

# **Coding/Compression Requirements from the Network Protocol's Viewpoint**

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## **Abstract**

Several major efforts are under way in the Internet community to bring real-time communications into reality. Most of the efforts are targeted to solve network-related issues and leave the coding/compression problems to be answered by industry market value becomes obvious. However, at this point, most of the codec vendors are focusing on the constant-bit-rate network, which is the current telephone network infrastructure. In this paper, we will demonstrate the importance of developing variable-bit-rate compression algorithms and standards. This effort is not only for user Internet community, but also for users in the next generation of the telecommunication network (broadband ISDN), which uses ATM (Asynchronous Transfer Mode) technology. Due to a strong dependency between coding/compression and networking for variable-bit-rate networks, we will attempt to specify the requirements for the compression algorithm designs and experiments from the networking point of view.

## 1. Introduction

The current Internet, based on packet-switched technology, possesses an asynchronous nature (also called variable-bit rate services). It is very desirable to design compression algorithms to take advantage of this characteristic. However, none of the coding schemes currently available (e.g., H.261 [LIOU91; FOX91], MPEG [GALL91; FOX91], JPEG [WALL91; FOX91]) are designed for this purpose.

In fact, all the compression schemes, in their natural form, generate asynchronous data. For synchronous networks, these data are buffered and sent out in constant rate. It is actually easier to design the compression algorithm for an asynchronous network than for a synchronous one. One example is the data generated by the ADPCM (Adaptive Differential Pulse Code Modulation) scheme. ADPCM is used for voice and video coding/compression, which transmits only the temporal difference between the current voice signal and the previous one. The signal difference varies according to the conversation, and results in a variable data rate.

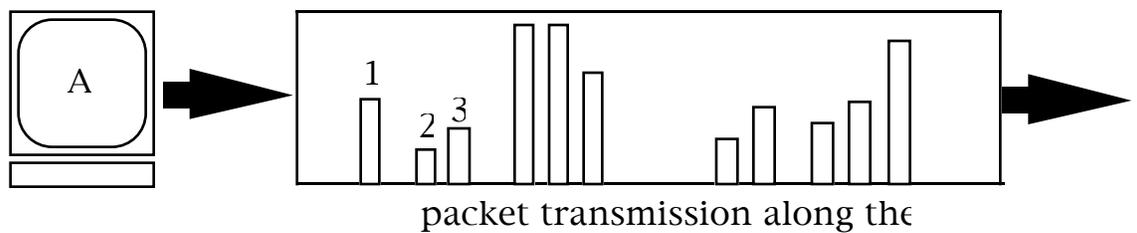
In the next few years, other areas will also drive coding/compression development for asynchronous networks. One area is LAN-based networks, which are in the category of asynchronous networks as well. According to most analysts, between 29 and 40 million computers will be on LANs in the U.S. in 1995 [FORR91]. Another area is the future B-ISDN (Broadband Integrated Service Digital Networks), which will be based on ATM (Asynchronous Transfer Mode) and also will be asynchronous in nature. For this reason, we believe that variable-bit-rate compression is the trend for video compression in the future.

In the Section 2 of this paper, we will show the basic characteristics of a variable-bit-rate network and demonstrate the dependencies between the compression algorithm and network traffic. In Section 3, we will list the requirements needed to design a compression algorithm for asynchronous networks.

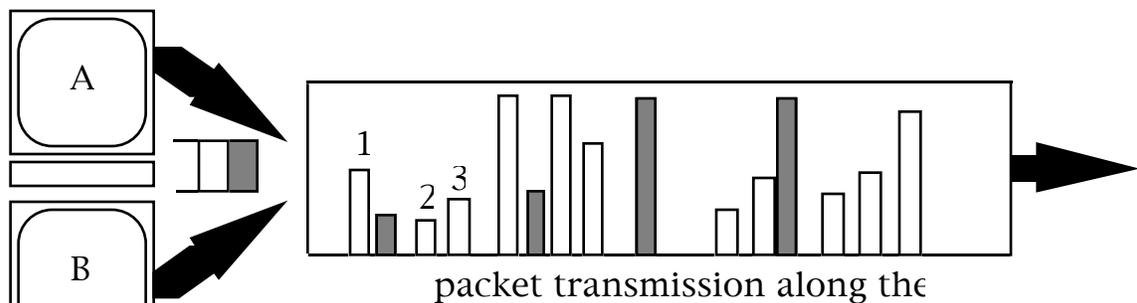
## 2. The Major Characteristics of the Variable-Bit-Rate Networks

The major characteristic of an asynchronous network is a traffic pattern with stochastic (as opposed to deterministic) interarrival characteristics (Figure 1a). If there is only one

sending the data through the network, the traffic pattern is the same as the traf there are more sources using the same link, their traffic will be multiplexed toge according to the time any individual message arrives. If there are multiple mess the same time, some of them will be buffered and delayed (Figure 1). When the so heavy that the buffer can not handle all the data, some messages will be dro further delayed through a congestion control mechanism. The delay can be dif each packet. For example, message 1 in Figure 1 is delayed due to multiplexing traffic, but messages 2 and 3 do not have any delay. This variation of the delay messages is called jitter. Jitter is not desirable for applications when it varies to general, the asynchronous network environment provides better network utiliz; introduces the potential for message delay and lost and out-of-sequence messag the basic idea of statistical multiplexing and resource sharing in packet-switched



(a) Traffic that has only on



Buffering if the resource is not

(b) Traffic combined by two

Figure 1. Delay and Congestion Caused by Asynchronous Networks.

For this kind of environment, an optimal scheme should be designed to work with a network having statistical traffic characteristics. This is different in synchronous environments, in which a constant-bit-rate channel is allocated for every application. In a synchronous network, once this type of channel is set up, the application does not interact with the network. It only has to regulate its flow. In the case of asynchronous networks, the compression scheme has to be designed to interact with asynchronous networks to deal with the extra delay generated by buffering as well as with media due to congestion.

To support real-time applications, we believe most future networks will have resource management schemes to support timing requirements. These management schemes provide information from the application to specify its traffic pattern, which normally includes average rate, peak rate, burst period, and jitter. There is a stronger relationship between an application and the network for an asynchronous network compared with a synchronous network. To illustrate this point, consider the case of compression of compressed video schemes. For a synchronous network, the compression scheme needs only one constant-bit rate. For an asynchronous network, the compression scheme needs four parameters:

### **3. Relationship Between Coding/Compression and Networking**

The following is a list of items that show the relationship between coding/compression and networking.

*Timestamping information should be provided by the compression/coding function [CASN91].*

The compressed data for every frame has to be marked with a timestamp. The purpose is for timing recovery of messages at the receiver side, considering the delay in the network and buffering at the receiver side. Furthermore, the receiver can synchronize media from different sources without the timing information. This also can be used by the networking layers to know the time limit for transmitting and forwarding the message across networks, which is a way to reduce the jitter.

For using timestamping, an accurate global clock for each host is required. This is done in the Internet community with the network time protocol [MILL89], which

synchronize the network clock to millisecond accuracy. In the future ATM environment we expect the network will provide this service.

*Packetizing information should be given by the compression/coding function.*

*Indication for the same packet:* The compressed data for each unit of the frame must be marked to show that they belong to the same frame. For example, the subband divides the image into several frequency bands. These bands then are packetized into several packets. There should be an indication to show that these packets belong to the same frame.

*Match compression unit (block size) with the network segment size:* Another example is that some compression schemes divide the image into blocks such as DCT (Discrete Cosine Transform), then send the blocks over networks. Because the network will segment data according to its characteristics (e.g., one ATM cell has 48 bytes of data), the block size should be matched closely with the network segment size.

*Indication for the location of the same image:* The information that shows the location of blocks in the same image must be given to the network protocol layers. In this case, there will be higher tolerance of out-of-order messages for blocks in the same image, but it doesn't make any difference for the receiving end which blocks are within the same image.

*Good error tolerance is needed in both compression/coding and network levels*

There are two different kinds of network errors. One kind is bit errors, which exist in the same way for both synchronous and asynchronous network environments. The other error pattern exists only in the asynchronous network environment. Due to the nature of the environment, the asynchronous network is more likely than the synchronous network to have packets lost, out of order, and delayed. These phenomena are undesirable because once the compression ratio becomes higher, the system is less tolerant of errors. There are several places where this problem can be dealt with. One place is the compression/coding layer. Some adequate redundant information can be added to allow error recovery. However, we don't want to add so much information that we nullify the benefits from compression. The other place is the network layer. The application should specify its reliability requirements to the network. This reliability index should be able to change from a fully reliable state to an unreliable state. For example, the ATM uses the *lost probability*

to show the degree of reliability. So, a strong relationship between compression control and network layer error control needs to be defined. For example, the application layer should be able to scale the error control functionality up or down to fit with network layer error control (Figure 2).

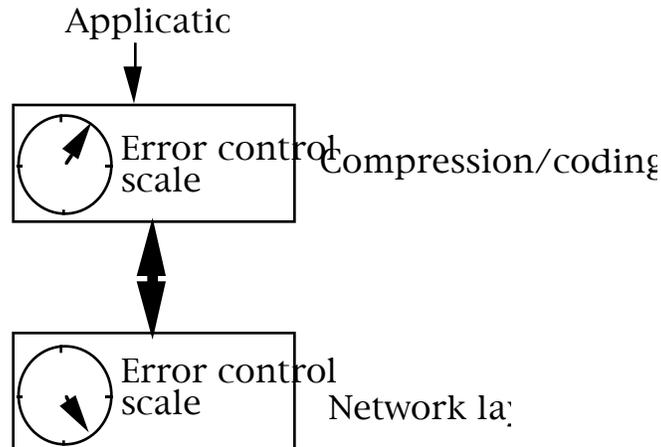


Figure 2. The Relationship Between the Error Control Adjustment in both Layers

*Scalable performance is needed.*

The image or video/audio quality should be scalable once the network bandwidth is available (say from 19.2 Kb/s to 300 Mb/s). Because the future ATM network provides bandwidth on demand, the compression schemes should be able to scale smoothly. For example, the standard of H.261 (also called Px64) is scalable to multiples of 64. On the other hand, the current MPEG 1 standard specifies the bit rate around 1.5 Mb/s. MPEG 2 has the bit rate around 10 Mb/s. Although there is a range that can be adjusted, MPEG 1 and 2, neither of which is flexible enough to scale in the whole range.

*Interoperability among various performance levels is required.*

If a host has a connection to a high-bandwidth network and is talking to two other hosts at the same time who are transmitting at different rates because of their network conditions, this host should be able to decompress the low- and high-bandwidth traffic without any problem. We call this "interoperability" among different bandwidth levels. If, for example, the receiver who can send at 128-Kb/s rate should be able to receive a 19.2-Kb/s signal without any problem.

*Adaptive coding schemes are needed.*

One relationship with the network is adaptivity related to network traffic. If the network is temporarily congested, this information should be able to feed back to the code to adjust the traffic (and quality) and then come back to the original quality once the congestion is over. Because the current codec already has a feedback mechanism to adjust its compression from the output buffer occupancy information, we need to extend this feedback mechanism from internal buffer to networks. The adjustment of the code also needs an implicit or explicit handshaking between the sender and the receiver to change compression parameters. It is worthwhile to note that the network congestion control function must take the network round trip time delay and the duration of the congestion into account. If the congestion is transient relative to the round trip delay, an end-to-end adjustment will not help. It will help when the congestion is longer than the network round-trip delay. When the congestion lasts about the same time as the round-trip delay, there is a possibility that the control will cause oscillation.

*Ways of coping with network resource management parameters are required.*

Future networks will require the application to specify its traffic parameters. For example, there have been suggestions to use average rate, peak rate, burst period, and jitter as traffic parameters. These parameters are used by the network to figure out the requirements for network resources. The compression algorithm should take advantage of the knowledge of the traffic parameters to design a lowest cost coding scheme.

*Compression should specify the priority to be put on packet header.*

The network should support priority levels to distinguish different real-time requirements of the applications. The compression/coding scheme should take advantage of this and specify the priority of the compressed data to the network layers. In this way, the network can map the priority requirement into packet headers. For example, the case of progressive image coding and transmission [DREI87; HUAN90] is first to send the most important features of the image followed by the less important ones. This scheme can use the priority level supported by the network to make sure the important parts arrive on time and the less important parts can have lower network priority. The layer coding scheme or hierarchical coding scheme will also benefit from the network priority.

*Ways of dealing with multipoint (multicast) requirements are needed.*

The current solution for multipoint conferencing does not depend on the network multicasting function. Its configuration is similar to the one in Figure 3. A central control unit handles all point-to-point connections to each participant. The audio and video messages from each source will go through the process of audio mixing and video switching and are then redirected to all the participants.

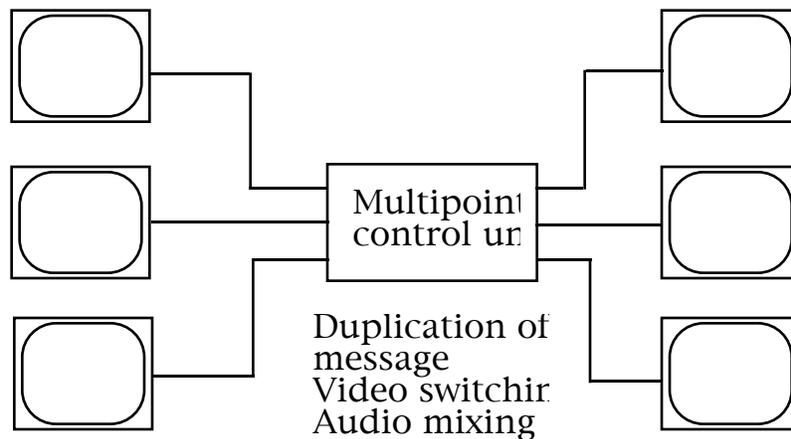


Figure 3. Current Multipoint Control.

For networks that have multicast capability (i.e., the network can deliver the message from one source to multiple receivers), we need to add only video switching and audio mixing capability. The current Internet has been implementing multicast capability [DEER92] and several ATM switches are incorporating this feature as well [FORE92; TURN88]. Where should the audio mixing and video switching functionality be in the Internet? In ATM network environments? Do we still need a centralized system? The answer is no, because a centralized solution is not flexible enough to support different numbers of participants. We feel that these functions should be distributed to each end (Figure 4).

One solution similar to this one is from IBM [KAND92]. Although there are multiple signals coming from every participant to a receiver, these signals are combined into a single frame for update. In this case, only one codec is necessary at every receiver. This technique currently applies only on motion JPEG, which should be extended to other coding/compression algorithms.

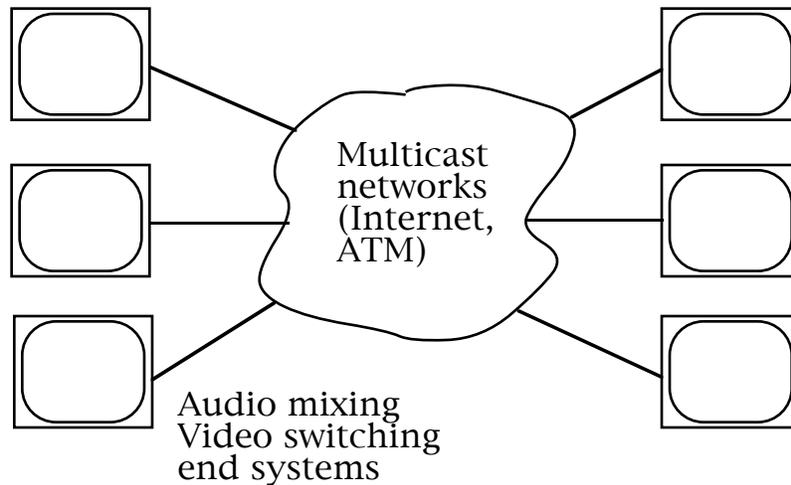


Figure 4. The Future Multipoint Control, which will Use the Network Multicast

*Software implementation for experiments is needed.*

Software implementation will not be required for future production use but is quite important at the early stages for experiments. We need a software codec implementation that can be used to test the integration of coding/compression function and the based on the current workstation technologies. A possible next step solution of software implementation is to build video-based DSP (digital signal processing) processors.

*Other Issues not Directly Related to Networking.*

*Support multiple coding format [WROC91].* The codec has to support several different coding formats.

*Real-time echo cancellation.* Real-time echo cancellation is important when the network delay is large, which is normally the case in wide-area communications. We need real-time echo cancellation to get rid of the echo. Ideally, this function should be in the codec.

*Audio Mixing and Video Switching and Integration with the Window System.* If the future multimedia station is on a personal computer or workstation, some functions have to be incorporated into the window system (e.g., X window). For a conferencing session,

multiple participants, there should be a way to show all the participants' image: screen, and indicate the person who is currently speaking. For example, we can this person's talking-head window. In other words, we can add this function to for video switching and audio mixing.

#### **4. Conclusion**

In this paper, we have shown the motivation to push the development of compression/coding algorithms and standards for the asynchronous network. demonstrate basic characteristics of asynchronous networks, which require significant interaction with coding/compression schemes. The major contribution is a list of requirements and possible opportunities for the coding/compression design for this purpose. We believe there is a need to remove the barriers between coding/compression researchers and networking researchers. In order to accelerate the development of multimedia communications, a mutual understanding of each others' requirements is important. This is especially true for the asynchronous network environment. This is an effort to specify the coding/compression requirements from the network point of view.

#### **5. Acknowledgements**

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