INTEROPERABILITY ON THE INFORMATION SUPERHIGHWAY: THE CONTINUING SAGA

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I. INTRODUCTION

Most of the participants who plan to build or use the "Information Superhighway" agree that **INTEROPERABILITY** is one of the most critical success factors. However, as in most cases in which agreement on broad general concepts is universal, we find that "the devil is in the details". It is hard to argue with the concept that Interoperability is required to have multiple equipment and system providers and to have access by the consumer to multiple programming and other service providers. However, the key questions are how to define Interoperability in the complex network structure of the "Information Superhighway" and how to implement Interoperability in the intensely competitive and highly political market.

History tells us that the only way to resolve Interoperability issues such as we find on the "Information Superhighway" is for all the players to understand accurately the technical and economic implications in sufficient detail so that rational business decisions can be made. In the past year the Communications Industry has made significant progress in understanding the technical details in many segments of the "Information Superhighway". Although progress has been made in understanding the economic details, uncertainty still exists both in the area of cost of installation and operation and in the area of derived revenue from the variety of potential programs and services. The large scale trials that are being implemented in 1994 by Time Warner, US West and others will address some of the unanswered technical and economic questions.

From a technical perspective it is useful to consider the Interoperability issues in the following categories:

1. Digital video compression
2. Digital audio compression
3. Digital system multiplexing and transport
4. Modulation and error correction
5. Security and conditional access
6. Network operating system

Of course, there are multiple sub-layers in each one of the above categories that must be addressed in detail to achieve Interoperability. This paper attempts to address important sub-layers in each of the major categories in which agreement has been reached on details and to identify several critical Interoperability details on which the industry must still reach agreement to have a truly Interoperable "Information Superhighway".

II. DIGITAL VIDEO COMPRESSION

The International Standards Organization (ISO) Moving Pictures Experts Group (MPEG) has led the way in defining standards for digital video compression that will allow Interoperability in this important part of the "Information Superhighway". It seems to be almost universally accepted now that the MPEG-2 digital video compression syntax will become the Interoperability standard for digital transmission both in the United States and in the rest of the world.

The MPEG-2 standard has been designed with inputs from the leading digital video compression experts around the world. It makes available a "tool kit" of digital video compression techniques that are the most powerful that can be practically implemented in real time video compression encoding today. MPEG-2 custom Integrated Circuit (IC) decoder chips exist today that can handle all of the techniques in the Main Profile Main Level of the MPEG-2 standard. The video encoding strategy can be selected from the MPEG "tool kit" according to specific applications and programming needs with the decoders capable of handling whatever choices are made in encoding.

There has been some confusion and perhaps misunderstandings on how MPEG-2 maintains Interoperability while allowing options for the user on the prediction modes for motion compensation. This dilemma is usually centered on the advantages and disadvantages of using what are called B-Frames in predicting vectors for motion compensation.

For reference MPEG-2 allows a choice in the prediction mode for motion compensation in the encoding strategy. The major choices are the following:

1. I-Frame only: In this mode there is no prediction of current frames either from previous or future frames. The compression process is based entirely on techniques that operate within each individual frame. I-Frame only modes are used when frame cut editing and multiple generation recording are required in studio and production applications.
2. P-Frame prediction: In this mode a current frame is predicted from motion compensation vectors that are derived from a previous frame only. Field based prediction is possible on a macro-block basis in this mode. However, two motion vectors, one for each field, are required.
3. P-Frame dual prime: In this mode prediction is done on the odd and even fields in a frame separately using information from the fields in the previous frame. However, only one motion vector with a small differential motion vector is transmitted for each block in the pair of fields and an algorithm is used to adjust the motion vector for the difference in line structure and time of occurrence for the odd and even fields.
4. B-Frames prediction: In this mode both a previous frame and a future frame are used to predict the current frame. Tests in the MPEG process have shown that this predictive mode is the most accurate of all those considered. However, it requires more memory in the decoder for an additional frame store than the other modes.

A video compression decoder designed to handle the MPEG-2 main profile can recognize and decode any of the predictive modes summarized above. However, these modes can not be used simultaneously on the same frame. MPEG testing has shown that for a given data rate the picture quality ranking and memory requirements are as shown in Figure 1.

For movies B-Frame prediction has a dramatic advantage over other prediction modes. Movies are completely framed based source material and field based prediction modes are not applicable. For programming services that are primarily movies B-Frame prediction can maintain good picture quality at significantly lower data rates than all other prediction modes.

Although there may be some differences in subjective evaluations of the degree of difference in picture quality among the four modes, experts in video compression rank them in the order given in Figure 1. Also, it is universally accepted that to implement B-Frame prediction for broadcast and studio quality resolution the incremental frame store will require an additional 8 Mbits of DRAM in the decoder. For lower resolutions less additional memory is required. No additional memory is required for horizontal resolution of 352 lines or less.

The difficult forecast is to determine the cost of additional DRAM in a decoder in future years. Today the volume unit for DRAM memory is 4 Mbits, therefore, two of the 4 Mbit units are required for P-Frame prediction either standard or dual prime. Four 4 Mbit units are required for B-Frame full resolution prediction. Even in very high volume production an 8 Mbit difference in memory can add $16 to the material costs of the decoder today.

We know that the cost of memory will come down in the future as the Semiconductor Industry goes to smaller geometry and larger memory unit sizes. The unit size for memory and the cost is driven by the Computer Industry today. Predictions are that the Computer Industry will skip the 8 Mbit unit and go directly to a 16 Mbit unit as the next memory unit size for volume production. However, the exact date that 16 Mbit DRAM will be available in volume with the necessary bus structure and low cost is very difficult to predict exactly. 16 Mbit DRAM unit volume production is estimated by some to peak in 1997 with a significant reduction in cost per Mbit. Bus structure and memory access time are critical parameters for DRAM used for video frame stores.

Today, typically four 4 Mbit parts each with a 16 bit bus are used to achieve the 16 Mbit requirement. They are typically accessed simultaneously (i.e. 64 bit bus) to achieve the required bandwidth. In order to use a single 16 Mbit part with a 16 bit data bus, a factor of 4 improvement in access time is required. Memory suppliers are starting to produce "Synchronous DRAM" which has the required access time. Parity in "cost/bit" between the "Standard 4 Mbit" parts and "Synchronous 16 Mbit" parts is expected in mid 1995.

The flexibility of the MPEG-2 syntax and the universality of the MPEG-2 decoder chips allow network operators to select the encoding techniques that best fit their applications and cost requirements. Although MPEG-2 supports all of the prediction modes described above, an operator could limit the memory in the decoder initially to hold down initial decoder costs. In the long term most likely all decoders will have 16 Mbits of memory as memory costs come down with time and the volume unit for memory becomes 16 Mbits.

III. DIGITAL AUDIO COMPRESSION

All of the digital audio compression techniques use the same fundamental principles of allocating the available bits in a fixed rate data stream according to the frequency content and dynamic range of the audio source material being compressed. All of the algorithms use a psychoacoustic model to set priorities dynamically for bit allocations in frequency and amplitude segments of the audio spectrum. In recent years, the two major contenders for digital audio approaches have been the MUSICAM group led by Philips and the Dolby Laboratories.

The audio compression algorithms chosen for the MPEG-1 standard for mono and stereo audio signals are based on the basic MUSICAM algorithm. It is this MPEG-1 stereo that has already been adopted internationally and is the core of the audio part of ISO/IEC 11172-3 audio standard. The proven high quality MPEG-1 stereo can provide the stereo pair for use for MPEG-2 video applications. Equipment is in production around the world using the ISO MPEG-1 audio standard for digitally compressed stereo.

Surround sound using stereo transmission channels can be provided for MPEG-2 video by the use of two channel matrix coded multichannel audio such as Dolby Laboratories Pro Logic® surround in combination with the MPEG-1 independent stereo coding. The MPEG-1 MUSICAM compression at compressed data rates of 256 kb/s and 384 kb/s for a stereo pair has sufficient transparency margin that it is both Pro Logic® compatible and capable of multiple cascades without causing audible coding artifacts.

In the recent completed formal listening tests conducted for MPEG, none of the 5 channel coders tested achieved transparency for all of the audio test selections at the tested bit rates. Although none of the coders achieved transparency, the performance of the PAC coder of AT&T and the AC3 coder of Dolby Laboratories indicated that higher quality audio could be achieved with techniques not backward compatible to MPEG-1 stereo. The desire to provide the best possible audio quality in multichannel audio within the MPEG standard has opened the door for a new multichannel extension in

MPEG-2 that is not backward compatible. This new mode will be pursued as a supplement to the necessary backward compatible modes. A call for new submissions has been issued by MPEG.

All of the multichannel coders are experiencing improvements in performance as further optimization is occurring and new encoding strategies are being tested. The standard has been constructed to allow these improvements to be made in the encoders without changing the transport and syntax that are specified in the standard.

The 5 channel digital audio compression standard for MPEG-2 has become a controversial issue with both technical and political disputes. Because the processing "engines" for all approaches in contention are very similar, it may be possible to develop a decoder chip set that will handle multiple approaches using a common processor and memory. The programs to execute the different approaches could be stored in separate ROMs.

IV. SYSTEM MULTIPLEX AND TRANSPORT

Outstanding progress has been made by the MPEG-2 System Sub-Committee in defining the required multiplex and transport approach for transmission of digitally compressed signals. Some of the most critical issues to resolve were the following:

1. Packet based approaches vs frame based approaches
2. Fixed length vs variable length packets and packet lengths
3. Number of bits available for overhead
4. How to map to MPEG into the Asynchronous Transfer Mode (ATM)
5. Program multiplexes imbedded in the system multiplex
6. Defining appropriate "hooks" for conditional access and encryption

The diagram in Figure 2 defines the basic packet and multiplex structure for which the MPEG-2 System Sub-Committee has reached consensus. A fixed packet of 188 bytes with a payload of 184 bytes has been designated. The key parts of the 4 byte header are the sync word and the prefix. The prefix includes error indicators, transport priority, packet identification and other indicator and control bits. The MPEG-2 system has been designed with an adaptation field for information that may be required for specific applications. Also, a single MPEG-2 transport packet can be fit into the payload of 4 ATM cells.

Key features of the MPEG-2 transport include the following:

1. Transport stream is independent of transmission data link to allow both terrestrial and satellite applications.
2. Synchronization of program service data (video, audio, etc.) is handled by MPEG-2 systems layer and is not dependent on transmission link timing.
3. Error protection requirements can be matched to transmission medium where errors actually occur.
4. MPEG-2 transport packets have error detection functions built in (priority bit, packet-error-indicator and packet continuity counter).
5. MPEG-2 systems packet-based transport structure allows for simple remultiplexing operations to be performed on data received over non-synchronized transmission data links.
6. MPEG-2 adaptation field allows flexibility in the handling of multiple conditional access streams, private data and other optional features.

Custom ICs for the MPEG-2 multiplex and transport are in the final stages of design and should be available for initial use in products by June of this year.

V. MODULATION AND ERROR CORRECTION

A. Satellite Applications

The Communications Industry both in the United States and in the rest of the world has adopted Quadrature Phase Shift Key

(QPSK) modulation for applications using digital video compression over satellite. The noise characteristics of the transmission path and the characteristics of amplifiers in satellite transponders make QPSK an optimum compromise between data rate capacity in a given transponder and ruggeddness at practical threshold levels at the receiving terminal.

Concatenated Viterbi and Reed-Solomon error correction coding is used with interleaving to reduce the effect of burst errors. Trade-offs can be made between the amount of the total digital data stream used for error correction, the maximum information data rate and the receiver threshold. Systems have been designed with rate 3/4 Viterbi coding (1/4 total data rate used for error correction) and with rate 7/8 Viterbi coding (1/8 of the total data rate used for error correction coding). At rate 3/4 the information rate in a typical 36 MHz C-band transponder is 27 Mb/s. At rate 7/8 the information rate in a comparable C-band transponder is 38 Mb/s. All things being equal the threshold at the receiving site is approximately 2 dB lower for rate 3/4 than for rate 7/8. This difference in threshold can be significant or insignificant depending on the power of the satellite used and the antenna gain and LNA noise temperature at the receiving site.

B. Broadband Cable Applications

The transmission path characteristics over broadband cable in the forward direction and the requirement to maximize information data rate in a given bandwidth led to a choice of multi level AM for modulation with only Reed-Solomon error correction.

Quadrature Amplitude Modulation (QAM) and Vestigial Side Band Amplitude Modulation (VSBAM) are the multi level AM techniques currently under consideration. Both QAM and VSBAM are well known techniques that have been utilized in numerous applications for many years. Scientific-Atlanta uses 4 VSBAM to modulate digital audio in the horizontal blanking interval in the B-MAC System. Multi level QAM has been used in many applications over the years including digital microwave radio and cable modems.

At a CableLab's seminar on digital modulation for broadband cable in December 1993 all of the presenters except one supported QAM. Many concluded that QAM had both performance and cost advantages over other modulation approaches for digital applications over broadband cable. The other presenter, Zenith Corporation, advocated VSBAM over other approaches. Zenith has implemented an 8 VSB system for trials to determine the broadcast standard for digital HDTV and has demonstrated a 16 VSB system over broadband cable.

The United States digital HDTV Grand Alliance has selected 8 VSB as the broadcast modulation for digital HDTV based on tests conducted of Zenith's implementation of 8 VSB and other implementations of 32 QAM. The tests show that Zenith's implementation of VSB was superior to the implementations of QAM. One reason for the test results is that Zenith used 8 VSB with a much higher overhead than the 32 QAM implementation. Theoretically 8 VSB has the same data rate as 64 QAM. See Figure 3 for comparison of rates for VSB and QAM. Zenith used the difference in rates between 8 VSB and 32 QAM for error correction overhead. A second reason for the test results is that Zenith incorporated a comb filter to reject rf carriers and color sub-carriers for NTSC to minimize co-channel interference. Co-channel interference is an issue in over-the-air broadcast and not in broadband cable applications.

Head-to-head tests of optimized implementations of 8 VSB vs 64 QAM and 16 VSB vs 256 QAM over broadband cable have not been conducted by independent testing groups. Many tests and simulations to date suggest that 64 QAM and 256 QAM have performance advantages on cable over Zenith's implementation of 8 VSB and 16 VSB for the following reasons:

1. Zenith's implementation requires a pilot signal with each channel to accomplish carrier recovery in the receiver. This pilot signal uses some

of the total energy available and has a significant effect in a multi channel broadband applications.

2. The Zenith implementation requires a very sharp filter with an alpha of 11%. In the QAM applications larger alphas are used which should improve the performance in a multi channel environment and reduce the cost of filters in the decoders.

Calculations show that the proposed VSB approach will have approximately 2 dB poorer Signal-to-Noise (S/N) in a broadband cable plant than the proposed QAM approaches. As shown in Figure 4, overshoot produced from Gibb's phenomenon from sharply truncated channels will require 1.2 dB more "back off" to eliminate amplitude clipping. This difference in overshoot results from the difference in alpha: the excess bandwidth ratio. The energy used in the pilot carrier in the VSB approach reduces S/N by another approximately 0.7 dB.

The European Digital Video Broadcast (DVB) group formerly called the European launching group has selected 64 QAM as the standard for digital video modulation in Europe. This decision was made after extensive simulation and testing by the members of the group. Although the European DVB has agreed on standards for the details of the Viterbi, interleaving and Reed-Solomon codes, complete agreement has not been reached in the United States on details.

Statements advocating compatibility between the modulation technique for over-the- air broadcast for digital HDTV and digital video compression modulation over broadband cable often fail to mention the real issues. The modulation scheme for digital HDTV is 8 VSB with a specialized trellis code for error correction that provides an information rate that is essentially no higher than 4 VSB or 16 QAM with error correction needed for cable. The information rate for the HDTV signal is only approximately 19 Mb/s in the 6 MHz channel. This information rate would be totally unacceptable in a broadband cable environment. Therefore, the 8 VSB implementation used for digital HDTV would not be acceptable in broadband cable. If a different implementation of VSB were used for cable than for digital HDTV then direct compatibility would not occur even though an 8 VSB demod with some extra circuitry will do 16 VSB. The issues of performance in the broadband cable plant and cost are much more important than a digital HDTV set with a dual mode VSB demodulator. Initial deployments by Scientific-Atlanta and General Instrument for applications over broadband cable will use QAM.

Scientific-Atlanta has tested 64 QAM in the fiber-to-the-serving-area network in the Time Warner system in Orlando with an IC implementation for demodulation and equalization. The decoder utilizes a three chip set: an equalization IC, a demodulation IC and a control IC. The full custom chip set is capable of either 64 QAM or 256 QAM operation. The test results show this 64 QAM implementation operates within 1 dB of theory and yields the desired Bit Error Rates (BER) with the appropriate Reed-Solomon Forward Error Correction (FEC).

The 256 QAM mode has been tested in the laboratory back-to-back with the custom IC implementation. These tests also show operation within 1 dB of theory. Field tests of the 256 QAM mode in an actual FSA Network will be made in the coming weeks.

VI. SECURITY AND CONDITIONAL ACCESS

It is usually instructive to consider security and conditional access in two major parts: (1) the encryption algorithm used to encrypt program content and (2) the conditional access data stream to distribute keys or information to generate keys. Usually the encryption keys used to encrypt the program content material are called seeds and are changed very rapidly to achieve temporal security. Temporal security can be thought of as changing a key so often that a pirate can not practically discover the nature of the key before it has been changed at the next time cycle. Often these seed keys are changed with time cycles of less than one second.

If a common seed and a common encryption algorithm is used to encrypt all program content, multiple conditional access data streams can be used to distribute the seeds or the necessary information to generate the seeds. The seeds in the conditional access data stream can be encrypted with different keys and different encryption algorithms from those used for the program content. Different program providers can use their own proprietary conditional access data stream approach or could agree on common conditional access data streams. Special provisions would have to be taken in any system with multiple conditional access data streams to prevent pirates who might break one conditional access data stream to create clones that could be used universally for all conditional access data streams.

The current approach being proposed by the European DVB is to have a common encryption algorithm for program content but to allow multiple proprietary conditional access data streams. There is strong political pressure in Europe to have multiple proprietary conditional access data streams to allow programmers to protect and control their own subscriber base without automatically giving other programmers access.

One practical approach to Interoperability for security and conditional access is the digital equivalent of the analog system we use today in the United States. There is a defacto standard over satellite and then each local cable operator selects from one of multiple vendor choices for his local security and conditional access system in his cable system. The local cable operator could likewise in a digital system decrypt the signals received over satellite and have a different local security and conditional access for both digital and analog signals in his system. This approach of different independent security and conditional access systems also reduces the possibility of a wide spread catastrophic pirate break of a common system. A clone in one local system would not be transferable as a clone in another system with a different security and conditional access system.

VII. NETWORK OPERATING SYSTEM

Even if we resolve all the issues in the previous areas that are required for Interoperability, we would not have Interoperable equipment or networks unless the industry develops an Interoperable network operating system. An operating system is essentially a collection of software programs in the headend and in the processor in the Home Communications Terminal (HCT) that lets a particular service provider or programmer deliver his services and programs to all subscribers regardless of whose HCT or headend equipment is in his system. The analog in the computer world is an operating system such as MS DOS in which any application software written to the MS DOS interfaces and specifications will operate on a given PC using that same version of MS DOS.

The part of the operating system that is in the HCT must take high level commands from a given application software program and convert them to software drivers for the various hardware and software functions in the HCT. For example, if a specific application requires digital data from a specific channel, the operating system must interpret the request for data and give the commands within the HCT to set the tuner to the correct frequency and to extract the data from a specific multiplex within that frequency channel. Also, if conditional access is involved, the operating system must check that the appropriate conditional access authorizations have been granted.

Many companies are working diligently to develop operating systems in hope that they will become the Microsoft of the "Information Superhighway". However, one of he major challenges for the operating system that will achieve practical implementation is to achieve the required performance without pushing the processing speed and memory size in the HCT beyond an acceptable cost point. The price of the HCT must be consistent with the anticipated revenue it generates. Another part of this trade-off is to determine how fundamental the Application Program Interface (API) can be to keep processing and memory cost low without inhibiting the wide spread development of creative third party applications.

Interoperability will be best served if the hardware and software designers of the products and systems in the Information Superhighway adhere to the ISO's Open System Interface (OSI) seven layer model for the system design structure. Interface specifications will be necessary at critical interfaces within the products and at defined interfaces in the system. Use of the seven layer OSI model will help facilitate the development of these interface specifications for Interoperability.

VIII. CONCLUDING REMARKS

The industry has come a long way toward Interoperability in the last two years. But, it is obvious that we still have a long way to go. I believe that action in at least the following four areas will ultimately produce Interoperability:

1. Strong industry standards groups such as MPEG-2
2. Industry consensus such as occurred for QPSK modulation on satellite
3. Industry pressure for multiple suppliers
4. Government and regulatory pressures for Interoperability. For example, current FCC regulations on video dial tone for Bell Operating Companies (BOCs).

I believe that Interoperability is inevitable because both the cable MSOs and telco BOCs will demand it. It may be a painful process and there may be costly false starts. I am convinced, however, that the only way to realize the vision of the "Information Superhighway" and the rewards it can bring to the Communications Industry and the consumer is to push for Interoperability as quickly as possible.

Prediction Mode	Picture Quality Ranking	Frame Store Memory Required
B-Frame	1	16 Mbits
P-Frame Dual Prime	2	8 Mbits*
P-Frame Standard	3	8 Mbits
I-Frame Only	4	N/A

*Requires memory with faster access than other modes

Figure 1 Comparison Of Prediction Modes For Picture Quality And Memory Requirements

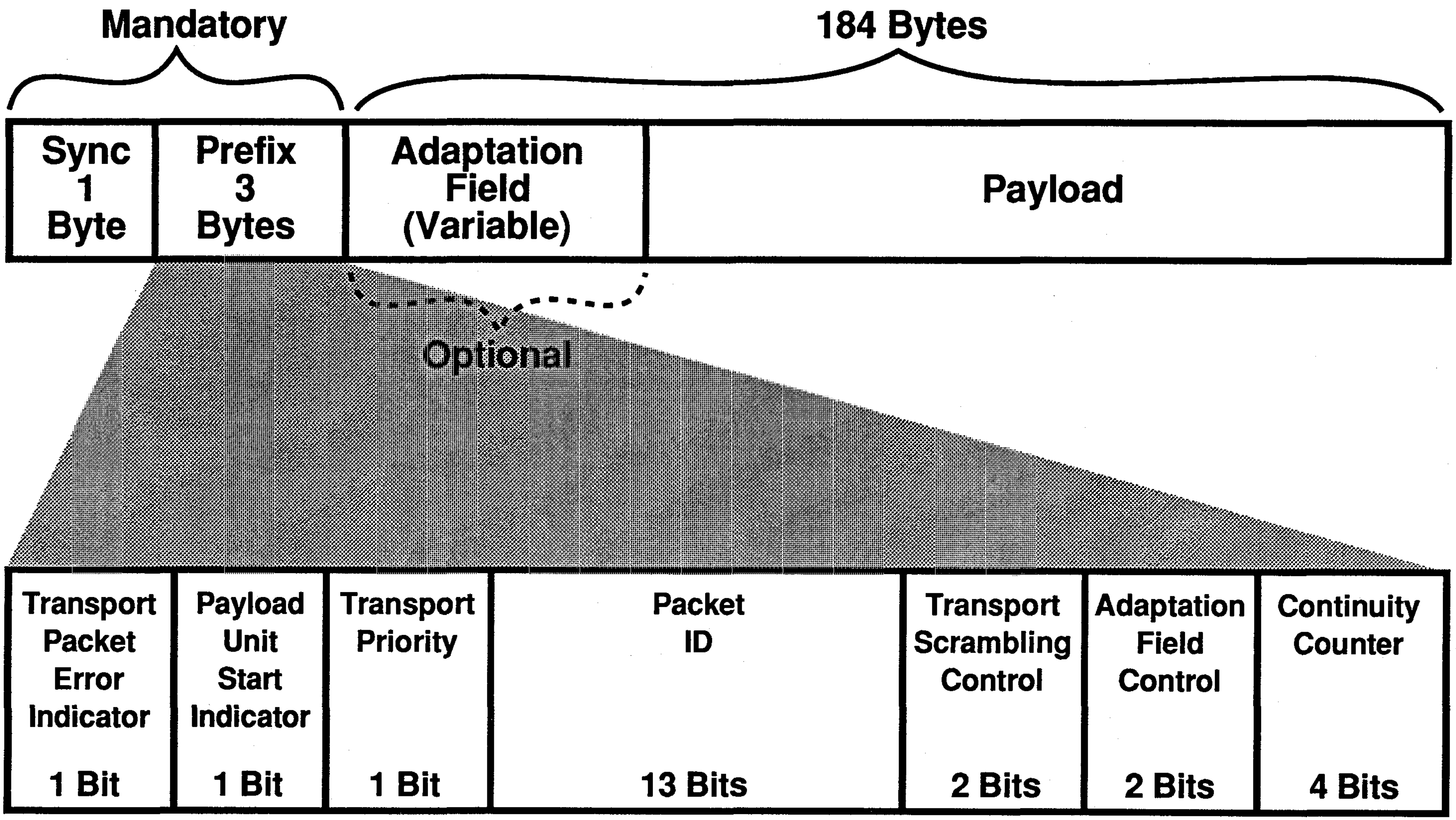

Figure 2 MPEG-2 Transport Packet Format

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M0394-18

QAM	Bits per Hz: Theory	Bits per Hz: Practical	VSB-AM
16 QAM	4	3.3 - 3.5	4 VSB
64 QAM	6	5.0 - 5.2	8 VSB
256 QAM	8	6.7 - 7.0	16 VSB

Bit rates in TV channel (6 MHz)

8 VSB or 64 QAM	16 VSB or 256 QAM
~ 30 Mb/s	~ 40 Mb/s

Figure 3 What Bit Rates Are Possible?

M0394-20

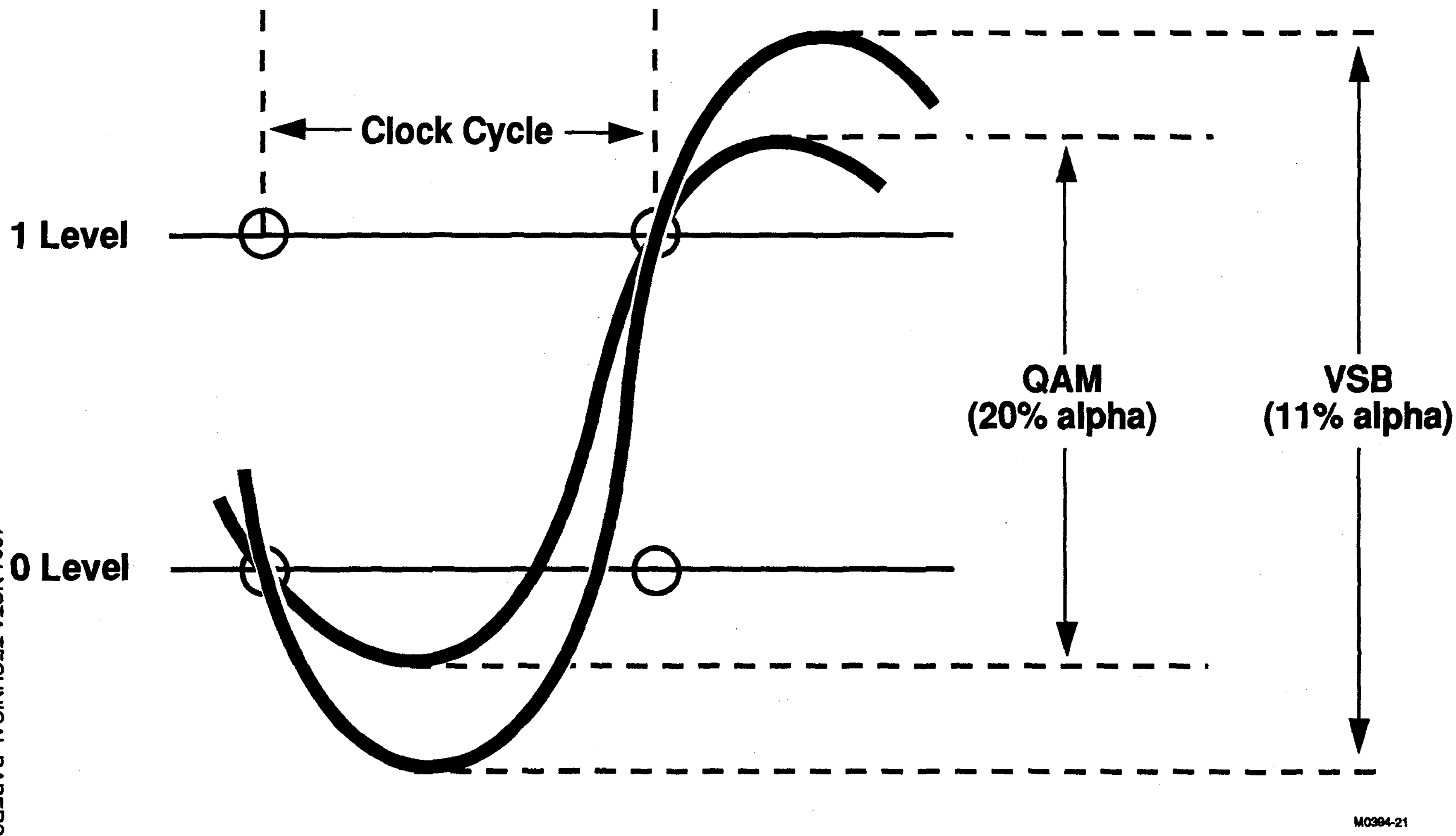

Figure 4 Overshoot for VSB (11% alpha) vs. QAM (20% alpha) with resulting 1.2 dB difference in peak power