Networking

All you need to know

Contents

- Our Philosophy
- Networking is at our heart
- Why Video over IP?
- OSI and the protocol Stack
- The importance of the transport layer
- Application layer protocols
- Codecs v Protocols
- Streaming
- New SMPTE Standards
- Security
- Ports
- Basics
- Free Course

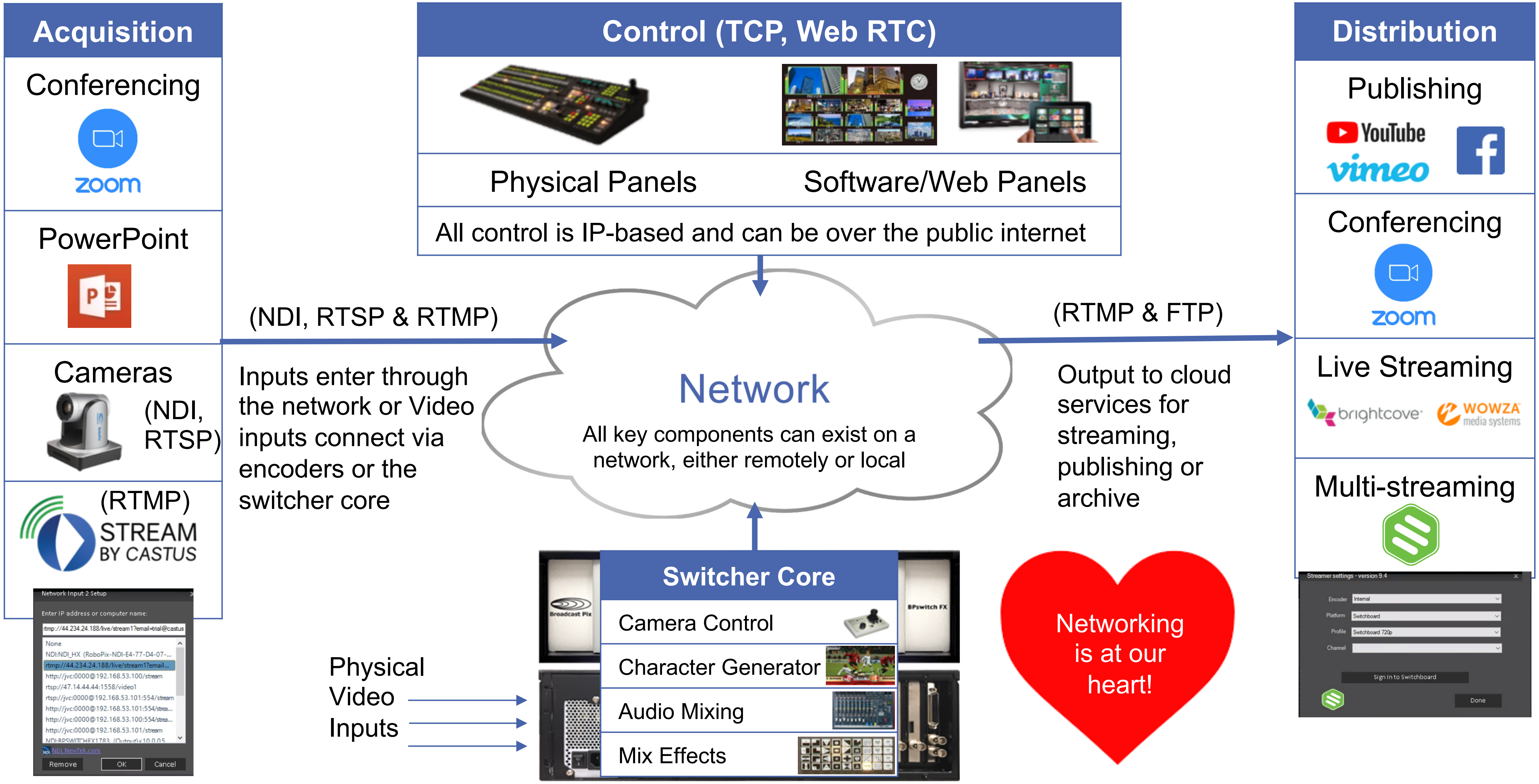
Our North Star..

- Simplify the complex

Our smart production solutions enable users to produce and stream compelling live content easily:

- Everything in the box
- Easy to install
- Repetitive tasks automated
- Easy to use Content mgmt.
- Configurable user interfaces
- Templates & Content included
- Education

Our abstracted architecture



Acquisition

Conferencing

zoom

PowerPoint

Cameras (NDI, RTSP)

(RTMP) STREAM BY CASTUS

Network Input 2 Setup

Enter IP address or computer name:

```
rtmp://44.234.24.188/live/stream1?email=trial@castus
None
NDI:NDI_HX (RoboPix-NDI-E4-77-D4-07-...
rtmp://44.234.24.188/live/stream1?email...
http://jvc:0000@192.168.53.100/stream
rtsp://47.14.44.44:1558/video1
rtsp://jvc:0000@192.168.53.101:554/stream
http://jvc:0000@192.168.53.101:554/strea...
http://jvc:0000@192.168.53.100:554/strea...
http://jvc:0000@192.168.53.101/stream
NDI:BPSWITCHFX1783 (Output)<10.0.0.5
```

(NDI, RTSP & RTMP)

Inputs enter through the network or Video inputs connect via encoders or the switcher core

Control (TCP, Web RTC)

Physical Panels | Software/Web Panels

All control is IP-based and can be over the public internet

Network

All key components can exist on a network, either remotely or local

Switcher Core

Camera Control

Character Generator

Audio Mixing

Mix Effects

Physical Video Inputs

(RTMP & FTP)

Output to cloud services for streaming, publishing or archive

Networking is at our heart!

Distribution

Publishing

Conferencing

zoom

Live Streaming

Multi-streaming

Streamer settings - version 9.4

Encoder: Internal

Platform: Switchboard

Profile: Switchboard 720p

Channel:

Why Video over IP?

Video as we all know and love it:

- Dedicated cable or interface
- Perfectly timed and synchronous
- Full bandwidth Video signal

Video over IP:

- General purpose, not dedicated network
- Asynchronous network that's not perfectly timed
- We have to chop up the video to fit
- Due to limited bandwidth, the Video is compressed

Why do it?

Are we Crazy?

Broadcast Pix

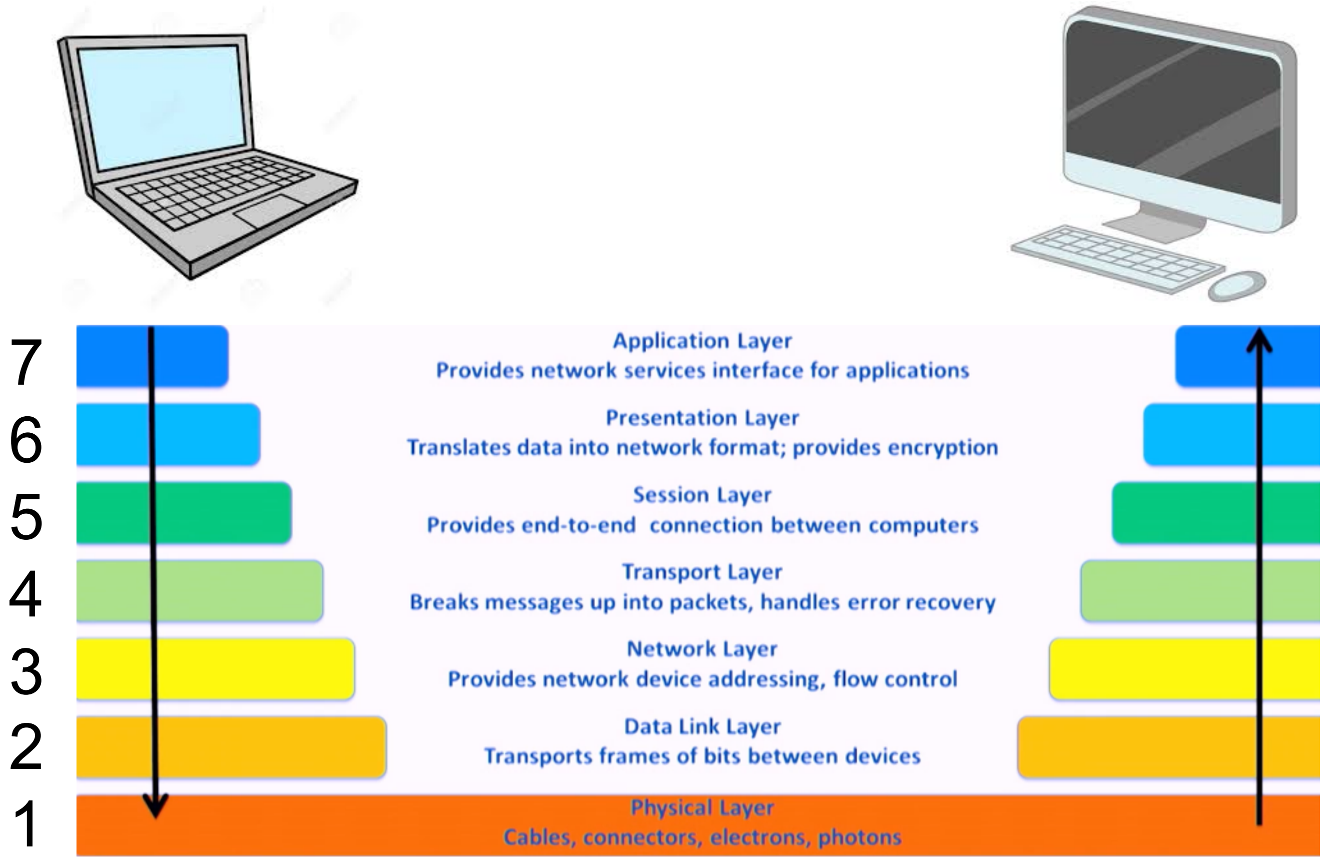
Advantages of IP

Advantage	Notes:
$’s Cost effective	Consumer off the shelf technology is good value because of the volume sold
Easy to install	Robotic cameras need 1 cable for video, control and power (POE)
Highly Scalable	Utilizes off the shelf switches and infrastructure Any number of IP inputs can exist on the network and be selected from the application
Secure & manageable	Video and other content takes advantage of the security and management features of IP
Distance not an issue	IP extends beyond the facility to the WWW with the use of a suitable router
Flexibility of integration	Its easy to bring in digital files and other computer-based applications
Speed of deployment	Infrastructure is readily available and there is a large pool of knowledgeable technicians

The entire Broadcast Technology industry is about the same size as HP’s High-Performance Computing Division

=> The size of the Broadcast Technology industry is almost insignificant when compared to the total IT industry

Open Systems Interconnect (OSI)



Application Layer
Provides network services interface for applications

Presentation Layer
Translates data into network format; provides encryption

Session Layer
Provides end-to-end connection between computers

Transport Layer
Breaks messages up into packets, handles error recovery

Network Layer
Provides network device addressing, flow control

Data Link Layer
Transports frames of bits between devices

Physical Layer
Cables, connectors, electrons, photons

Please Do Not Throw Sausage Pizza Away

- **OSI** is a conceptual model that standardizes the communication functions between computing systems with the goal of interoperability by using standard communication protocols
- It partitions the flow of data into seven abstraction layers, from the lowest, physical implementation to the highest, the application
- Each intermediate layer provides functionality to the layer above it by adding additional header information and becomes more abstract to the layer below it by removing header information
- Development of the model started in the late 70s and became a working product of the International Organization for Standardization (ISO) in the 1980’s

What Is a Protocol?

A protocol is a set of rules governing how data travels from one communicating system to another. These are layered on top of one another to form a protocol stack. That way, protocols at each layer can focus on a specific function and cooperate with each other. The lowest layer acts as a foundation, and each additional layer adds complexity.

For example, you’ve likely heard of an IP address, which stands for Internet Protocol. This protocol structures how devices using the internet communicate. The Internet Protocol sits at the network layer and is typically overlaid by the Transmission Control Protocol (TCP) at the transport layer, as well as the Hypertext Transfer Protocol (HTTP) at the application layer.

Simplified Protocol Stack

<table>
<tr><td>7</td><td>APPLICATION</td><td colspan="2" rowspan="3">The functionality of most applications we use, map to the Application, Presentation and some of the Session layer
The remaining session layer functionality is part of the Transport layer</td><td colspan="4">The application layer is closest to the end user and interacts with software applications that require a communicating component</td></tr>
<tr><td>6</td><td>PRESENTATION</td><td colspan="4">The presentation layer establishes context between application-layer components and maps any different use of syntax and semantics</td></tr>
<tr><td>5</td><td>SESSION</td><td colspan="4">The session layer controls the connections between computers and the the local and remote application</td></tr>
<tr><td>4</td><td>TRANSPORT</td><td colspan="2">TCP Transmission Control Protocol</td><td colspan="2">UDP User Datagram Protocol</td><td colspan="2">RTP Real Time Transport Protocol</td></tr>
<tr><td>3</td><td>NETWORK</td><td colspan="6">IPv4 and IPv6, DNS, Subnets, DCHP</td></tr>
<tr><td>2</td><td>DATA</td><td colspan="6">Ethernet: Message delineation, Medium Access Control, Device addressing, Error Correction</td></tr>
<tr><td>1</td><td>PHYSICAL</td><td>DOCSIS
Used by Cable Companies</td><td>DSL
Used by Telco’s</td><td>CELLULAR
Wireless</td><td>PHYSICAL CABLE
Copper, Fibre Optic</td><td>WIFI
Local Wireless</td><td>BLUETOOTH
Low Power Wireless</td></tr>
</table>

There are 3 main protocols used in the Transport Layer (4) are they critical to the design and performance of Video Networks;

TCP, UDP and RTP

Broadcast Pix

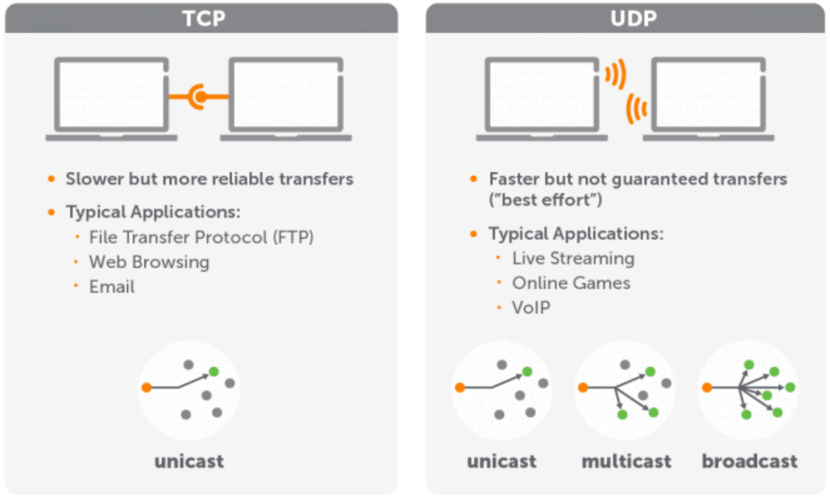
Transport Layer protocols

PROTOCOL	TCP Transmission Control Protocol	UDP User Datagram Protocol	RTP Real Time Transport Protocol
SUMMARY	Data is transmitted in the form of packets between systems over a network. It includes error-checking, which guarantees the delivery and preserves the order of the data packets.	Like TCP except it doesn't include the error-checking, sequencing and data recovery, so low overhead. Faster and more efficient, but less reliable than TCP	Specifically designed to support media streaming - Built on top of UDP but with additional functionality
DESCRIPTION	A connection is created between sender and receiver and data is exchanged with acknowledgments and retries. Delays often occur due to retransmission – so not great for continuous video	No connection is created – packets of data are 'just sent' irrespective of any issues at a rate determined by the sender. No feedback from receiver, so no retransmission	Adds sequence numbers, payload type and timestamp information to UDP to improve the quality. RTSP further adds a control protocol for feedback from the receiver - but increases overhead
EXAMPLES	HTTP, FTP, email(POP3, SMPT)	SNMP, RIP, RTP	SMPTE 2022, 2021 and AES 67
MULTICASTING	Does not support multicasting	Supports multicasting	Supports multicasting
BENEFIT	Data integrity	Time critical applications, can synchronize streams using timestamp Can cause problems with firewalls, because not a negotiated connection	
IDENTIFICATION	6 in the IP packet header	17 in the IP packet header	

Historically TCP was used for production environments, where quality is more important than timely delivery and UDP was used for streaming, where timely delivery was more important than quality

The dramatically increased performance of Computers and Networking is now blurring those lines

In summary and RTP

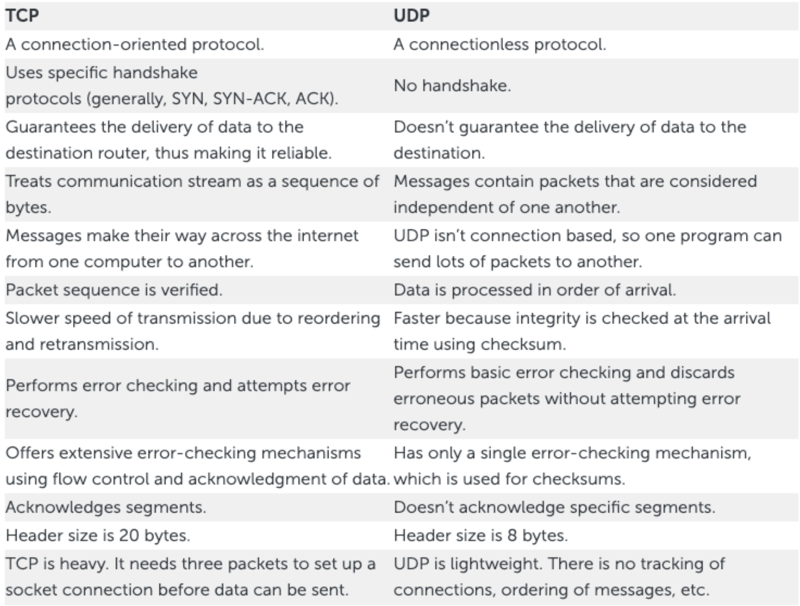


TCP

- Slower but more reliable transfers
- Typical Applications:
  - File Transfer Protocol (FTP)
  - Web Browsing
  - Email

UDP

- Faster but not guaranteed transfers ("best effort")
- Typical Applications:
  - Live Streaming
  - Online Games
  - VoIP



TCP	UDP
A connection-oriented protocol.	A connectionless protocol.
Uses specific handshake protocols (generally, SYN, SYN-ACK, ACK).	No handshake.
Guarantees the delivery of data to the destination router, thus making it reliable.	Doesn't guarantee the delivery of data to the destination.
Treats communication stream as a sequence of bytes.	Messages contain packets that are considered independent of one another.
Messages make their way across the internet from one computer to another.	UDP isn't connection based, so one program can send lots of packets to another.
Packet sequence is verified.	Data is processed in order of arrival.
Slower speed of transmission due to reordering and retransmission.	Faster because integrity is checked at the arrival time using checksum.
Performs error checking and attempts error recovery.	Performs basic error checking and discards erroneous packets without attempting error recovery.
Offers extensive error-checking mechanisms using flow control and acknowledgment of data.	Has only a single error-checking mechanism, which is used for checksums.
Acknowledges segments.	Doesn't acknowledge specific segments.
Header size is 20 bytes.	Header size is 8 bytes.
TCP is heavy. It needs three packets to set up a socket connection before data can be sent.	UDP is lightweight. There is no tracking of connections, ordering of messages, etc.

RTP and RTSP – Real Time Streaming Protocol

Whilst RTP is a transport protocol built upon UDP, it is usually coupled with RTSP, which is a presentation-layer protocol that lets end users command media servers via pause and play capabilities using TCP to maintain an end-to-end connection

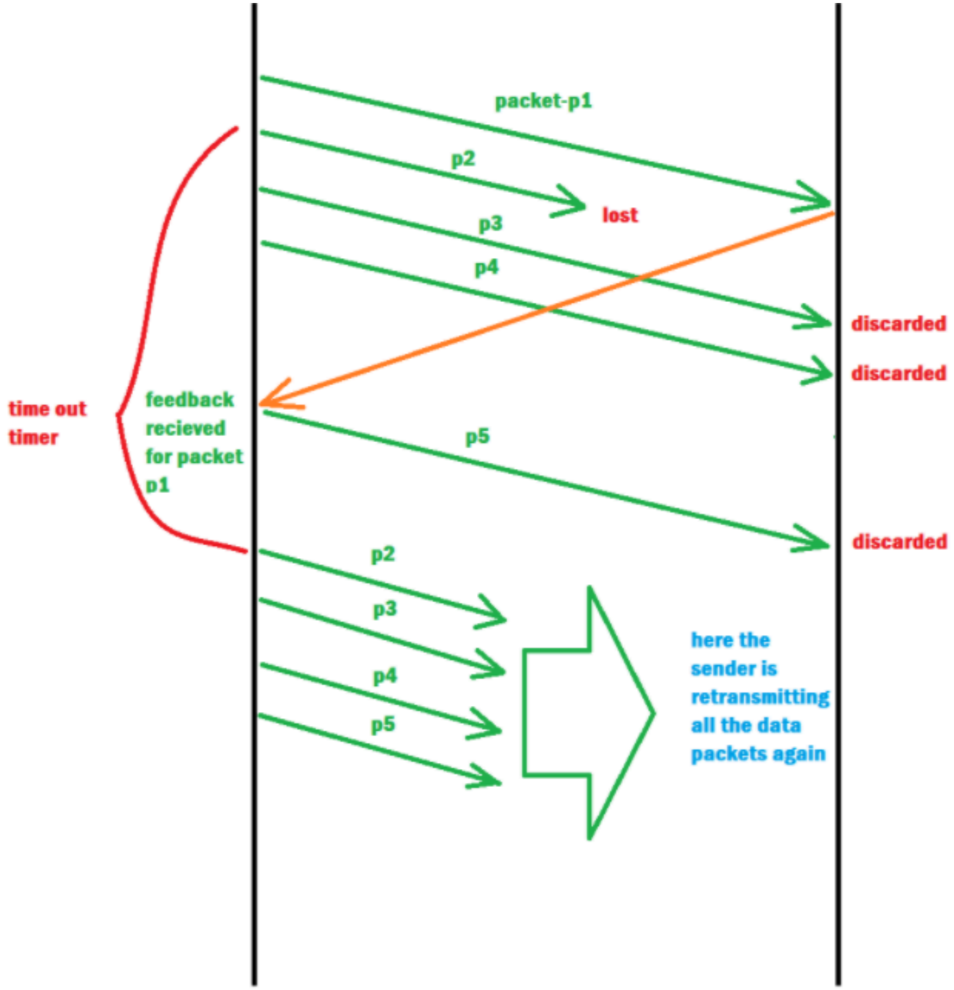
RTSP/RTP is usually used for video contribution as opposed to multi-device delivery and remains standard in surveillance and closed-circuit television (CCTV) architectures mainly because RTSP support is still ubiquitous in IP cameras

When streaming content to users, RTSP frequently relies on a combination of reliable transmission over TCP (for control) and best-efforts delivery over UDP (for content delivery) to display audio/video content on client-side applications before the complete file has arrived

Application Protocols

Protocols	Uses	Description
HTML - Hypertext Markup Language	TCP	Is the standard markup language for documents designed to be displayed in a web browser. HTLM5 introduced the video element for the purpose of playing videos in web pages. HTML is CODEC, or video format agnostic
HTTP - Hypertext Transfer Protocol	TCP	Is the foundation of data communication for the World Wide Web (WWW) where hypertext documents include hyperlinks that can be clicked through.
FTP - File Transfer Protocol	TCP	Uses separate control and data connections between a client and the server. Users authenticate themselves with a username and password, but can connect anonymously if configuration allows
HLS - HTTP Live Streaming	TCP	HTTP-based adaptive bitrate streaming protocol chunks the stream into a sequence of small HTTP-based file downloads. It can traverse any firewall, unlike UDP-based protocols such as RTP.
DASH - Dynamic Adaptive Streaming over HTTP	TCP	MPEG-DASH is an adaptive bitrate protocol like HLS which breaks the content into a sequence of small chunks served over HTTP. Clients can seamlessly adapt to changing network conditions without stalls or re-buffering
RTC - Web Real-time Communication	UDP	Enables direct communication of audio and video from inside web pages, eliminating the need to install plugins or applications. Has applications for non-browser devices, including, video conferencing, mobile etc.
RIST – Reliable Internet Streaming Protocol	RTP	Open specification, RIST provides reliable, high performance media transport by using RTP / UDP at the transport layer to avoid the limitations of TCP. Reliability is achieved by using NACK-based retransmissions ARQ
SRT – Secure Reliable Transport	UDP	Provides TCP type reliability using UDP without sacrificing latency. Originally designed for fast reliable file transmission, SRT added additional features in order to support live streaming
NDI – Network Device Interface	TCP UDP	NDI uses mDNS for self discovery. When a source is requested, a TCP connection is established on the appropriate port. NDI 3.x has options to use UDP multicast or unicast with forward error correction (FEC) instead of TCP.
RTMP – Real Time Messaging Protocol	TCP	RTMP is a TCP-based protocol that powers Flash. It splits streams into fragments, their size is negotiated dynamically between the client and server. It encapsulates MP3 or AAC audio and FLV1 video

What is ARQ?



Automatic repeat request (query) is an error-control method for data transmission using acknowledgements and timeouts to achieve reliable data transmission over unreliable communication channels (such as UDP)

- If the sender does not receive an acknowledgment before the timeout, it re-transmits until it receives one or exceeds a specified number of retransmits
- Variants include Stop-and-wait ARQ, Go-Back-N ARQ, and Selective Repeat/Selective Reject ARQ
- All three protocols usually use some form of sliding window to help the transmitter determine if any packets need to be retransmitted
- These protocols reside in the data link or transport layers (layers 2 and 4) of the OSI model

Codecs v Protocols

EnCoder Compress

Decoder Decompress

An encoder encodes or compresses the Video for transmission or storage and the decoder function reverses the encoding or decompresses the Video for playback or editing

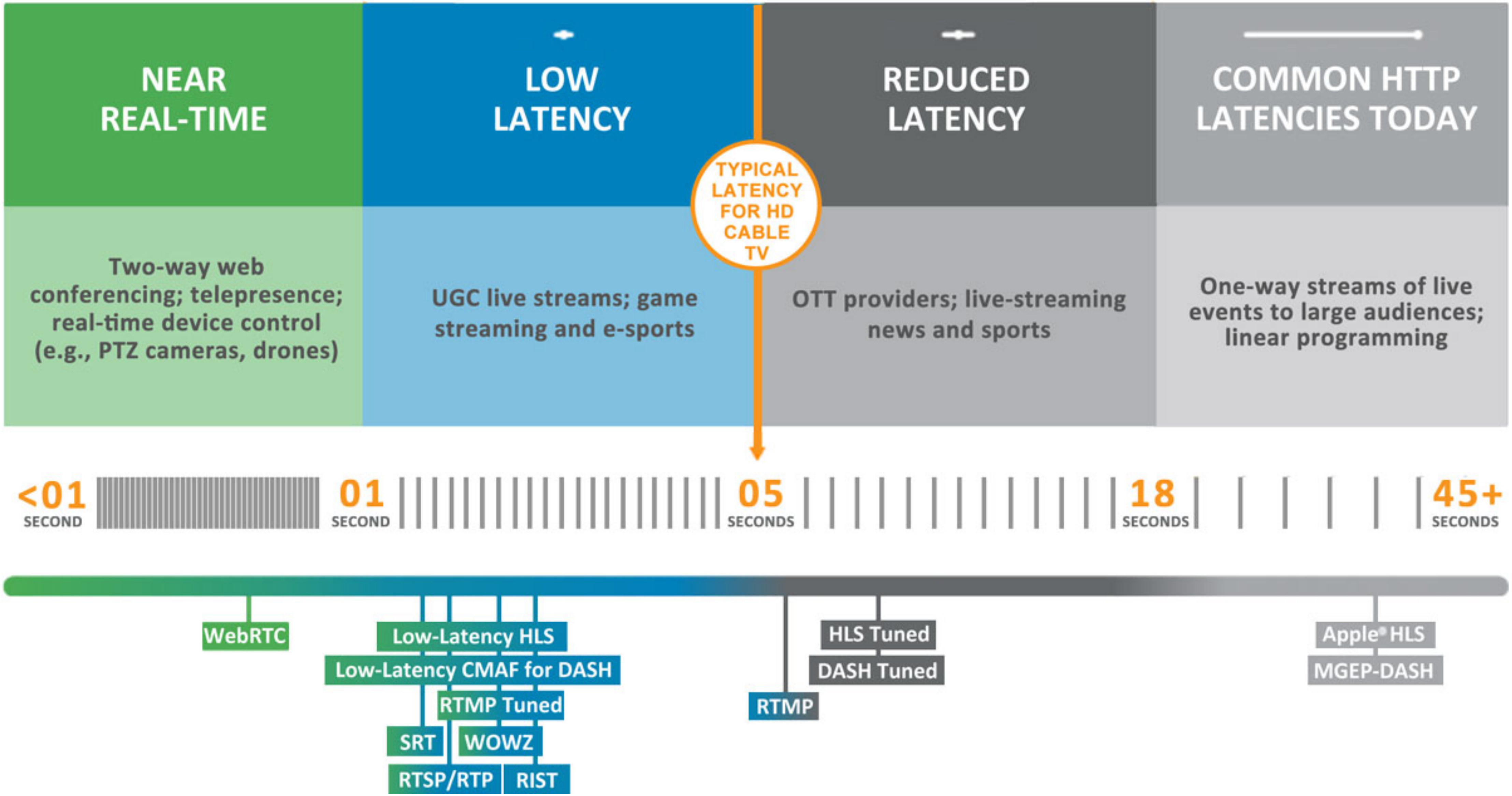
There are two main types of Compression, MPEG and JPEG and their attributes are listed in the table

Protocols wrap the compressed Video to facilitate its transmission across a network

The video has to be compressed in order to be transmitted across a general-purpose network in a timely way

Codec	MPEG Video	JPEG Video
Description	This is interframe compression – i.e. the algorithms transmit movement over multiple frames	This is intraframe compression – i.e. the algorithms operate between fields in a single frame
Attributes	Higher compression rates, high latency, difficult to edit / produce	Lower compression rates, low latency, easy to edit or produce
Use	Transmission where no production / packaging is needed	Contribution where production / packaging is required
Examples	MPEG 2. MPEG 4 H.264. H.265 AV1 HEVC VP6. VP7. VP8. VP9 (Google)	JPEG J2000 and derivatives

Streaming latency continuum



NEAR REAL-TIME

Two-way web conferencing; telepresence; real-time device control (e.g., PTZ cameras, drones)

LOW LATENCY

UGC live streams; game streaming and e-sports

TYPICAL LATENCY FOR HD CABLE TV

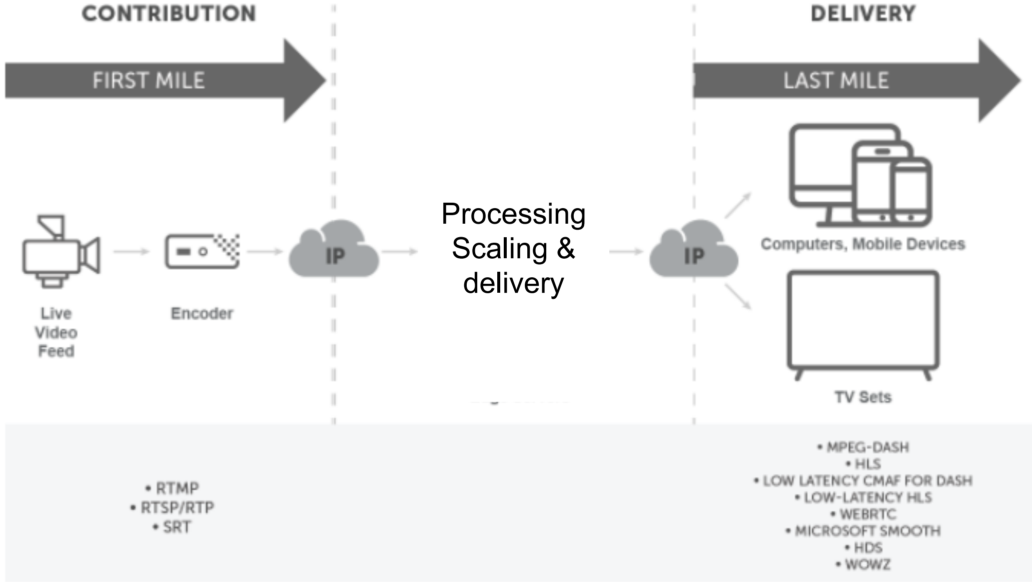
REDUCED LATENCY

OTT providers; live-streaming news and sports

COMMON HTTP LATENCIES TODAY

One-way streams of live events to large audiences; linear programming

Streaming Protocols



Each time you watch a live stream or video on demand, streaming protocols are used to deliver data over the internet. These can sit in the application, presentation, and session layers.

Online video delivery uses both streaming and HTTP-based protocols.

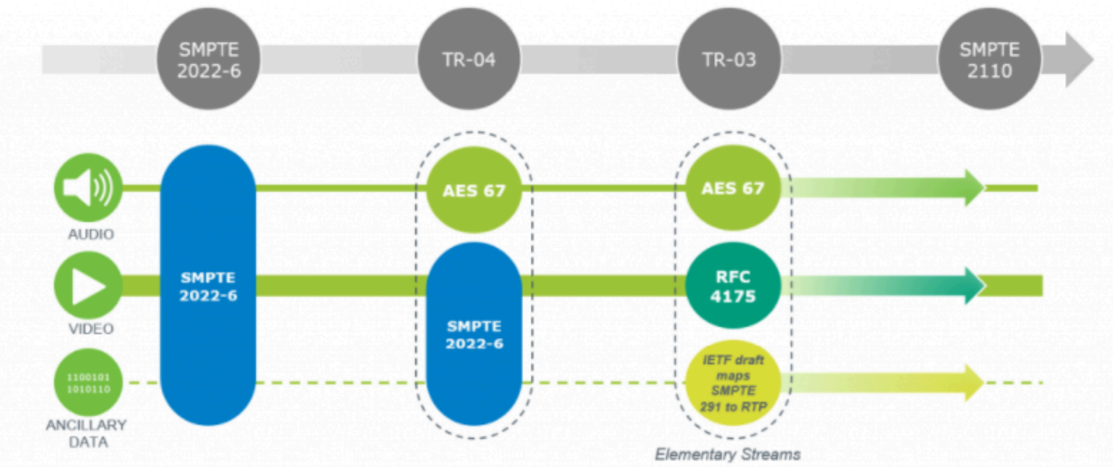
Streaming protocols like RTMP enable speedy video delivery using dedicated streaming servers, whereas HTTP-based protocols rely on regular web servers to optimize the viewing experience and quickly scale.

There are also a handful of emerging HTTP-based technologies like the MPEG DASH and HLS seek to deliver the best of both options to support low-latency streaming at scale

Considerations When Choosing a Streaming Protocol

- Selecting the right protocol starts with defining what you’re trying to do.
- Latency, playback compatibility, and viewing experience can all be impacted.
- Many operators use different protocols for different steps; RTMP and SRT are great for first-mile contribution, while both DASH and HLS lead the way when it comes to playback. But you may be looking to deploy a one-to-few conference, in which case WebRTC would be better suited
- Streams deployed over HTTP are not technically “streams.” Rather, they’re progressive downloads sent via regular web servers.
  Using adaptive bitrate streaming, HTTP-based protocols deliver the best video quality and viewer experience possible — no matter the connection, software, or device. Some of the most common HTTP-based protocols include MPEG-DASH and Apple’s HLS

New SMPTE standards



SMPTE 2022 is a suite of standards that describes how to send digital media, including SDI over an IP network using RTP:

- Video formats supported include from MPEG-2 to uncompressed Serial Digital Interface
- **HBRMT** High bit rate media transport is a standard for data encapsulation and forward error correction (FEC) of high bit rate contribution-oriented services up to 3 Gbit/s over Ethernet networks and is designed to incorporate both SDI uncompressed and JPEG 2000 compressed video and audio formats

SMPTE 2110 is a suite of standards that describes the carriage, synchronization and description of separate elementary essence streams over an IP network for live production:

- Individual audio, video and ancillary data tracks or clips are carried as separate individual streams referred to as "essences”.
  - E.g. 5.1 JPEG mp4 clip could have 9 essences: a video, 6 separate audio & 2 close caption essences, English and Chinese
- Real Time Transport (RTP) is used to transmit streaming essences
- Session Initiation Protocol (SIP) manages the connection and distribution of RTP streams including multicast
- Precision Time Protocol (PTP) provides global micro-second accuracy timing of all essences

Transport Layer Security (TLS)

Transport Layer Security, or TLS, is a widely adopted security protocol designed to facilitate privacy and data security for communications over the Internet

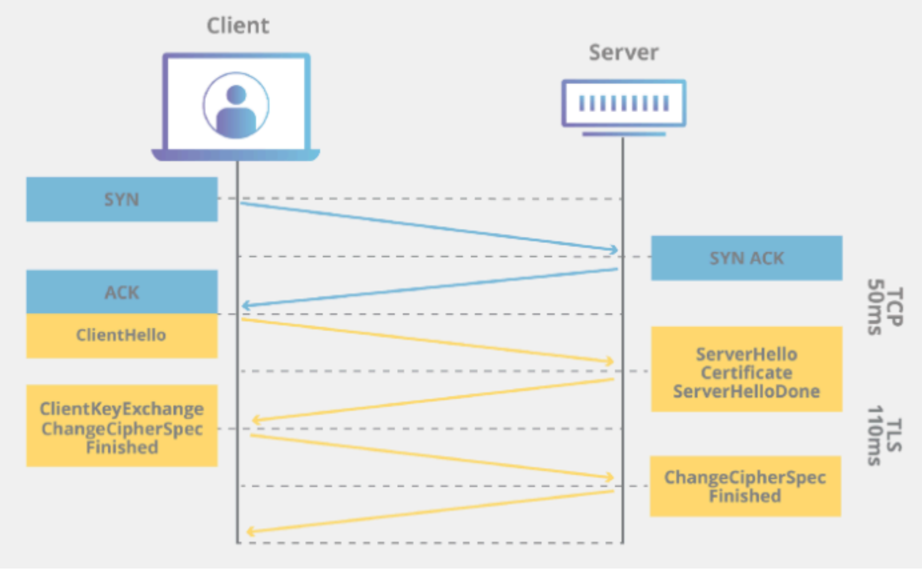
A primary use case of TLS is encrypting the communication between web applications and servers, such as web browsers loading a website

TLS can also be used to encrypt other communications such as email, messaging, and VoIP so once Video is being transported within the IP domain, these security techniques can be deployed

HTTPS is an implementation of TLS encryption on top of the HTTP protocol, which is used by all websites and many web services

TLS encryption protects web applications from data breaches and other attacks; There are three main components to what the TLS protocol accomplishes:

- Encryption: hides the data being transferred from third parties
- Authentication: ensures the parties exchanging information are who they claim to be
- Integrity: verifies the data has not been forged or tampered with



For a website or application to use TLS, it must have a TLS certificate installed on its origin server (also known as an "SSL certificate")

A TLS certificate is issued by a certifying authority to the domain owner and contains important information about who owns the domain along with the server's public key, both of which are important for validating the server's identity

A TLS connection is initiated using a sequence known as the TLS handshake between the user's device and the web server

A note on Ports

Ports are used by TCP and UDP to separate data for different applications

One device and IP address can support multiple simultaneous UDP and TCP ports

In TCP

- Port numbers are assigned by sender and agreed by receiver
- Ports can be negotiated between sender & receiver, non-standard Ports can be used
- Useful for multiple sessions on same machine
- E.g. HTTP uses Port 80, POP3 uses Port 110

  NDI uses: 5960, 49152 to 65535

In UDP

- Port numbers are generated by the sender
- Typically Ports 16384 through 32767
- Firewalls typically block UDP

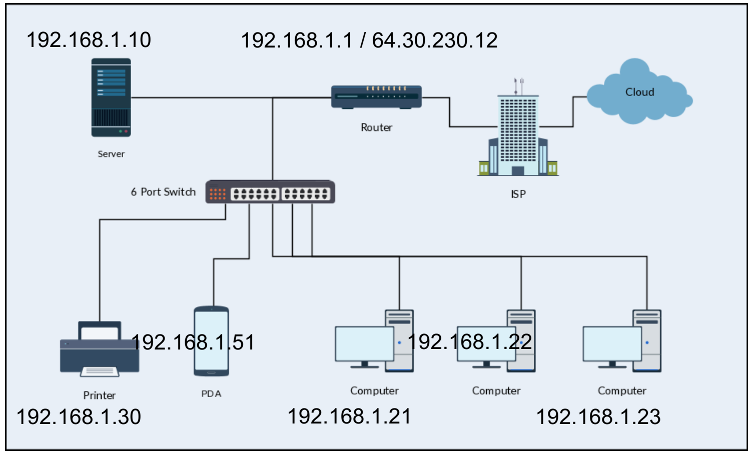
  =>exception rules need to be created

Example B’Pix ports	Port
Soft panel Control	80
API Control	9995
Remote Panel Control	9998
Panel Control	9999
Bpnet	21, 80, 443, 8000-8999, 3478, 30000-35000
Remote Commander	3478

A socket is a unique combination of IP address and Port number and is used to establish a secure link between devices

Networking basics

Ethernet is a family of computer networking technologies	
LAN; Local Area Network	WAN; Wide Area Network
Subnets (IPv4) in four octets (255.255.255.0) Private addresses: CLASS C : 192.168.0.0 (e.g – 192.168.1.100) CLASS B: 172.16.0.0 – 172.30.0.0 (e.g – 172.16.12.10) CLASS A: 10.0.0.0 – (e.g – 10.1.10.1) Most broadcasters will operate their LAN's in the Private range	IPv4 based now, IPv6 for the future Public addresses are assigned through ANA to ISP's, who in turn assign to businesses and users Internet is a public WAN and uses public addresses A Private cloud can use private addresses

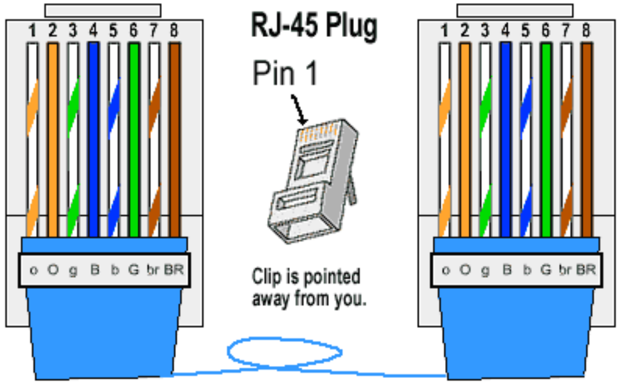


The router is the door between a public and private network:

- Some are one-way doors, others are two-way doors
- Shows different addresses internally and externally:
  - Usually X.X.X.1 in a private subnet
  - Can be nearly anything in a public subnet

TRANSMIT half
10Gb/s

RECEIVE half
10Gb/s

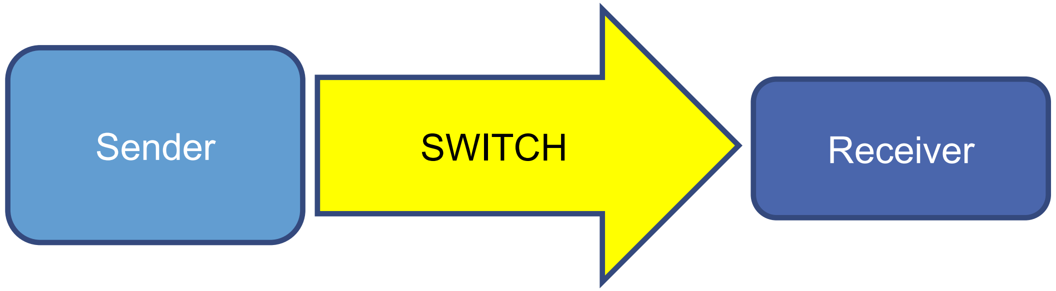


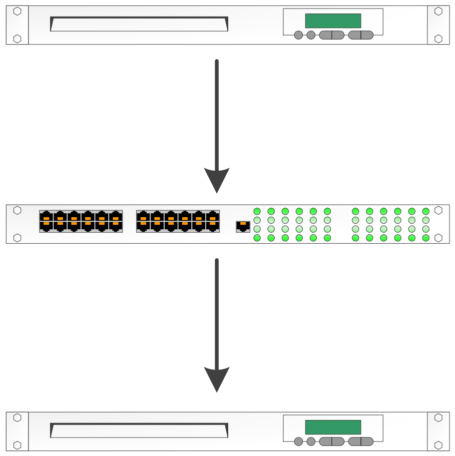
https://en.wikipedia.org/wiki/Ethernet

Unicast and Multicast

UNICAST: One-to-One

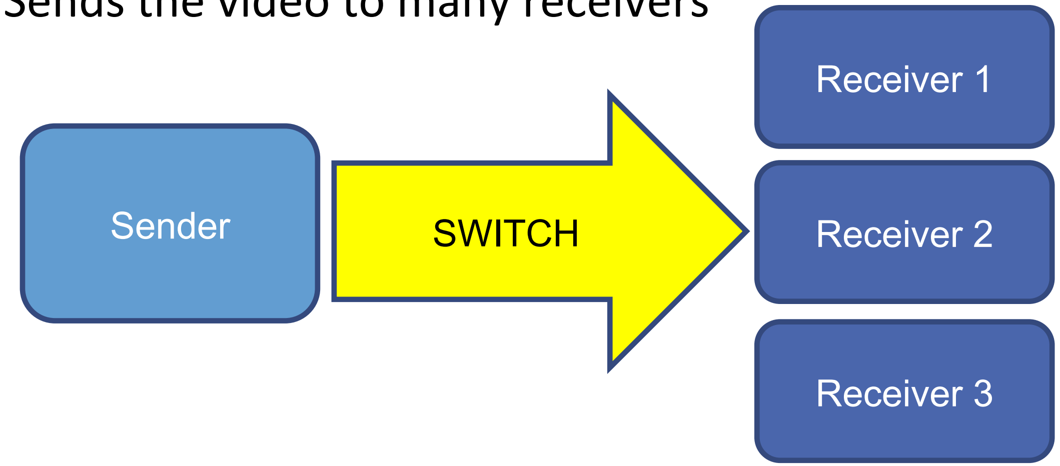
Sends the video ONLY to the receiver

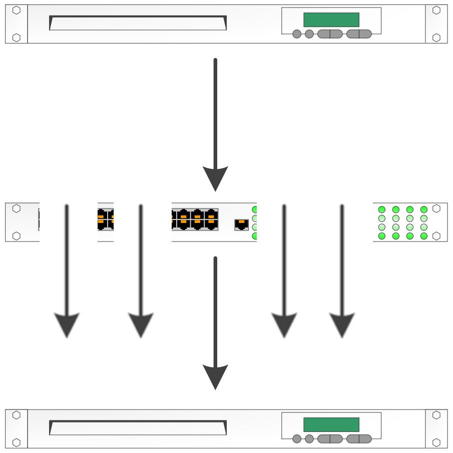




MULTICAST: One-to-Many

Sends the video to many receivers





Free IP training for Media Professionals

Buy the basic IP for Media Professionals course here:

https://learn-ip-video.thinkific.com/users/checkout/auth

…..and enter the following voucher code on checkout:

Bcastpixfree

Enjoy your course!

Other resource: Attend virtual VidTrans for for free (probably the only time ever) Monday March 1 and Tuesday March 2. Sign up here:

https://vsf.tv/upcoming_events/2021-03_VidTrans2021-Virtual/index.shtml

Broadcast Pix

Our abstracted architecture

Acquisition

Conferencing

zoom

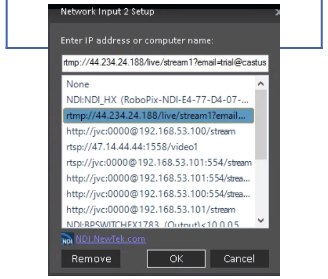
PowerPoint

Cameras

(NDI, RTSP)

(RTMP)

STREAM BY CASTUS



Network Input 2 Setup

Enter IP address or computer name:

rtmp://44.234.24.188/live/stream1?email=trial@castus

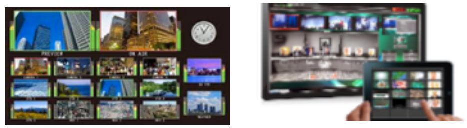
- None
- NDI:NDI_HX (RoboPix-NDI-E4-77-D4-07-...
- rtmp://44.234.24.188/live/stream1?email...
- http://jvc:0000@192.168.53.100/stream
- rtsp://47.14.44.44:1558/video1
- rtsp://jvc:0000@192.168.53.101:554/stream
- http://jvc:0000@192.168.53.101:554/strea...
- http://jvc:0000@192.168.53.100:554/strea...
- http://jvc:0000@192.168.53.101/stream
- NDI:BPSWITCHFX1783 (Output)<10.0.0.5

NDI.NewTek.com

(NDI, RTSP & RTMP)

Inputs enter through the network or Video inputs connect via encoders or the switcher core

Control (TCP, Web RTC)



Physical Panels

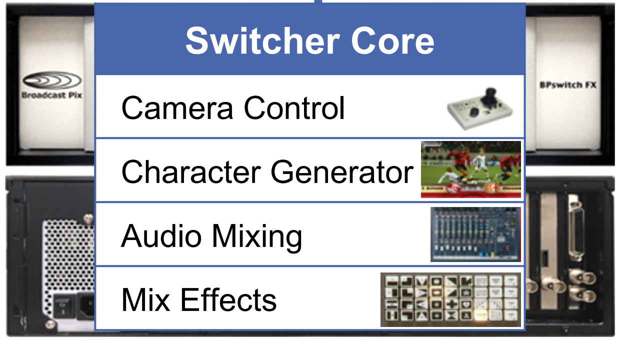
Software/Web Panels

All control is IP-based and can be over the public internet

Network

All key components can exist on a network, either remotely or local

Physical Video Inputs



Switcher Core

- Camera Control
- Character Generator
- Audio Mixing
- Mix Effects

(RTMP & FTP)

Output to cloud services for streaming, publishing or archive

Networking is at our heart!

Distribution

Publishing

YouTube

vimeo

Conferencing

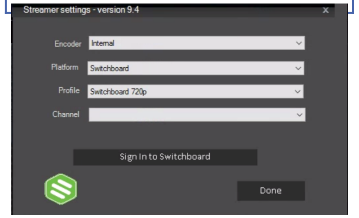
zoom

Live Streaming

brightcove

WOWZA media systems

Multi-streaming



Streamer settings - version 9.4

Encoder: Internal

Platform: Switchboard

Profile: Switchboard 720p

Channel:

VIDEO
AUDIO
Broadcast Pix
DATA
CONTROL