IP/MPLS Networks: Optimize Video Transport for Broadcasters

What You Will Learn

- Evolutionary trends of video transport over IP/MPLS networks
- Characteristics of packet video traffic in IP/MPLS networks and effects on user Quality of Experience
- Broadcast specific IP/MPLS video network designs
- Optimization techniques for transport of packet video in IP/MPLS networks

Introduction

IP is becoming the prevalent network technology for video transport, and, video is becoming an increasingly significant component of IP network traffic. The requirements for supporting multiple services, including audio and video, on converged IP and Multiprotocol Label Switching (IP/MPLS) networks have promoted developments in IP quality of service (QoS), resiliency, availability, and scalability.

As a result, IP/MPLS networks can now deliver the service quality that is needed by the premium video services that demand the highest quality. In a world that increasingly uses IP, IP/MPLS networks provide several benefits over previous technologies for the transport of premium video services. These benefits include flexibility to adapt to market demands, efficient use of bandwidth, reduced operating expense, ease of service, and application integration. IP is becoming the preferred converged network technology capable of meeting the requirements of both real-time and file-based video services, not only in secondary distribution networks such as Internet Protocol Television (IPTV) over residential broadband (for example, DSL and cable), but also increasingly for primary distribution and contribution networks. IP/MPLS transport networks are now the primary means for the delivery of next-generation broadcast services.

Cisco combines industry-leading experience in deploying carrier-class IP/MPLS networks with a deep understanding of video services gained from the acquisition of Scientific Atlanta. As a result, Cisco is exceptionally capable of meeting the requirements of broadcasters for transporting premium video services. The acceleration in video deployments increases the need to address the primary challenges facing broadcasters and service providers looking for IP-based video transport solutions, including how to deliver video service-level requirements and how to effectively manage and monitor the video transport service. Cisco has used its combined network and video application expertise to meet this challenge by developing network architectures and technologies for Video Optimized Network transport, incorporated into the industry's leading portfolio of video-enabled products and solutions.

This portfolio includes the Cisco® CRS-1 and CRS-3 Carrier Routing System products, Cisco ASR 9000 Series Aggregation Services Routers, Cisco 7600 Series Routers, and Cisco Digital Content Manager (DCM) Model D9900 & D9901 intelligent IP adapters. These products provide the following features to cost-effectively support the most stringent service-level agreements (SLAs) and enhanced video monitoring capabilities demanded for the highest-quality video transport services:

- Fabric-based multicast replication, rather than line-card based replication, giving outstanding multicast scale and performance for point-to-multipoint services
- Multipriority differentiated services schedulers, delivering the lowest possible delay and jitter in multiservice scenarios
- Point-to-multipoint MPLS traffic engineering for admission control and bandwidth reservation with fast reroute, providing 50 ms recovery times in case of link and line-card failures
- Multicast only fast reroute (MoFRR), providing rapid recovery in case of node failure
- IP over dense wavelength-division multiplexing (IPoDWDM), allowing reductions in capital and operating expenses, and the fastest possible detection of network failures, for even faster failure recovery
- Embedded video flow monitoring, providing pervasive router support for ubiquitous and cost-optimized detection and identification of network problems that reduce video quality

In combination with the router-embedded video flow monitoring capabilities, the Cisco Video Assurance Management Solution (VAMS) addresses video service management and monitoring, delivering an end-to-end service-aware view of the condition of the video transport network.

Also, the Cisco video adapter and encoder series of products, developed by the Cisco Service Provider Video Technology Group (SPVTG, previously Scientific Atlanta), allows the Cisco combined network and video solution to deliver lossless video transport for premium contribution and distribution services.

This suite of capabilities is the result of network-level and system-level development focused explicitly on the requirements of video transport and broadcasters, led by the Cisco Service Provider System Unit (SPSU), which has initiated a customer-inclusive systems-based development approach.

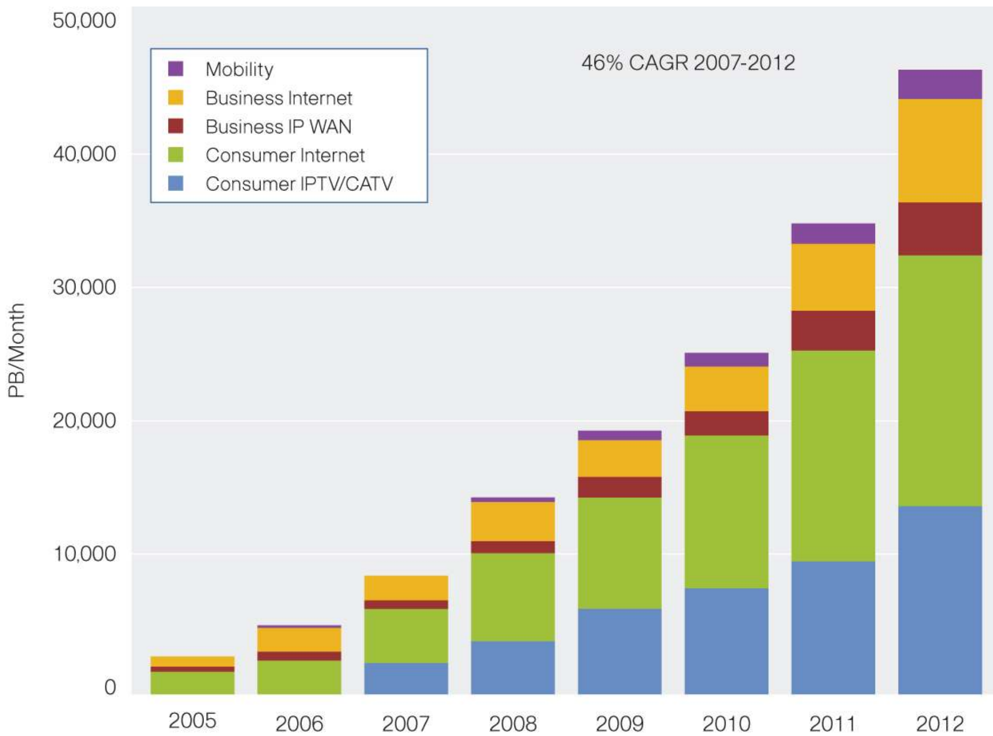
Developments in Video Transport

Creating broadcast video content has long been a process that followed a set sequence from acquisition to broadcast delivery. Traditional media such as tapes, and the sequential processing of content and traditional mechanisms for editing that content, have dictated the way in which broadcasters have delivered and have constrained their ability to use digitized media. Broadcasters worldwide employ a range of existing network platforms and technologies, including SDH, ATM, and DTM (Dynamic Synchronous Transfer Mode).

Several factors influence the evolution of broadcast networks. The capability to handle content as native digital files provides huge benefits in reducing time to air and allowing content to be processed multiple times by different editing, journalistic, and production teams. High-definition (HD) video leads broadcasters and production companies to invest in new platforms and technologies, while the emergence of file-based workflow imposes requirements to support both real-time and file-based transmission of productions. At the same time, market pressures promote reductions in operating expenses.

IP/MPLS is the prevalent network technology for video transport, and video is becoming an increasingly significant component of IP-based network traffic. The Cisco Visual Networking Index, which tracks IP traffic growth and the effect of visual network applications, predicts that by 2012 the Internet will be 75 times larger than it was in 2002, and that video will account for almost 90 percent of all consumer traffic on the Internet.

Figure 1. Cisco Forecasts 44 Exabytes per Month of IP Traffic in 2012



Major findings of the Cisco Visual Networking Index include the following.

- Global IP traffic will quadruple from 2009 to 2014. Overall, IP traffic will grow at a compound annual growth rate (CAGR) of 34 percent. Annual global IP traffic will exceed 3/4 of a zettabyte (767 exabytes) in four years. Global IP traffic grew 45 percent during 2009 to reach an annual run rate of 176 exabytes per year or 15 exabytes per month. In 2014, global IP traffic will reach 767 exabytes per year or 64 exabytes per month. The average monthly traffic in 2014 will be equivalent to 32 million people continuously streaming the movie Avatar in 3D for the entire month.
- Global Internet video traffic will surpass global peer-to-peer (P2P) traffic by the end of 2010. For the first time since 2000, P2P traffic will not be the largest Internet traffic type. The global online video community will surpass 1 billion users by the end of 2010. This number of people is exceeded only slightly by the populations of China (1.3 billion) and India (1.1 billion), making this user group equivalent to the third largest country in the world.

- P2P traffic is growing in volume, but declining as a percentage of overall IP traffic. P2P file-sharing networks now carry 3.5 exabytes per month and will continue to grow at a moderate pace with a CAGR of 16 percent from 2009 to 2014. Other means of file sharing, such as one-click file hosting, will grow rapidly at a CAGR of 47 percent and will reach 4 exabytes per month in 2014. Despite this growth, P2P as a percentage of consumer Internet traffic will drop to 17 percent by 2014, down from 39 percent at the end of 2009.
- Internet video is now over one-third of all consumer Internet traffic, and will approach 40 percent of consumer Internet traffic by the end of 2010, not including the video exchanged through P2P file sharing.
- The sum of all forms of video (TV, Video-on-Demand [VoD], Internet, and P2P) will continue to exceed 91 percent of global consumer traffic by 2014. Internet video alone will account for 57 percent of all consumer Internet traffic in 2014.
- Advanced Internet video (3D and high-definition video) will increase 23-fold between 2009 and 2014. By 2014, 3D and high-definition Internet video will comprise 46 percent of consumer Internet video traffic.
- The growth of video communications traffic is accelerating. Though still a small fraction of overall Internet traffic, video instant messaging and video calling are experiencing high growth. Video communications traffic will increase sevenfold from 2009 to 2014.
- Real-time video is growing in importance. By 2014, Internet TV will be over 8 percent of consumer Internet traffic, and ambient video will be an additional 5 percent of consumer Internet traffic. Live TV has gained substantial ground in the past few years. Globally, P2P TV traffic is now over 280 petabytes per month.
- VoD traffic will double every 2.5 years through 2014. Consumer IPTV and cable television (CATV) traffic will grow 33 percent between 2009 and 2014.

Meeting the requirements for supporting multiple services, including voice and video, on converged IP/MPLS networks, and improvements in quality of service (QoS), resiliency, availability, and scalability mean that IP networks can now deliver the service quality required for the highest-quality video. In a world that increasingly relies on IP, IP/MPLS networks provide the following advantages over previous technologies for the transport of premium video services:

- Flexibility to rapidly adapt to market demands and improve time to market with new services
- More efficient bandwidth utilization through packet-level statistical multiplexing
- Operating expense efficiencies through the capability to combine multiple services-for example, real-time video, file-based video, voice, and data-on a single converged infrastructure
- Improved service and application integration, with the capability to provide a unified infrastructure for all video services

IP/MPLS transport networks are becoming vital to the delivery of next-generation broadcast services. IP is becoming the preferred network convergence technology, capable of meeting the requirements of both real-time and file-based video services, not only in secondary distribution networks such as IPTV over residential broadband (for example, DSL and cable), but also increasingly for primary distribution and contribution networks.

This document describes the requirements, challenges, and relevant IP technologies that are involved in the delivery of video services for broadcasters, and describes how Cisco is developing these technologies to address market needs.

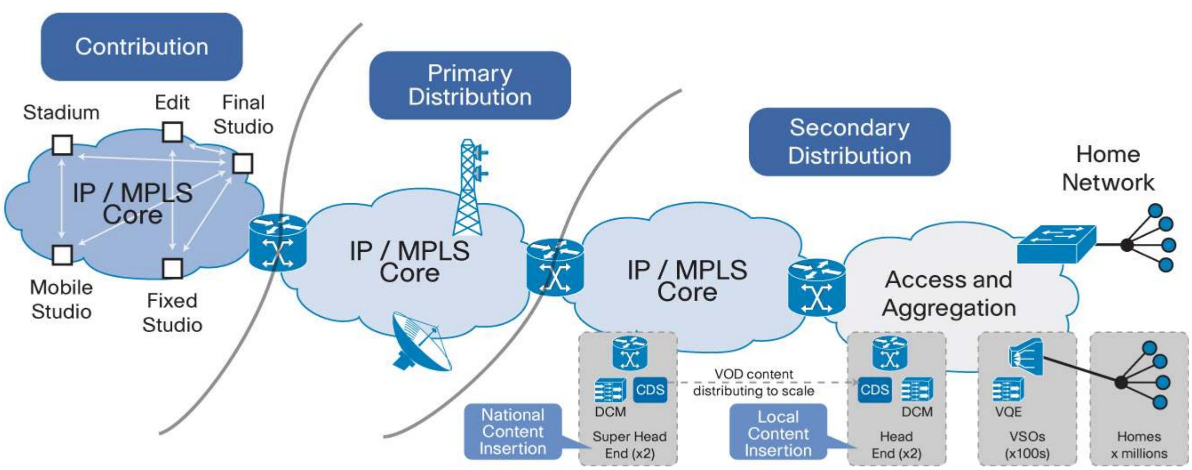
Challenges for Video Transport

The creation of video content and its distribution is a multistage process that follows a lifecycle from acquisition at the source, through production and packaging of the content, to distribution to viewers. IP video transport service providers can be categorized by the types of services that they provide: contribution, primary distribution, or secondary distribution. In some cases providers and networks may support more than one of the following services.

- Video contribution providers manage the capture and initial processing of video content and its transport prior to distribution. For example, the content may be transferred from one studio to another or from an event location such as a sporting event to a preproduction facility where the content is reworked before distribution. In general, the closer to the origin of the content, the stricter are the requirements for quality and bandwidth. Therefore video contribution services demand the very highest video quality, with bounded delay and minimal loss. The video streams used in video contribution may be uncompressed, or may use encoding such as JPEG 2000; higher compression ratios may be used where quality requirements are lower and bandwidth availability is more restrictive (for example, weather feeds or news gathering). Uncompressed video contribution services are typically very high speed, with SD stream rates of 270 Mbps and HD stream rates of 1.485 or 2.970 Gbps. The adaptation specifications used in these scenarios are defined by the Society of Motion Picture and Television Engineers (SMPTE), in standards such as SMPTE 292M, 372M, and 424M. Contribution services can be point-to-point (unicast) or point-to-multipoint (multicast).
- Primary video distribution providers manage the transport of video content from production environments to secondary distribution. Primary distribution services are normally compressed, ranging from MPEG-2, MPEG-4 to JPEG 2000 depending on the quality requirements. Networks supporting primary distribution services may share requirements from video contribution and secondary distribution providers.
- Secondary video distribution providers manage the transport of video content from primary video distribution providers to end consumers. IPTV, cable TV, and video over DOCSIS® are examples of secondary distribution services. Secondary distribution services are normally compressed using MPEG-2 or MPEG-4 standards, with rates of 2-4 Mbps for SD and 8 to-20 Mbps for HD. Secondary distribution services can be point-to-point (unicast-based VoD) or point-to-multipoint (multicast-based IPTV).

Figure 2 shows the relationships between the provider categories.

Figure 2. Video Service Provider Categories



Like other service provider networks, broadcast networks must combine simplicity, versatility, scalability, security and manageability with cost effectiveness. Additionally, all categories of video transport services share the following challenges:

- What transport technology to use: native IP or IP/MPLS?
- How to meet the required SLAs for premium video services?
- How to manage and monitor the service, to help ensure that the IP network delivers the required SLAs for video, and to identify and isolate problem areas?

Each category of video transport service places its own requirements on IP/MPLS network deployments.

An understanding of the detailed network requirements of each video service offering and an understanding of how IP-based technologies behave in the delivery of these services are the first steps in designing a network optimized for their transport. The rest of this document focuses on network requirements and technologies optimized for contribution and primary distribution services.

Video SLA Requirements

IP-based video applications have stringent SLA requirements for delay, jitter, and packet loss. Video applications do not degrade gracefully if these SLAs are not met. Rather, the utility of the video stream drops rapidly and the resulting user experience degrades to unacceptable levels. Among contribution and primary distribution video services, real-time streaming services have the most stringent SLA requirements. The essential SLA requirements for an IP-based video transport service can be defined in terms of delay, jitter, loss, and availability.

Network Delay

One-way network delay characterizes the time difference between the receipt of an IP packet at a defined network ingress point and its transmission at a defined network egress point. Network delays include four components: propagation delay along the network path (approximately 5 ms per 1000 km for optical fiber), switching delay and queuing delay at network elements on the path, and serialization delay (the time it takes to transmit the bits of the packet sequentially onto a link). In addition, application response might be subject to network control protocol processing delays (such as multicast processing) and delays due to processing in application end-systems.

The biggest effect of delay is on interactive services, for example, two-way audio. If the delay is too long, participants may mistake system delays for pauses in conversation and take these delays as their cues to begin to speak. By the time their words arrive at the other end, the other speaker may have already started to speak, with the result that the normal practice of conversation breaks down. International Telecommunications Union (ITU) recommendation G.114 suggests that about 150 ms of end-to-end one-way delay is sufficient to ensure that users will be very satisfied for most two-way audio applications. However, the best practice is to minimize end-to-end delays for any video applications with two-way interactive audio.

In the latest generation of IP/MPLS routers used in high-speed networks, switching delays are typically a few tens of microseconds and serialization delays are negligible. The Differentiated Services (DiffServ) IP QoS architecture (defined in RFC 2475) is used to control queuing delays and help ensure that service providers can meet their network delay SLAs. DiffServ is the baseline technology for IP QoS deployments today; it allows differentiated delay, jitter, and loss commitments to be supported on the same IP network for different types or classes of service. Traffic is classified at the edge of the network and then marked using the differentiated services code point (DSCP) in the IP packet header into a limited number of traffic aggregates or classes. Within the IP/MPLS network, scheduling and queuing control mechanisms are applied to the traffic classes based upon the DSCP marking; all traffic conditioning and dropping is handled intelligently at the network layer using IP DiffServ QoS mechanisms. DiffServ may be used in conjunction with the Integrated Services architecture (IntServ; defined in RFC 1633), which uses the Resource-Reservation Protocol (RSVP; defined in RFC 2205) for explicit admission control and bandwidth reservation.

In practice, with an effective DiffServ implementation and design, network delays will generally represent a relatively small proportion of the end-to-end delays. For example, in a national network within Europe, the end-to-end one-way network delays are typically less than 10 ms. Other contributors to the end-to-end delay include de-jitter buffers on the receiving decoder or IP adapter and encoding and decoding delays. Where minimal delay is required, uncompressed transport should be used to avoid encoding and decoding delays. Current JPEG 2000 solutions achieve 150 ms encoding and decoding delays, and next-generation JPEG 2000 solutions will achieve end-to-end delays of less than 100 ms.

Network Jitter

Network jitter is the variation in network delay caused by factors such as fluctuations in queuing and scheduling delays at network elements. We can generally consider jitter to be a variation of the one-way delay for two consecutive packets. De-jitter buffers are used to remove the delay variation the network causes by turning variable network delays into constant delays at the receiver. If a video de-jitter buffer is appropriately sized to accommodate the maximum value of network jitter possible, jitter won't delay playout beyond the worst-case end-to-end network delay. DiffServ IP QoS mechanisms are used to control network delays, and thus to set the maximum network jitter.

Packet Loss

Packet loss characterizes the packet drops that occur between a defined network ingress point and a defined network egress point. We consider a packet lost if it does not arrive at the specified egress point within a defined time period. Network packet loss has three primary causes.

- Congestion: When congestion occurs, queues build up and the network drops packets. DiffServ IP QoS mechanisms and capacity planning are used to help ensure that congestion does not occur, so no packets are dropped due to congestion. These mechanisms may be augmented with RSVP-based admission control.
- Lower-layer errors: Bit errors, which might occur due to noise or attenuation in the transmission medium, can result in dropped packets. Actual bit error rates vary depending on the underlying Layer 1 or Layer 2 technologies used, which are different for different parts of the network. In practice, fiber-based optical links might support bit-error rates as low as 1 bit in error out of 1013 transmitted bits (1e-13), whereas asymmetric DSL (ADSL) services might have bit-error rates as high as 1e-3. Some link-layer technologies employ reliability mechanisms, such as Forward Error Correction (FEC), to recover from commonly occurring bit-error cases and thus reduce the effective packet loss rate.
- Network element failures: Most networks are built to be resilient; however network element failures, such as link or node failures, can result in losses of network connectivity, which cause packets to be dropped until the network connectivity is restored around the failed network element. The resulting packet loss period depends on the capabilities of the underlying network technologies and implementations; these are described in the next section.

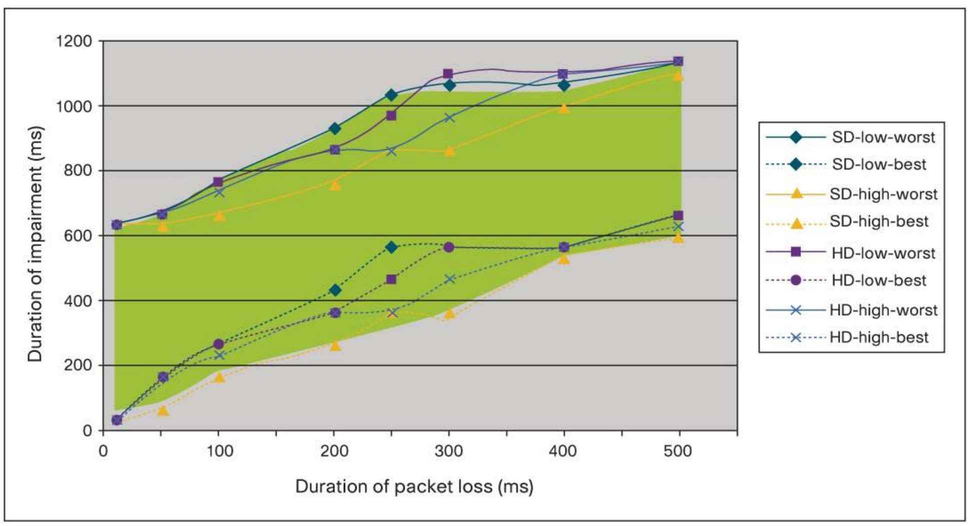
In summary, the DiffServ architecture is used in IP/MPLS networks to help ensure that the required delay, jitter, and loss SLAs can be achieved; this may be combined with RSVP-based admission control. With appropriate network engineering and capacity planning, the only network events that should result in a visual impairment to the video service are packet losses due to lower-layer errors or network element failures, which should be limited.

Nonetheless, any loss may result in a degradation of the transported video signal. Although limited loss may be acceptable to an end viewer, maintaining quality is essential in the early stages of the production and distribution chain. With compressed video, the effect of even limited packet loss can be significant and any single unrecovered video packet loss may result in a visual impairment. Depending upon factors such as the specific encoding and compression scheme, losses of different packet types appear as different types and durations of visual impairment or artefact. With MPEG encoding, a single packet loss causes a slice error at the minimum (Figure 3); multiple packet loss can result in blocking and pixelization or ghosting, or freezing and even the complete loss of the video image.

Figure 3. Artefacts Caused by Packet Loss: Slice errors, Blocking and Pixelization, and Ghosting (Source material copyright SMPTE, used with permission)

The graph in Figure 4 plots the results of testing that demonstrates the relationships between network outage duration (contiguous packet loss) and the resulting visual impairment duration. This testing was undertaken for MPEG-2 encoded video traffic with high- and low-motion video clips, with both SD and HD. The bounded area in the graph shows the range of the resulting visual impairment, from worst case to best case for each test case and scenario.

Figure 4. MPEG-2: Relation of the Duration of Iimpairment to the Duration of Packet Loss in Standard-Definition (SD) and High-Definition (HD) Videos with High and Low Motion in Best- and Worst-Case Scenarios



The results show that even 50 ms of packet loss can result in a significant visual impairment, which may last up to about 700 ms. With MPEG-4 encoding, impairments resulting from an equivalent duration of packet loss will generally be greater than for MPEG-2. The relative effect of loss will generally be less with JPEG 2000 because it uses intra-frame based compression only, so errors in one frame cannot propagate into subsequent frames. For more details on this study, see the following documents:

- Jason Greengrass, John Evans, Ali C. Begen, "Not All Packets Are Equal, Part I: Streaming Video Coding and SLA Requirements," IEEE Internet Computing, vol. 13, no. 1, January 2009, pp. 70-75 http://www.cisco.com/en/US/solutions/collateral/ns341/ns524/ns610/jg-je-ab-ieee-int-comp-jan09.pdf
- Jason Greengrass, John Evans, Ali C. Begen, "Not All Packets Are Equal, Part II: The Impact of Packet Loss on Streaming Video," IEEE Internet Computing, vol. 13, no. 2, March 2009, pp. 74-84 http://www.cisco.com/en/US/solutions/collateral/ns341/ns524/ns610/jg-je-ab-ieee-int-comp-mar09.pdf

Thus when the highest quality is required, even limited loss may be unacceptable. In these cases, loss recovery may be required, as discussed in the next section.

Availability

Network availability is the fraction of time that network connectivity is available between a network ingress point and a network egress point. For video, however, simply having connectivity is not enough, so service availability is often a more meaningful metric. Service availability is a compound metric, defined as the fraction of time the service is available between a specified ingress point and a specified egress point within the bounds of the other defined SLA metrics for the service, for example, delay, jitter, and loss.

The highest levels of service availability in IP/MPLS networks are achieved by combining the following approaches:

- Selecting carrier-class network elements with resilient processors and power supplies, resulting in high mean time between failures (MTBF) and low mean time to repair (MTTR) and thus high element availability
- Designing the network to be resilient with no single points of failure, employing redundancy in both network elements and links; this requires identifying any shared risks (for example, a situation in which a failure of a duct may affect two links) and constructing the network so that the failure of those shared risks will not unacceptably affect service
- Using IP and MPLS fast-convergence and fast-reroute technologies, augmented by rapid failure-detection capabilities, such as IPoDWDM technology, to rapidly switch to alternate paths in the presence of network element failure, minimizing packet loss resulting from the failure
- Employing advanced high-availability techniques to minimize the impairment and packet loss that may result from route processor upgrades or failures
- Using DiffServ QoS, admission control, and capacity planning to meet the SLA requirements for delay, jitter, and loss, if necessary in cases of network element (link and node) failure
- Using transport and application-level approaches to recover from any loss experienced, thus providing lossless transport
- Using a service management solution that is closely coupled to the video transport network, to rapidly isolate and identify faults that affect service when they occur

The use of these techniques and technologies, which are described in more detail in the next section, provide network and service availability of greater than 99.999 percent with IP/MPLS networks.

Video-Enabling Transport Technologies

These combined SLA requirements place exacting requirements on the network. Network infrastructures that were designed for best-effort delivery, with relatively low bandwidths and high latencies will not be able to meet the requirements of next-generation video services. Networks optimized for premium video services, however, can support other applications. As a result, the latest network infrastructures are heavily influenced by video applications.

In addition to the defined SLA requirements IP-based video contribution services have a unique set of characteristics compared to other IP-based applications. These services use very high-bandwidth flows, which are inelastic (do not change their rate of sending depending upon the available capacity), are generally unidirectional, and may require both point-to-point and point-to-multipoint connectivity.

Medianet Technologies

To support these combined requirements and characteristics, Cisco is developing a suite of networking technologies that span the range of video-optimized products and define the medianet - an intelligent network optimized for rich media. For more details on Cisco medianet, see http://www.cisco.com/web/solutions/medianet/sp.html.

The following medianet technologies can be applied to enhance the SLAs offered for transported video services, individually or in combination, allowing broadcasters and service providers to choose the optimal set of capabilities to meet their video service requirements.

Fabric-Based Multicast Replication

Efficient multicast replication is fundamental to supporting point-to-multipoint video services such as video contribution. The Cisco video-optimized platforms use innovative service-intelligent switching fabrics built for massive multicast replication and forwarding, which provide superior in-fabric replication compared to products that use inefficient line-card-based replication.

Multi-Priority DiffServ Schedulers

The Cisco DiffServ IP/MPLS QoS implementation is mature and very widely deployed in the industry. Cisco has led the industry in developing multiple priority-scheduling implementations that have been optimized through focused development and years of experience supporting premium services, including video. These implementations provide end-to-end jitter of less than 1 ms in high-speed multiservice IP/MPLS networks as described in the following document:

- Clarence Filsfils and John Evans, "Deploying Diffserv in IP/MPLS Backbone Networks for Tight SLA Control," IEEE Internet Computing, vol. 9, no. 1, January 2005, pp. 58-65. http://www.cisco.com/en/US/prod/collateral/routers/ps167/prod_white_paper0900aecd802232cd.pdf

The following study demonstrates that these SLAs are achievable in high-speed multiservice IP/MPLS networks today; worst-case jitter was measured at less than 700 microseconds for probes sent at 1 Mbps during a 7-day period between the U.S. East- and West-coast POPs of a Tier-1 IP network service provider.

- Steve Casner, Cengiz Alaettinoglu, Chia-Chee Kuan, “A Fine-Grained View of High-Performance Networking,” Packet Design, NANOG 22, May 20-22, 2001

Fast IP-Routing Protocol Convergence

In IP networks, convergence is the process by which routing protocols find alternative paths around failed network elements (links or nodes). Cisco leads the industry in the development of implementation and protocol optimizations such as IP prefix prioritization, which can reliably deliver significantly subsecond network convergence times for unicast routing protocols such as Open Shortest Path First (OSPF), Intermediate System to Intermediate System (IS-IS), and Border Gateway Protocol (BGP), and for multicast routing protocols such as Protocol Independent Multicast (PIM). For details, see the following document:

- Pierre Francois, Clarence Filsfils, John Evans and Olivier Bonaventure, "Achieving subsecond IGP convergence in large IP networks," ACM SIGCOMM Computer Communication Review, Vol. 35, Issue 3 (July 2005), pp. 35-44 http://www.cisco.com/en/US/solutions/collateral/ns341/ns524/ns610/cf-je-ccr-igp-convergence.pdf

As a specific example, for 400 multicast groups, convergence times of about 200 ms are realistically achievable today with optimized implementations. Fast Convergence represents the baseline for all video transport deployments and is available today through Cisco IOS® Software and Cisco IOS-XR Software across the range of video-optimized network products offered by Cisco.

Fast Convergence may be enhanced with other technologies where faster recovery from failures is required, for premium video transport services.

MoFRR

MoFRR is a simple enhancement to PIM Sparse Mode (PIM-SM) processing (defined in RFC 4601), which can provide the capability to instantiate multiple spatially redundant branches of the same multicast tree between a source and receiver. MoFRR can be used to further reduce the multicast convergence times to a few hundreds of milliseconds in link and node failure cases, without significantly increasing network complexity. More information about MoFRR is available in the following documents:

- A. Karan, C. Filsfils, D. Farinacci, “Multicast only Fast Re-Route,” IETF draft: http://www.ietf.org/internet-drafts/draft-karan-mofrr-00.txt
- Dino Farinacci, Clarence Filsfils, “A Simple and Efficient 0(50msec) Resilience Technology for IPTV,” NANOG 42, February 2008 http://www.nanog.org/meetings/nanog42/presentations/DinoFilsfils_iptv.pdf

MPLS Traffic Engineering with Fast Reroute

MPLS Traffic Engineering (MPLS TE, defined in RFC 3209) uses the resource RSVP (defined in RFC 2205) to signal MPLS TE tunnels, which can be used to explicitly define video connection paths through the network and provide admission control and bandwidth reservation. MPLS TE can also be used in conjunction with DiffServ for DiffServ-aware (class aware) traffic engineering. The Cisco MPLS TE implementation is the most widely deployed in the industry. Cisco has developed point-to-multipoint (P2M) MPLS traffic engineering (defined in RFC 4875) across its range of video-enabled routers, in support of premium multicast-based video transport applications, such as video contribution environments.

Where MPLS TE is deployed, Fast Reroute (FRR, defined in RFC 4090) can be used in addition to Fast Convergence. FRR allows precalculated backup TE tunnels to be used to protect against link, interface, and line-card failures such that traffic can be rerouted around the failed element, restoring connectivity within 50 ms of the failure.

IPoDWDM

Cisco introduced IPoDWDM to the industry in December 2005. Since then, providers have used the power of IPoDWDM to combine and distribute video content rapidly and efficiently over all-IP networks. The Cisco IPoDWDM solution allows the reduction of transport elements, reducing CapEx and OpEx. Also, through integration of the transponder, the router line card participates in G.709 signaling and gains information on the link bit error rate (BER). As a result, in some failure cases the router can detect that links are degrading before they have actually failed (before they have started to drop packets), and can proactively switch to an alternative path. This proactive protection capability can be used in conjunction with Fast Convergence and FRR, with the potential to significantly reduce the losses resulting from network failure. The Cisco IPoDWDM solution is available across the Cisco core, edge, and aggregation platforms.

High-Availability Technologies

Cisco high-availability technologies include Nonstop Forwarding (NSF), Stateful Switchover (SSO), and In Service Software Upgrade (ISSU), which minimize the impact and packet loss that may result from upgrades or failures of router control-plane processors.

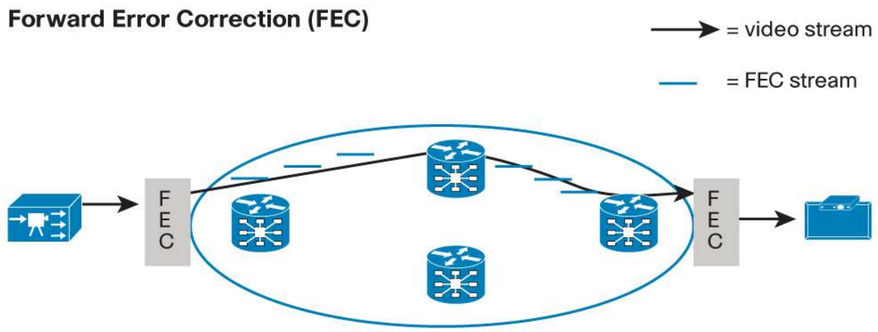
Transport and Application Approaches to Loss Recovery

The preceding network-level technologies may be further augmented by transport- and application-level approaches to recover from any loss experienced, so as to provide lossless transport for premium video services.

FEC

FEC (Figure 5) adds redundancy to the transmitted data to allow the receiver to detect and correct errors (within some limits) without the need to resend any data. FEC is most useful for recovering from intermittent packet loss due to lower-layer bit errors.

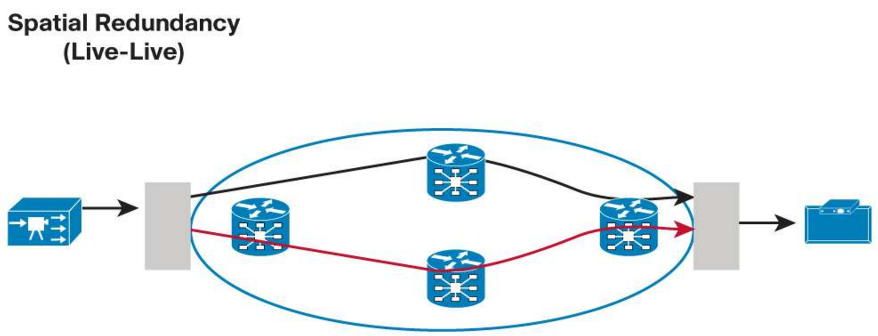
Figure 5. Forward Error Correction (FEC)



Spatial Redundancy

With spatial redundancy (Figure 6), also known as. live-live, two identical streams are sent over diverse paths between the sender and receiver. If a network failure affects one stream, the other will still be received and the playout will be will be uninterrupted. Spatial redundancy is most useful for recovering from burst losses due to network failure events, such as link or node failures. Several IP/MPLS network technologies can be used to help ensure that the streams follow spatially diverse paths, including MoFRR, MPLS TE, and Multi-Topology Routing (MTR).

Figure 6. Spatial Redundancy (Live-Live)



These loss-recovery capabilities are available in the Cisco DCM products. In addition, the video adapter and encoder/decoder products developed by the Cisco SPVTG allow the Cisco combined network and video solution to deliver lossless video transport, satisfying the most stringent video SLA requirements for premium contribution and distribution services.

Service Management and Monitoring

The ability to verify that the IP network is delivering the required SLAs for video, and to identify problem areas, is essential to successful IP video deployments. This verification includes both element management and service management. Service management includes service monitoring at the device, system, and application levels. Cisco has developed the following capabilities to provide enhanced video transport service management and monitoring.

Embedded Network Video Monitoring

Embedded network video transport monitoring can detect issues with individual video flows at identified routers. This capability is being developed across the Cisco video-optimized routers to provide metrics including: Media Delivery Index (MDI), Media Rate Variation (MRV), Media Loss Rate (MLR) and Media Stop Events (MSE).

These embedded video monitoring capabilities provide pervasive router support and deployment for comprehensive and cost-optimized video-quality problem detection and isolation, with increasing specificity of troubleshooting down to a particular device or link.

Trap and Clone

As a complement to embedded video monitoring, trap and clone provides the ability to tap into a video stream within the network, copy the packets from that stream and transport them to a location where a video quality analyzer can assess the quality of the video stream. This capability makes efficient use of video quality analyzers for detailed video stream quality monitoring, with reduced CapEx.

VAMS

The Cisco VAMS is an innovative solution for video services, which provides an end-to-end view of the condition of the video transport network. VAMS combines information about the network topology and status with an assessment of the quality of the transported video services, through embedded or standalone video quality monitoring, to facilitate the rapid isolation, identification, and impact analysis of faults that may affect the quality of transported video services.

Video-Optimized Network Platforms

Cisco has embedded video intelligence in its core and edge portfolio of router products, offering a wide selection of products that provide medianet-enabled, video-optimized transport and that address the full range of network requirements.

Figure 7. Cisco Network Platforms

The Cisco portfolio of router products supports the strictest SLAs and enhanced video monitoring capabilities demanded for the highest quality video transport services.:

Cisco CRS

The Cisco CRS is a carrier-routing system that offers continuous system operation, superior service flexibility, and system longevity. Powered by Cisco IOS XR Software, the Cisco CRS can scale system capacity up to 92 Tbps (CRS-1) and 322 Tbps (CRS-3).

The Cisco CRS-1 and Cisco CRS-3 carrier routing systems can power OC-768c/STM-256c IP and DWDM interfaces while supporting up to 1152 40-Gbps line cards of Packet-over-SONET (PoS), wavelength-division multiplexing (WDM), and Ethernet interfaces. The Cisco CRS routers help simplify today’s networks while protecting investments for decades to come.

The Cisco CRS-1 is built on Cisco IOS XR Software, a self-healing operating system for multishelf, multiterabit carrier infrastructures. This microkernel-based operating system provides highly specific process independence, fault containment, and isolation. With these capabilities, the Cisco CRS-1 can be maintained, upgraded, enhanced, and scaled without requiring service interruptions.

The Cisco CRS combines:

- Cisco QuantumFlow Array, a chipset architecture engineered in multiple dimensions of scale, services, and savings
- Cisco Silicon Packet Processor (SPP), a highly sophisticated 40-Gbps application-specific integrated circuit (ASIC)
- Cisco IOS XR Software, with Cisco Service Separation Architecture (SSA) and Cisco Service-Intelligent Switch Fabric to provide maximum service flexibility and capability

With comprehensive service separation and complete line-rate feature flexibility, the Cisco CRS can deliver the capabilities needed by the most advanced converged network services today and tomorrow. For more information about the Cisco CRS, see http://www.cisco.com/en/US/products/ps5763.

Cisco ASR-9000 Series Routers

The Cisco ASR 9000 Series offers up to 400 Gb of capacity per slot, for up to 6.4 terabits of total capacity. With these routers, the edge gains a robust and service-tested operating environment that meets the reliability and performance requirements for a nonstop video experience.

In 10-slot and 6-slot versions, the Cisco ASR 9000 Series routers are power-efficient, space-efficient, and reliable enough to maintain nonstop video and content-rich service delivery from the network edge. Modular power supplies are brought online only as capacity is increased; the Cisco ASR 9000 Series consumes only the power needed to support the configuration demands of the system and side-to-back airflow makes the 6-slot Cisco ASR 9000 Series capable of further reducing heat dissipation and power consumption in space-constrained facilities

The Cisco Advanced Video Services Module (AVSM) is a major innovation, providing terabytes of streaming capacity at the aggregation edge while simultaneously offering content caching and ad insertion. The Cisco AVSM also features integration of Cisco VQE capabilities of fast channel change, and error correction, all on a single blade that slides into the Cisco ASR 9000 Series routers. For more information about the Cisco ASR-9000 Series, see http://www.cisco.com/go/asr9000.

Cisco 7600 Series Routers

Cisco 7600 Series Routers offer integrated, high-density Ethernet switching, carrier-class IP/MPLS routing, and 10-Gbps interfaces. Video and multicast admission control capabilities in Cisco 7600 Series Routers provide superior end-to-end admission control for broadcast TV and VoD.

The Cisco 7600 Series supports high-performance IP/MPLS features as well as scalable personalized IP services at the network edge, improving operational efficiency and maximizing return on network investment. The Cisco 7600 Series is the industry's first carrier-class edge router to offer integrated, high-density Ethernet switching, carrier-class IP/MPLS routing, and 10-Gbps interfaces, helping service providers to deliver both consumer and business services over a single converged carrier network.

Cisco 7600 Series Routers have advanced availability, scalability, security, flexibility and other features that broadcasters and service providers demand for video transport services. For more information about the Cisco 7600 Series, see http://www.cisco.com/go/7600

Cisco DCM Gateway

The Cisco DCM Gateway is a very compact serial digital interface (SDI) gateway that delivers outstanding IP adapter capabilities for video contribution applications. With 1 Gigabit Ethernet and 10 Gigabit Ethernet interfaces and superior IP connectivity and redundancy features, the DCM Gateway can support up to 12 video streams from a single rack unit (1RU) form factor. With a flexible and scalable architecture that will be compatible with future software and hardware upgrades, the DCM Gateway provides the following capabilities:

- A basic gateway card that facilitates lossless video transport, with support for both Code of Practice (CoP4) FEC and Hitless Merge with spatial redundancy
- Video quality improvements and adaptation to the evolution of transport protocols through software downloads
- Optional JPEG 2000 compression module
- 1RU with up to two gateway blades; up to 6 ports standard-definition or high-definition 720p/1080i per blade; up to 3 3G-HD 1080p per blade
- Transport of video, embedded audio, and VBI
- Advanced adaptive clock recovery and genlock

For more information about the Cisco DCM, see http://www.cisco.com/en/US/prod/collateral/video/ps9159/ps9195/ps9828/7018133.pdf

Service Partner

Cisco has teamed with Dimetis GmbH, a full-service broadcast operations support system (BOSS) supplier, to provide its video-aware BOSS© Broadcast Platform for the monitoring, control, and scheduling of IP/MPLS-based contribution and distribution links. The Dimetis BOSS Broadcast Platform is based on service-oriented architecture (SOA), and incorporates a modular and flexible design to accommodate new-generation services. Dimetis, headquartered in Dietzenbach, Germany, is a leading software and hardware systems integrator, providing standards-based and customized BOSS solutions. Its BOSS Broadcast Platform is deployed in various configurations at some of the world's largest carriers and TV broadcasters including: AboveNet, ARD, Deutsche Telekom AG, HanseNet, Level 3 and Broadwing, ORF+ORS, SRG SSR, Telekom Austria, Westdeutscher Rundfunk (WDR), and Zweites Deutsches Fernsehen (ZDF). For more information about Dimetris products, visit http://www.dimetis.de.

Conclusion

The move to IP/MPLS-based networks supporting video transport increases the need to address the main challenges facing broadcasters looking for next-generation video transport solutions. The expertise in carrier-class IP/MPLS networks and video technology gained through the acquisition of Scientific Atlanta makes Cisco exceptionally capable of addressing these challenges. In combination with the Cisco customer-inclusive systems-based development approach, the result is an outstanding suite of video-optimized products and solutions to address broadcaster requirements. These products and solutions take full advantage of embedded video intelligence capable of supporting the strictest SLAs, and provide the enhanced video monitoring capabilities demanded for premium video transport. These capabilities allow converged infrastructures to be shared by different services with reduced expenditures, accelerating the deployment of premium video services and the optimal delivery of the highest-quality video.

Visit the following websites for examples of how Cisco products have been used to meet the needs of broadcasters and video service providers around the world:

- Lossless Video Architecture: http://www.cisco.com/en/US/netsol/ns577/networking_solutions_solution_category.html
- Mahangar Telephone Nigam LTD.: http://newsroom.cisco.com/dlls/2010/prod_120710b.html
- Abertis Telecom: http://www.cisco.com/en/US/solutions/collateral/ns341/ns898/abertis_video_case_study.html
- Genesis Networks, Inc: http://www.cisco.com/en/US/solutions/ns341/ns525/ns537/ns705/C36-494046-00_CS_Genesis_v4a.pdf
- National Broadcasting Company (NBC): http://newsroom.cisco.com/dlls/2008/prod_080608b.html
- Radiotelevisione Italiana (RAI): http://newsroom.cisco.com/dlls/2007/prod_091007b.html

For more information

For more information about the Cisco video-optimized network architectures, technologies, and products, contact your local Cisco account team and ask for SPSU Video Optimized Transport Systems. For more information about Cisco products and services, visit http://www.cisco.com

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