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Digital Video Synchronization and IP Networking

John C. Bellamy, Donald H. Buell, Michael M. Salour
IPITEK, Inc.

Motivations for "Everything over IP"

Lower Cost Transmission

The highly innovative and competitive data communications market place provides low cost solutions for basic transmission and switching equipment. If these basic capabilities are adequate for an application, lowest possible transmission and switching costs are realized.

Simplified Management

Around the time that the Internet, as we know it, was emerging from a closed government and academia development environment, major telecommunications carriers defined a new management protocol called Telecommunications Management Network (TMN). TMN was an ambitious undertaking to allow network operations, administration, management and provisioning of all equipment types (network elements) from diverse manufacturers. The Internet Engineering Task Force (IETF) looked at TMN and decided that the proposed architecture was too bloated to meet their immediate needs. Thus, the origination of a "Simplified Network Management Protocol" or SNMP came about. A key feature of the IP architecture and SNMP is the extensive use of "auto-discovery" provisioning. There are virtually no restrictions on the network topology. A new topology arises by connecting another node to the network. The network discovers the new node and routes traffic to the new node according to the IP addresses that it declares are within its domain.

One Network

From a service providers point-of-view, the major advantage of adding as many service as possible to an IP network is the ability to maintain and manage a single network. In addition to providing basic telephone services, the telecommunications service providers have developed many separately managed networks for such services as: PSTN, ISDN, frame relay, DDS, X.25, and ATM to mention just a few. Supporting these myriad of services is expensive and inevitably leads to limitations in service areas. The labeled multiplexing nature of an IP network allows services of vastly different bandwidth requirements to co-exist in the same network. The challenge, however, is to establish and maintain Qualities of Service (QoS) for real time applications. Inevitably, QoS implementations involve establishing virtual circuits within a modified IP architecture.

End User Control

In contrast to the typical circuit-switched telecommunications network, much of the intelligence of an IP network is at the edge (i.e. in the network gateways or the user terminals themselves). The basic paradigm of the IP architecture is to minimize intelligence within the network so innovative, unforeseen applications can arise, at the edges, from innovative developers. The most prevalent example of the benefit of such an approach is the development of the World Wide Web. The “www” command invokes browsing and document processing software within the end user’s terminal (PC) that enables switching connections to hyperlinks within a document retrieved from a distant web page. As shown in Figure 1, The World Wide Web is an application that rides on top of the IP protocol. Whenever a user clicks on a hyperlink within a webpage the browser application software extracts embedded IP address information which is passed to the IP layer to set up a connection to another distant web page.

End user control is, of course, a two-edged sword. It is very difficult to constrain malicious users and cheats that have access to the addressing and control functions of the network.

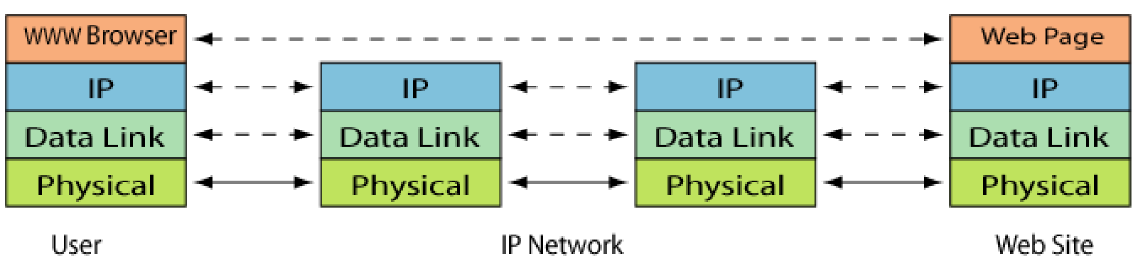

Figure 1 WWW Application Layer on top of the IP Layer

Synchronization Considerations

A central requirement of choosing IP for transport of real time traffic (e.g. voice or video) involves managing delay and delay variation (jitter) in the network. A particularly important aspect of delivering acceptable video quality is reconstruction of a clock at a decoder that is synchronized to the encoder clock.

SDI Synchronization

Figure 2 illustrates the basic function of the Program Clock Reference (PCR) information embedded in a SMPTE-259 [1] digital video stream. The fundamental purpose of the PCRs [2,3] is to provide synchronization between the encoder and decoder. Because the sample clock in the encoder is locked to the video line rate and chroma subcarrier of the input signal it is critical that video reconstruction in the decoder use a synchronized clock.

For the situation with a dedicated digital transmission link as shown in Figure 2, the decoder clock is easily recovered by a phase-locked-loop wherein the differences between the local counter values and the received PCR values serve as phase measurements. Any detected differences are used to adjust the frequency of the local oscillator to drive the average differences to zero. It is assumed that the delay from the PCR source to the receiver is constant so that differences in the local counter value and the received PCR value are only attributable to a phase error in the local clock which is minimized by the PLL.

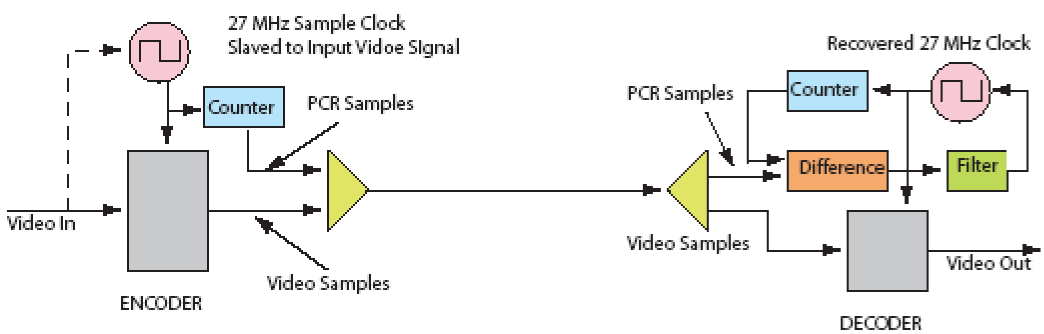

Figure 2 SDI Encoder/Decoder Synchronization

Genlock Synchronization

When the destination location has a local video reference it is often more desirable to synchronize the incoming video to the local clock using a generator lock "genlock" circuit. A means of accomplishing this in a digital environment is shown in Figure 3. Frames of digital video data are stored in a video frame buffer from which the decoder extracts an entire frame at a time. In the case of 270 Mbps SDI, the size of a single video frame is 9 megabits.

If the source clock and the local clock are not synchronized slips are inevitable which means that every once in a while a video frame is discarded (if the source is fast) of repeated (if the source is slow). The frequency of slips is determined by the magnitude of difference in the two clocks. For the case of the two clocks differing by 20 ppm (rather loose clocks) the time between slips is 28 minutes.

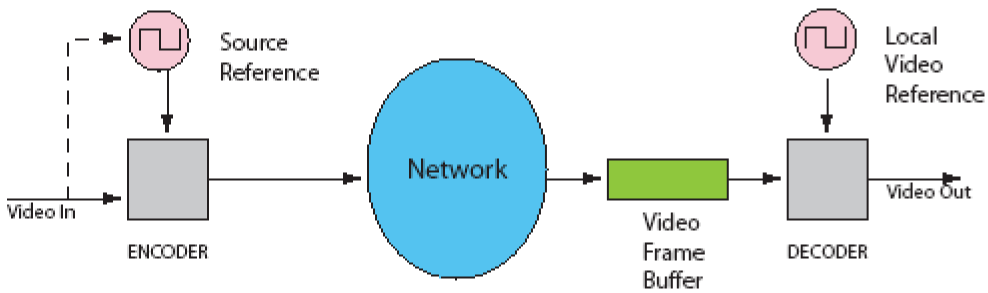

Figure 3 Slip Buffer Video Synchronization

Video over IP Synchronization

Extending the use of SDI (or other video encoding methods) across a network as shown in Figure 4 introduces a significant complication into the synchronization process. Because the network will, in general, not deliver the PCR values with constant delay, the difference measurements in the decoder contain delay variations in addition to the effects of clock offsets. The delay variations introduce phase noise that must be removed by the phase-locked-loop recovery circuit.

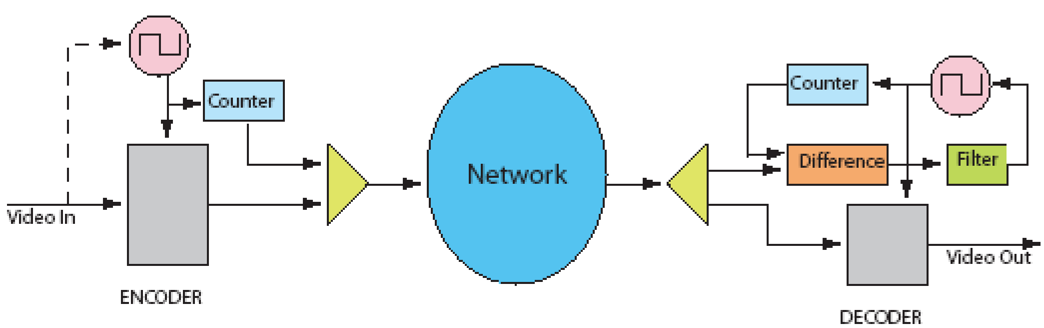

Figure 4 Network Transport of SDI

In general, delay variations can not be adequately removed by the PLL in the decoder. This is particularly true when the video is being carried across an IP network wherein encoding and routing operations produce large delay variations (packet jitter). The use of a Quality of Service (QoS) provision in the network helps alleviate the problem but can not solve it by itself.

Figures 5 and 6 depict two methods of accommodating network induced jitter. In the first example, a data buffer is used to smooth the delivery of the egress data. The fill level of the buffer is used as phase measurement to control the output rate so the buffer never overflows or underflows. If the buffer is large enough the reaction time of the output clock can be slow enough that only low frequency jitter passes to the output link. The main disadvantage of the approach shown in Figure 5 is implied need of a large delay in the channel and the inability of the circuit to completely remove all jitter.

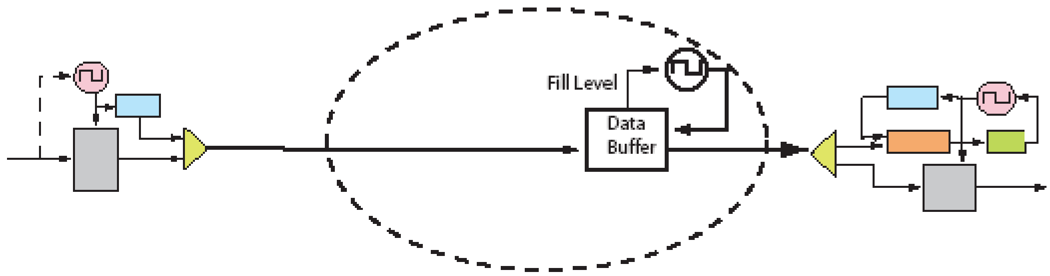

Figure 5 Use of Data Smoothing Circuit to Reduce Network Induced Jitter

In contrast to the above method, the implementation shown in Figure 6 explicitly recreates the data rate at the ingress of the network at the egress point. As indicated, the insertion of time stamps into the data stream allows the output circuitry to determine the desired output rate by distributing the received data across a time interval equal to that indicated by the time stamps. For this system to work it is imperative that the ingress and egress points have access to a common clock or, in the very least, access to two separate but highly accurate clocks.

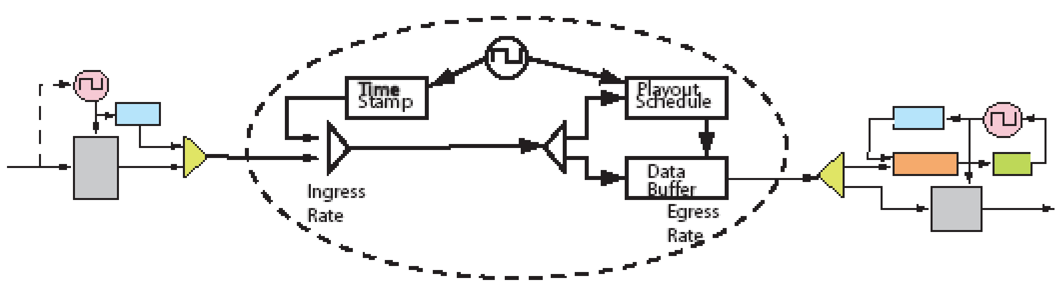

Figure 6 Use of Time Stamps to Replicate an Ingress Rate at the Egress of a Network

Layer-3 versus Layer-2 Switching

The data communications industry utilizes two networking approaches that can be broadly categorized as either layer-3 switching or layer-2 switching. Layer-3 switching, which is commonly called routing, involves network nodes processing entire IP addresses and individually determining (from local internal tables) which output link to forward a packet onto. If a node receives a packet with an unknown IP address it communicates with other nodes to learn the location of the address and adds the appropriate entry into its routing table.) Notice that a router has knowledge of the network as a whole in the sense that it processes the final destination IP address to determine the appropriate output link.

In a layer-2 switching environment it is first necessary to set up a connection through the network for an impending communication. Associated with each link of the end-to-end connection is a connection ID that a receiving node uses to select an output link for forwarding packets. A connection ID (which may be changed from one link to another) has no global significance. A layer-2 switching node uses a connection ID as an index into a table defining a map from local inputs to outputs. In general, layer-2 switching provides better control of delay and delay variations than does layer-3 switching. QoS considerations require a connection oriented network in all cases. (A layer-3 network can provide connection oriented services in addition to conventional routing services.)

Multi-Protocol Label Switching

Multi-Protocol Label Switching (MPLS) is one approach to augmenting the IP protocol to incorporate layer-2 services with the specific intent of providing QoS. As the name implies, MPLS is designed to carry traffic of multiple services—not just IP traffic. All traffic in an MPLS network is carried in pre-established connections. The individual data flows are inserted into MPLS packets which contain layer-2 connection ID's. Figure 7 depicts an MPLS network carrying traffic between two IP network domains. The ingress gateway of the MPLS network uses an incoming IP address to determine the appropriate connection ID when encapsulating the IP packet into an MPLS packet. Intermediate nodes in the MPLS network ignore the IP address. (The IP address is merely part of the data within the MPLS envelope.) At the egress gateway MPLS wrappers are removed and packets are routed at layer 3.

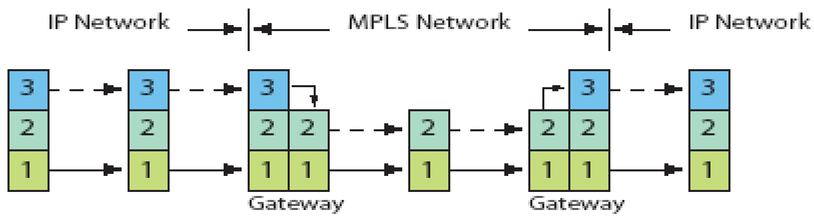

Figure 7 IP Communications Through an MPLS Network

Ethernet Networking

Another application of layer-2 networking is illustrated in Figure 8 wherein an Ethernet local area network is extended through high speed Ethernet links and switches to be a regional network with a gateway to an IP network. Traffic in the Ethernet domain is switched according to Ethernet MAC addresses. Again, the IP address is embedded in the Ethernet frames and the intermediate Ethernet nodes ignore them.

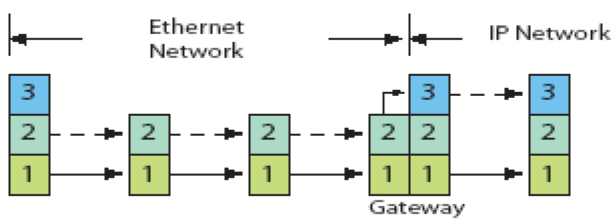

Figure 8 Ethernet Network Interfaced to an IP Network

The significant feature of either of the network situations in Figures 7 and 8 is that layer-2 switching provides lower end-to-end delays and delay variations. Because QoS considerations for real time traffic inherently requires end-to-end connection set up, doing so in the layer-2 domain is a natural solution. Ethernet technology has expanded beyond the local area network applications to become a viable networking solution with high speed (e.g. 10 Gigabit) links and high speed switches that can be arranged in rings such as Resilient Protection Rings (RPRs)[4].

References:

[1] SMPTE 259M-1997, 10-Bit 4:2:2 Component and 4fSC Composite Digital Signals – Serial Digital Interface, Society of Motion Picture and Television Engineers, 1997.

[2]] ISO/IEC 13818-1, Information technology – Generic coding of moving pictures and associated audio information: Systems, ISO/IEC, Geneva, 2000

[3] ISO/IEC 13818-9, Information technology – Generic coding of moving pictures and associated audio information: Extension for real-time interface for systems decoders, ISO/IEC, Geneva, 1996.

[4] IEEE 802.17-2004 -- Telecommunications and information exchange between systems Local and metropolitan area networks Specific requirements Part 17: Resilient packet ring (RPR) access method and physical layer specifications.

Acronyms

CMTS	Cable Modem Termination System
HD	High Definition video
HD-SDI	High Definition Serial Digital Interface (SMPTE 292M)
HFC	Hybrid Fiber Coax
IETF	Internet Engineering Task Force

MAC	Media Access Control
MPTS	Multiple Program Transport Stream (DVB-ASI)
PTS	Presentation Time Stamps
PCR	Program Clock Reference
PES	Packetized Elementary Stream (MPEG data from a single audio or video source)
PS	Program Stream (combined video and data stream)
RPR	Resilient Packet Ring (IEEE 802.17)
QoS	Quality of Service
RSVP	ReSerVation Protocol (RFC 2205)
SD	Standard Definition video
SDI	Serial Digital Interface (uncompressed, ANSI/SMPTE 259M,)
SDTI	Serial Digital Transport Interface. Extension of SDI to support compressed digital video signals (SMPTE 305M)
SPTS	Single Program Transport Stream (DVB-ASI)