An Overview of Compressed Video Transport Protocols over IP

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"The nice thing about standards is that you have so many to choose from."

Andrew Tanenbaum, *Computer Networks*, 2nd ed., p 254

- There are many protocol options in use today for transporting compressed video over an IP network
- We will present an overview of these options, and discuss where they are best applied

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Video Delivery Parameters

- A compressed video delivery protocol may need to care about:
  - Are we delivering to one or multiple receivers?
  - Is network latency a concern?
  - Will the network reliably deliver the packets, or will there be packet loss? (some networks drop packets…)

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The Effect of Packet Loss

- Video compression works by removing redundancy from the transmission
  - Every bit of compressed video is very important
- There is a simple way to look at the effect of packet loss:
  - Every packet that is dropped by the network causes a glitch in the video

What is “acceptable” loss?

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

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Protocol Basics

End-to-end IP applications run on top of one of two protocols:

- User Datagram Protocol (UDP)
  - “Raw” network service
  - Packets are delivered as fast as possible, but may be dropped
  - Support for multicast (network replicates the packets)
- Transmission Control Protocol (TCP)
  - “Reliable” network service
  - Flow control, unbounded latency
  - Unicast only (sender has to replicate for multiple recipients)

The Tradeoff

- Fundamentally, there is a tradeoff between LATENCY and PACKET LOSS RESILIENCY:
  - Decoders cannot “wait forever” – packets have expiration dates
  - You can give yourself time to deal with packet loss by pre-buffering before the decoder – the more time you give yourself, the better job you can do to recover from lost packets

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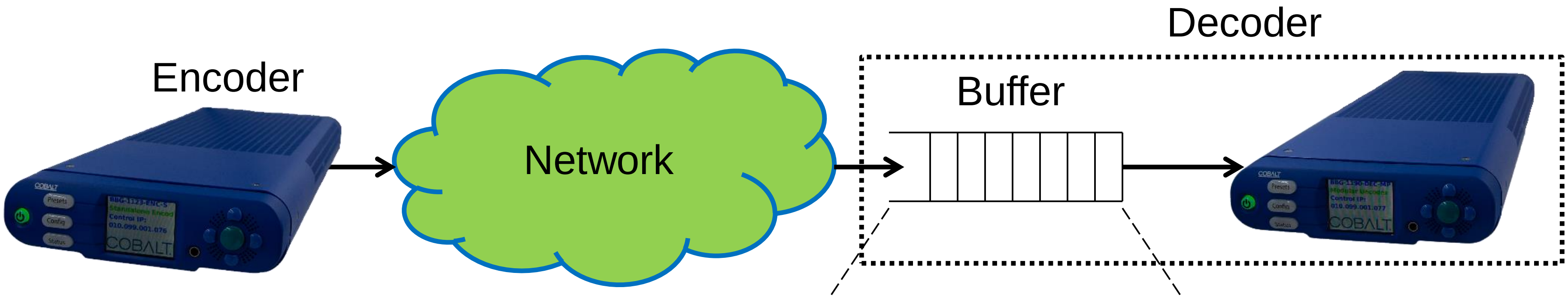
The Tradeoff



Protocol Latency

Gives you time to recover from lost packets – the more time you have, the better job you can do!

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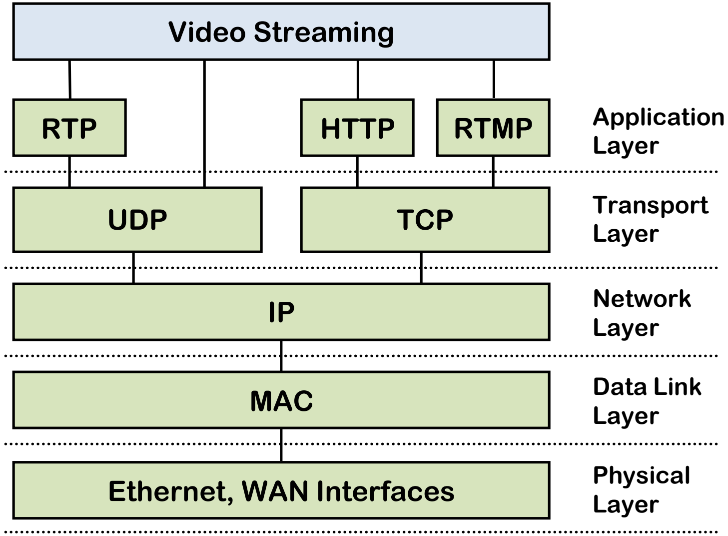
Protocol Roadmap

- UDP-based protocols
  - Raw UDP
  - RTP
  - RTP with SMPTE 2022 FEC
  - RTSP
  - SRT
  - RIST
- TCP-based protocols
  - RTSP (tunnel mode)
  - RTMP
  - HLS



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Raw UDP

- Very simple: just transmit the video in the payload of UDP packets
- Characteristics:
  - Zero protocol latency
  - No packet loss recovery (best effort)
  - Multicast support
- Decoder support: lowest common denominator, supported by professional decoders, IP set-top boxes, software decoders, etc.

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RTP

- Thin layer on top of UDP, adding timestamps and sequence numbers
- Characteristics:
  - Zero protocol latency
  - No packet loss recovery (best effort)
  - Multicast support
  - Capable of packet re-ordering (currently not very useful)
- Decoder support: mostly professional IRDs and some software decoders

RTP Plus SMPTE 2022 FEC

- Video is sent using standard RTP
- Additional FEC packets are also sent using RTP
- If there is packet loss, the receiver MAY be able to rebuild the lost packets from the received packets and the FEC packets.
- The FEC protocol parameters allow a certain amount of tuning of overhead, latency, and recovery capabilities.

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RTP Plus SMPTE 2022 FEC

- Characteristics:
  - Non-zero, tunable protocol latency
  - Multicast support
  - Packet re-ordering support
  - May add significant overhead (typical 25%)
  - May be able to work over the Internet
    - Depends on ISP capacities, congestion, and other factors – it is a risk!
- Decoder support: limited mostly to professional IRDs

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FEC Examples

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

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RTSP

- Real Time Streaming Protocol (RTSP) is really a control protocol typically implemented in video servers
  - It exposes a “VCR-like” control interface to start, pause, stop playback
- The RTSP control interface is implemented over TCP
- Using the control interface, the client (decoder) negotiates the streaming parameters
- The actual streaming is done using plain RTP
  - Video and Audio are sent as elementary streams on separate ports
  - Usually no packet loss recovery at the RTP level

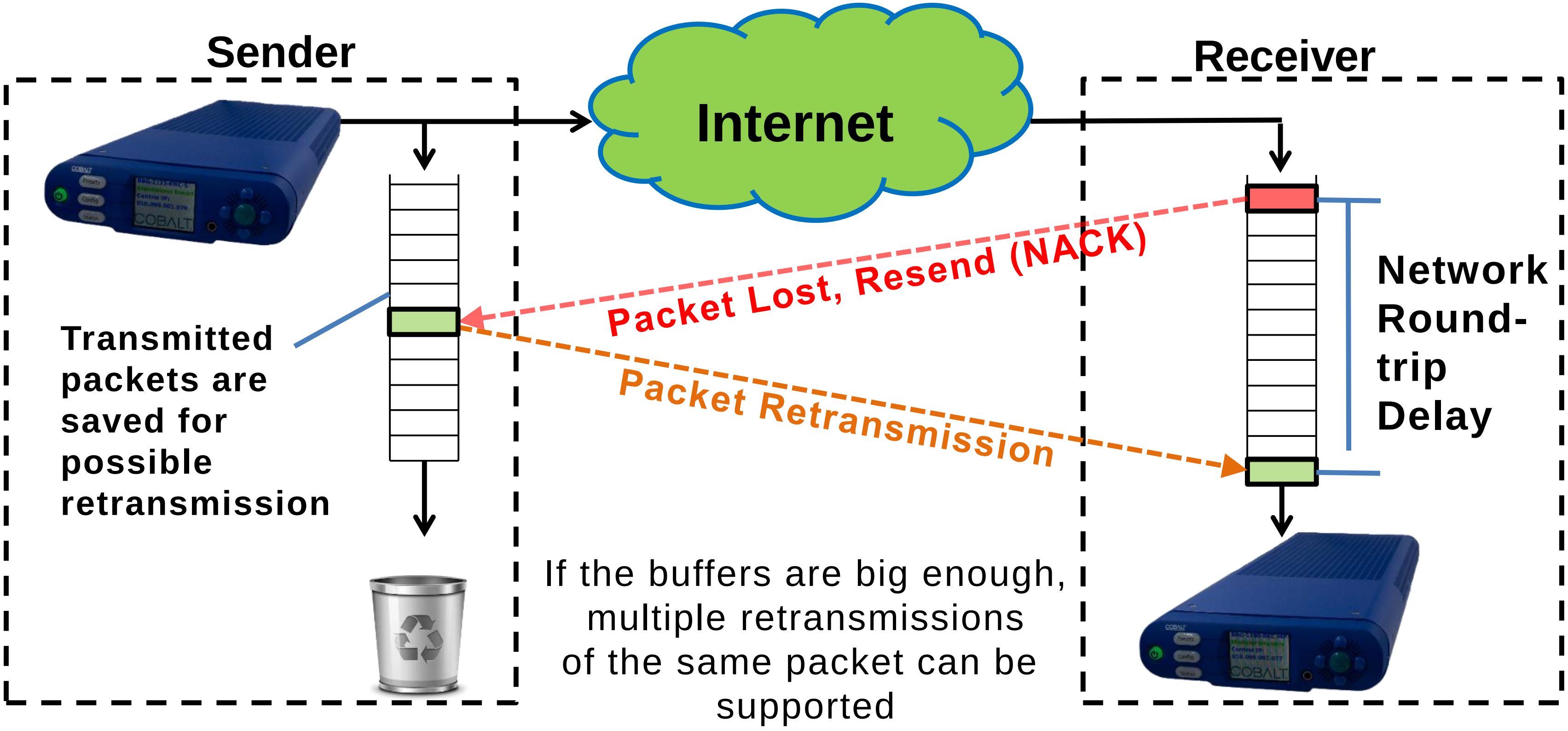
SRT

- Secure Reliable Transport (SRT) is a proprietary protocol developed by Haivision and later placed in the public domain
- Protocol is based on UDT (UDP-based Data Transfer Protocol)
  - Protocol designed for high-speed file transfer over UDP
- Operation:
  - Packets received correctly are acknowledged (similar to TCP)
  - There is an explicit NACK for dropped packets
  - Sender retransmits requested packets or un-acknowledged packets
- Device support:
  - Available in a number of professional encoders and IRDs
  - Available in the VLC software player

RIST

- Reliable Internet Stream Transport (RIST) is a specification published by the Video Services Forum intended for low-latency video contribution/distribution
- Lost packets are recovered using a variant of Selective Retransmission (ARQ – Automatic Repeat reQuest)
- Highlights:
  - Media transmission is done using standard RTP/UDP
  - Packets received correctly are not acknowledged (no flow control)
  - Receiver requests retransmission of lost packets using standard RTCP messages
  - Designed to be firewall-friendly

RIST Illustration



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RIST Discussion

- In RIST, the latency-reliability tradeoff is fully configurable by the choice of the buffer and number of times a packet can be retried
  - Latency of the protocol can be fine-tuned for the network conditions
- Using RTP as the base protocol ensures compatibility with non-RIST devices
- Supported in a number of encoders, decoders, and gateways from multiple vendors
  - Multi-vendor interoperability
- Supported in the VLC public-domain software decoder

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TCP-Based Transport

- Quick review of TCP:
  - Connection-oriented: a client explicitly connects to a server; data transmission can go in either direction (or both ways)
  - No multicast support
  - Protocol uses acknowledgments and retransmission to make sure that all bytes are received, *no matter how long it takes*
  - Protocol also provides flow-control – receiving side only acknowledges the data when it is ready to receive more
    - The ability to flow-control an encoder is limited to non-existent
  - Flow control is also used for network congestion

Encoding to a TCP Connection

- The simplest use of TCP is to create a connection between the encoder and the device that is consuming the stream
- Encoder pushes the data through the connection and hopes that the end-to-end bandwidth is enough
  - Buffering required at the encoder …
- Protocols using a raw TCP connection:
  - RTSP (tunnel mode)
  - RTMP

RTSP, Tunneled

- Basic RTSP is only suitable for local managed networks
  - No packet loss recovery on RTP
  - UDP ports are dynamically negotiated – not firewall friendly
- RTSP has a mode where the RTP data is tunneled over the TCP control connection
  - Same resiliency as TCP, single connection
- Encoder support: mostly built-in encoders in surveillance cameras
- Decoder support: software decoders, some professional IRDs

Real Time Messaging Protocol (RTMP)

- Proprietary protocol designed by Macromedia for its Flash player (later acquired by Adobe)
- Protocol specification was placed in the public domain by Adobe
- Used primarily by Flash players to retrieve content from servers
- Protocol has an option for the client to publish a stream to the server – this is what encoders use
- Protocol is becoming obsolete as it is media-specific
  - Out of the modern encoding standards, only supports H.264 and AAC audio
  - Limited to Flash container format
  - Not very well documented

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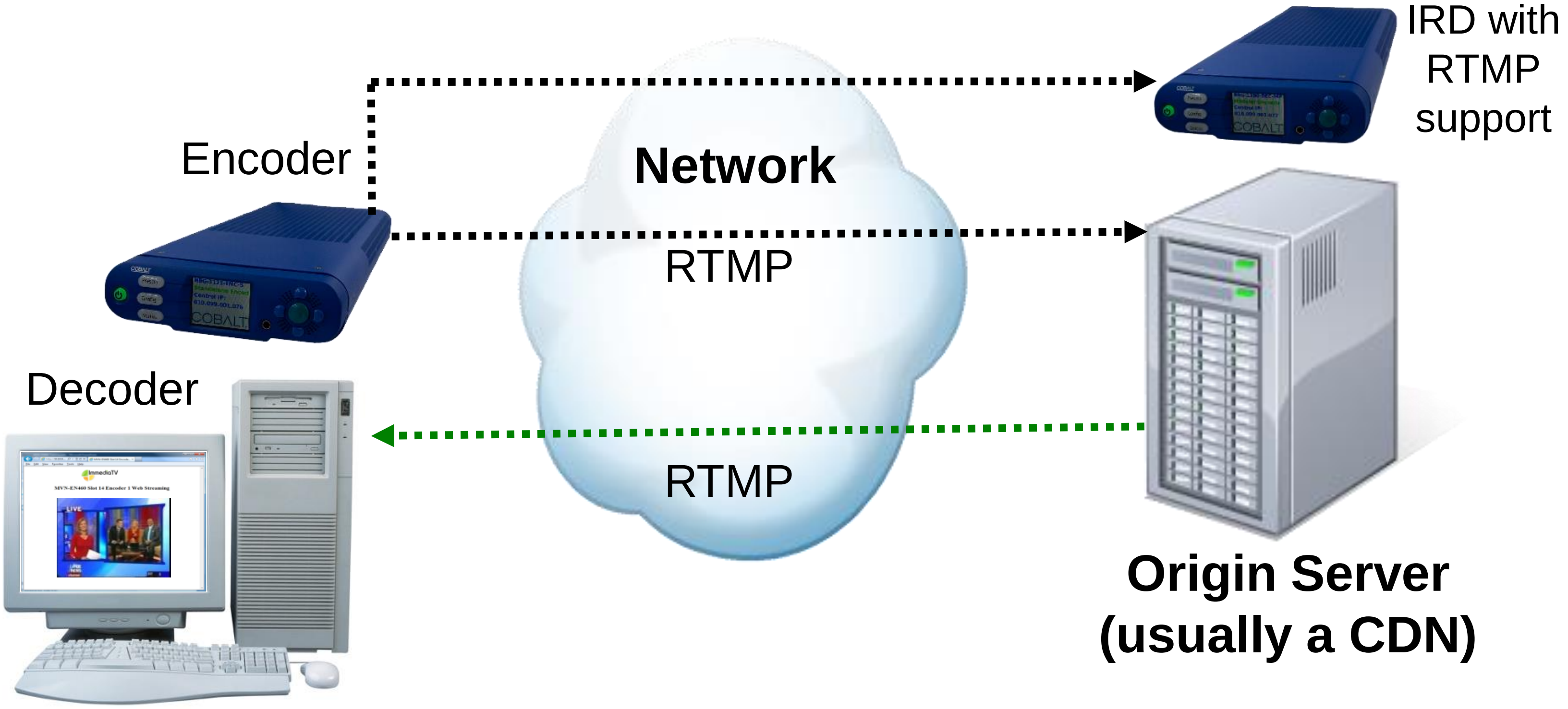
RTMP Operation



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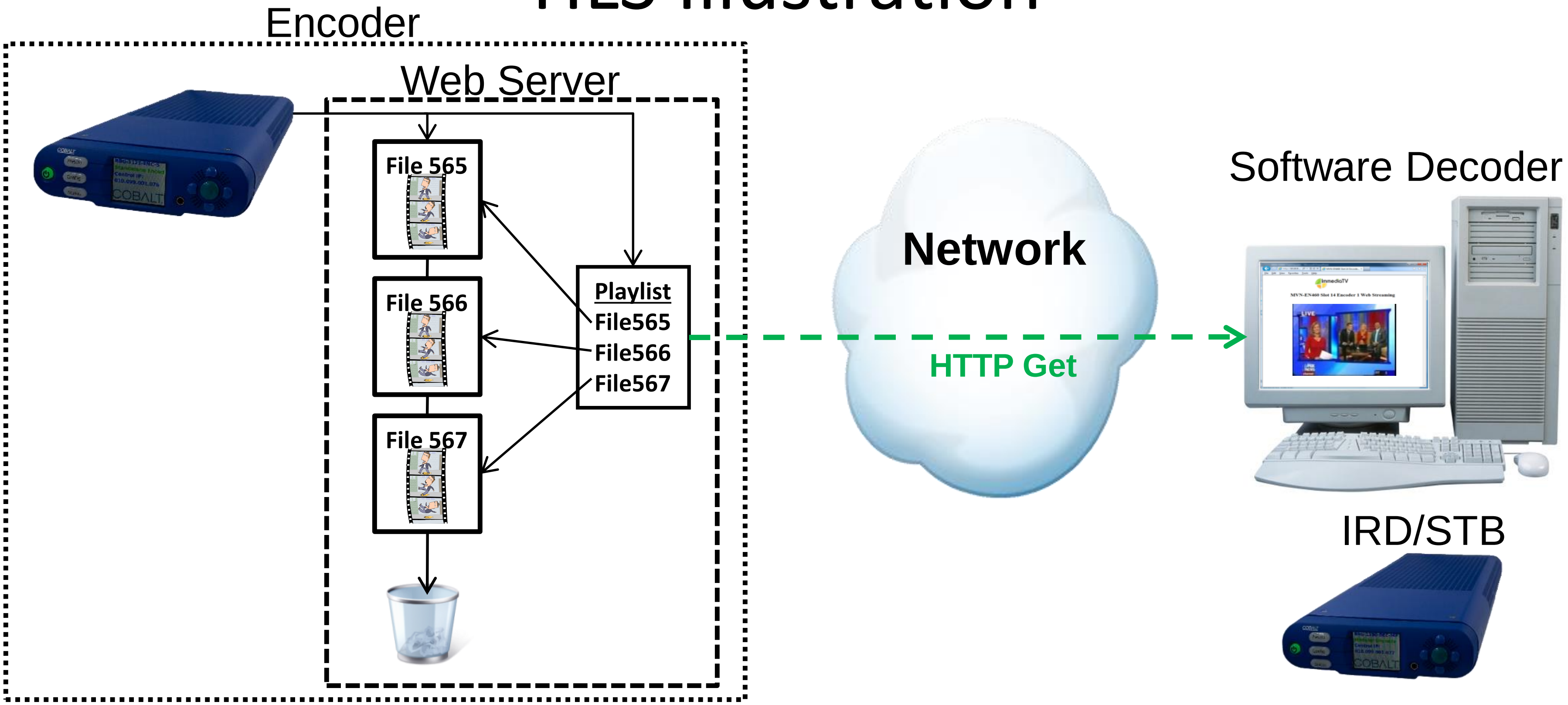
RTMP Discussion

- Characteristics:
  - Latency will depend on what processing is done in the server – typically on the order of several seconds or more
  - Resilient to packet loss (uses TCP)
  - Scalability is done at the server (commercial products by Adobe, Wowza, and open-source variants)
  - De facto standard for publishing live streams in the Internet
    - Industry is starting to move away from it as the protocol is obsolete
- Decoder support: Software decoders, some IRDs
  - Servers usually do protocol conversion for other decoders

HTTP Live Streaming (HLS)

- HLS is a protocol designed by Apple to provide streaming using a standard (unmodified) web server
- The video stream is divided into “chunks” of a few seconds each
- The decoder downloads the chunks as files from the web server with standard HTTP transactions, using a playlist
- Protocol supports adaptive streaming (multiple bit rates)
- Encoder can publish to a local (built-in) web server or to a remote server using HTTP PUT/POST
- MPEG-DASH is similar

HLS Illustration



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HLS Details

- Characteristics:
  - Very high latency: 3-4 times the chunk size (which varies from 2 to 30 seconds)
  - Extremely robust
  - Scalability can be done using external web servers
  - No TCP flow-control issue on the encoder side when publishing locally
- Decoder support: native support on all Apple and Android devices; supported in a number of IP set-top boxes and some professional IRDs

Protocol Comparison Matrix

	Low Latency	Medium Latency	High Latency
High Packet Loss Resiliency		RIST SRT	HLS/DASH RTMP/RTSP
Medium Packet Loss Resiliency	RTP+FEC		
No Packet Loss Resiliency	RTP/RTSP UDP		

TCP-based

UDP-based

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Q&A

- Questions?
- Thanks!

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