC
Residual Packet Loss Percentage(p_c) after Correction
100.0000000000
0.0100000000
0.0000010000
0.0000000001
0.1
1.0
10.0
Percentage Packet Drop (p)
Number of Requests/Packets
N = 10
N = 20
N = 5
R = 1
R = 2
R = 4

Percentage Packet Loss (p_c) for ARQ and Row-Column FEC
Residual Packet Loss Percentage(p_c) after Correction
100.0000000000
0.0100000000
0.0000010000
0.0000000001
0.1
1.0
10.0
Percentage Packet Drop (p)
Number of Requests/Packets
M = 10, N = 10
M = 5, N = 20
M = 5, N = 5
R = 1
R = 2
R = 4

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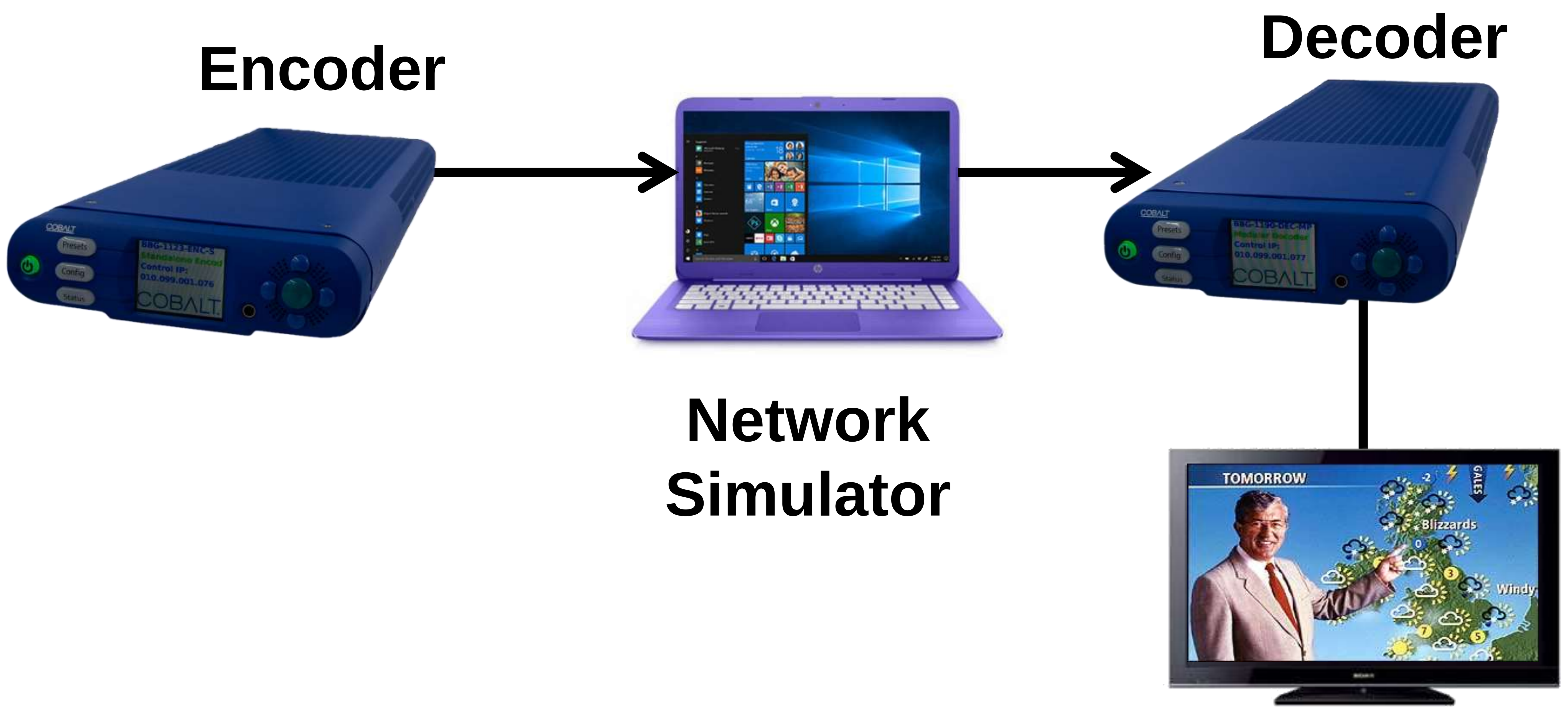
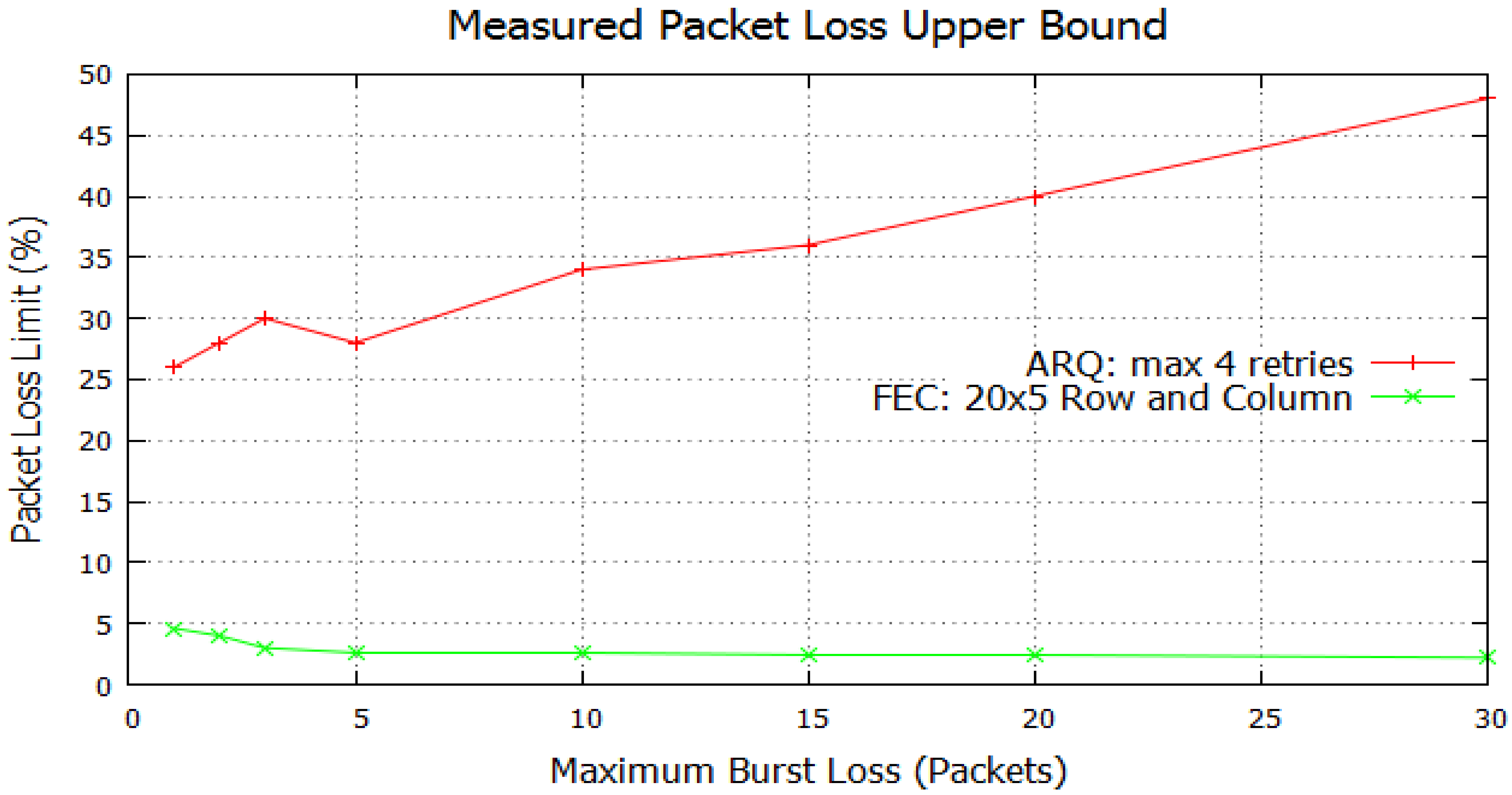
Overhead for ARQ and FEC
Percent Overhead
0.100
0.010
0.001
0.1
1.0
10.0
Percentage Packet Drop (p)
Number of Requests/Packets
M = 10, N = 10
M = 5, N = 20
M = 5, N = 5
N = 10
N = 20
N = 5
R = 1
R = 2
R = 4

Network Simulator

- Windows-based network simulator custom-built for this test
- Random drops, random burst drop size
- Test scenario:
  - End-to-end real-time video
  - Select max burst loss
  - Increase loss percentage until video is “not watchable” (subjective)



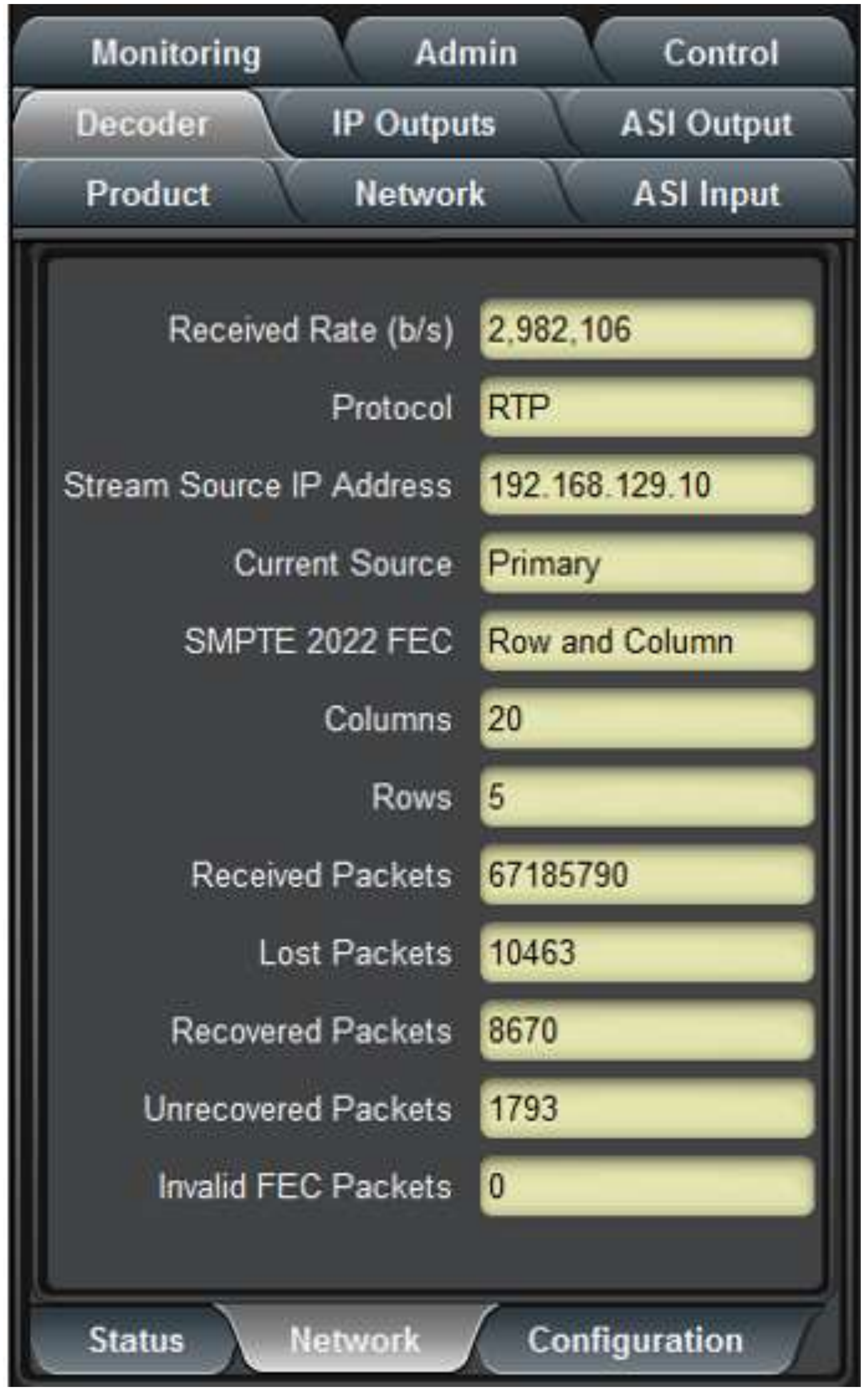
Simulator Results



Field Test Data

- Locations:
  - Santa Clara, CA
  - Champaign, IL
- ISP: Comcast
- Network Round Trip Time: 75 ms
- Number of hops: 12
- Target bit rate: 3 Mb/s
- Equipment:
  - 9223 Encoder
  - 9990-DEC Decoder

RTP/SMPTE-2022 Test Data



Parameters: 20x5 matrix, row and column

Test Duration	65 hours
Test Start Date	05/19/17, 3:50PM
Network Packet Loss	0.0158%
Corrected Packet Loss	0.0027%
Correction Ratio	83%
Bandwidth Overhead	25%
Network Glitch Interval	1 minute 13 seconds
Corrected Glitch Interval	7 minutes 12 seconds
Protocol Latency	702 ms

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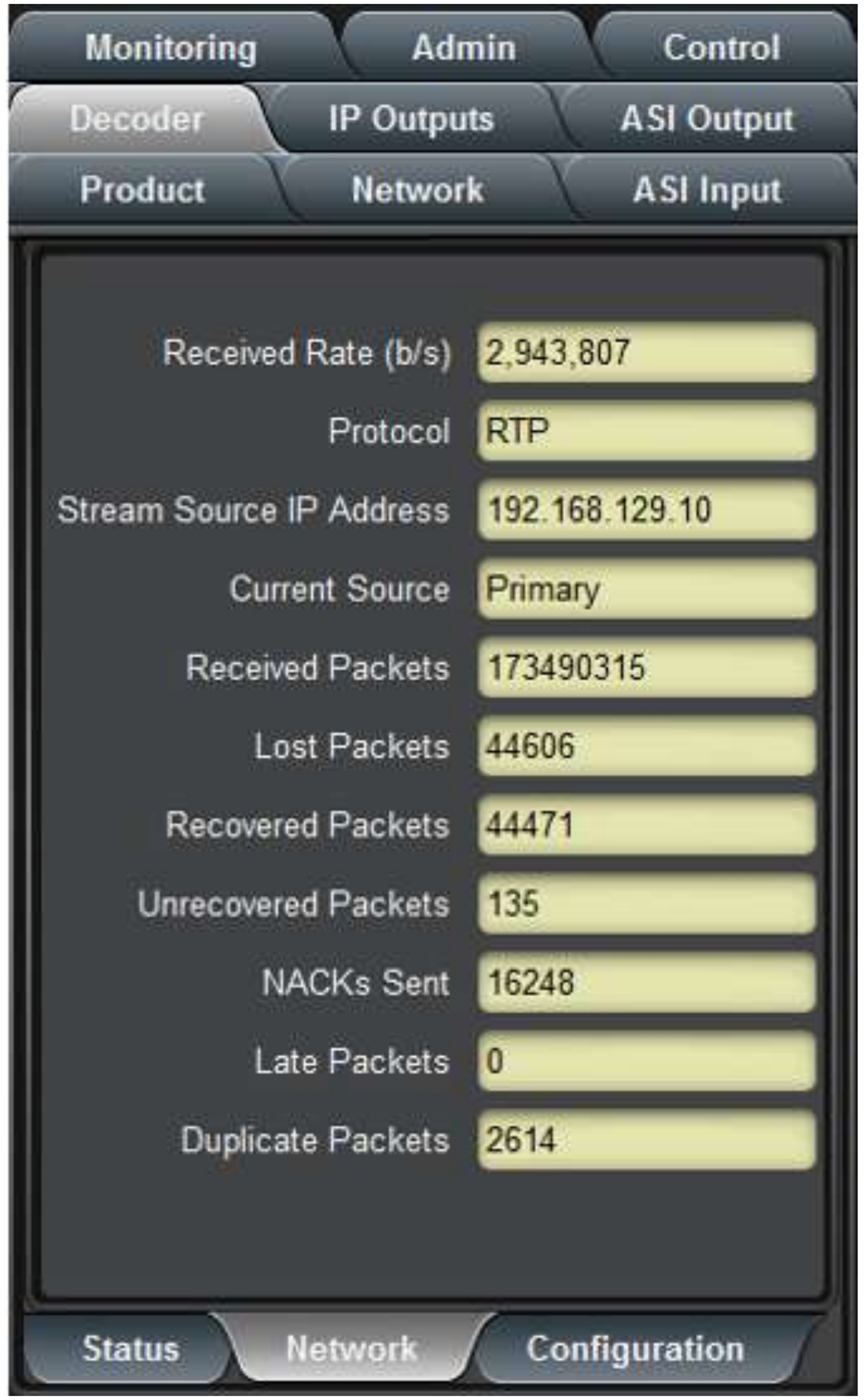
RTP/ARQ Test Data



Parameters: up to 4 retries allowed

Test Duration	169 hours
Test Start Date	05/24/17, 12:30PM
Network Packet Loss	0.0257%
Corrected Packet Loss	0.000078%
Correction Ratio	99.7%
Bandwidth Overhead	0.027%
Network Glitch Interval	46 seconds
Corrected Glitch Interval	4 hours 7 minutes
Protocol Latency	400 ms

FEC/ARQ Comparison

Scaling:

- Latency
  - ARQ latency is constant
  - FEC latency decreases with increasing bit rate
- Overhead
  - ARQ overhead will increase with packet loss
  - FEC overhead is constant

Parameter	2022 FEC	ARQ
Network Packet Loss	0.0158%	0.0257%
Corrected Packet Loss	0.0027%	0.000078%
Correction Ratio	83%	99.7%
Bandwidth Overhead	25%	0.027%
Network Glitch Interval	1 minute 13 seconds	46 seconds
Corrected Glitch Interval	7 minutes 12 seconds	4 hours 7 minutes
Protocol Latency	702 ms	400 ms

ARQ Standardization Status

- The Video Services Forum (VSF) started a group around NAB 2017 to standardize a low-latency video transport protocol over the Internet
- **RIST**: **R**eliable **I**nternet **S**tream **T**ransport
- ARQ has been selected as the base protocol
- VSF TR-06-1 was published October 2018

RIST
RELIABLE INTERNET STREAM TRANSPORT

Thank You

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