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1 Introduction

During this period, we began the devolopment of simulators for the various HDTV systems proposed to the FCC. This simulators will be tested using test sequences from the MPEG committee. The results will be extrapolated to HDTV video sequences.

Currently we have completed the simulator for the compression aspects of the *Advanced Digital Television* (ADTV) proposed by David Sarnoff Laboratories, NBC, North American Phillips, and Thomson Consumer Electronics. This proposal has also been supported by Texas Instruments. We are at various stages of development in terms of the other HDTV proposals.

In the following section we give a brief overview of the ADTV system. In section 3, we look at some coding results obtained using the simulator. We compare these results to those obtained using the CCITT H.261 standard. In section 4, we evaluate these results in the context of the CCSDS specifications and make some suggestions as to how the ADTV system could be implemented in the NASA network.

Some caveats are in order here. The sequences we used for testing the simulator and generating the results are those used for testing the MPEG algorithm. The sequences are of much lower resolution than the HDTV sequences would be, and therefore the extrapolations are not totally accurate. We would expect to get significantly higher comperession in terms of bits per pixel with sequences that are of higher resolution. However, the simulator itself is a valid one, and should HDTV sequences become available, they could be used directly with the simulator.

In addition to the work described in this report, the following papers appeared or were submitted during the past six months, describing other work under this grant and its predecessor. The papers are included in the appendix.

- Y.G. Chen, K. Sayood, and D.J. Nelson, "A Robust Coding Scheme for Packet Video", *IEEE Transactions on Communications*, vol. 40, pp. 1491-1501, September 1992.
- K. Sayood, "Data Compression in Remote Sensing Applications," *IEEE Geoscience and Remote Sensing Society Newsletter*, no. 84, pp. 7-15, September 1992.

- A.C. Hadenfeldt and K. Sayood, "Compression of Color-Mapped Images," submitted to *IEEE Transactions on Geoscience and Remote Sensing*

- K. Sayood and S. Na, "Recursively Indexed Differential Pulse Code Modulation," *Proceedings IEEE/DIMACS Workshop on Quantization and Coding*, Piscataway NJ, November 1992.

- K. Sayood, F. Liu, and J.D. Gibson, "A Joint Source Channel Coder Design," accepted for 1993 International Conference on Communication, Geneva, Switzerland, May 1993.

- A.C. Hadenfeldt and K. Sayood, "Compresion of Color-Mapped Images," accepted for *1993 International Conference on Communication*, Geneva, Switzerland, May 1993.

- B. Gorjala, K. Sayood, and G. Meempat, "An Image Compression Scheme for use in Token Ring Networks", submitted to *1993 International Conference on Communication*, Geneva, Switzerland, May 1993.

2 Advanced Digital Television

There are three key elements in the ADTV system.

- ADTV uses MPEG++(Moving Pictures Expert Group) draft proposal as its compression scheme.

- ADTV incorporates a Prioritized Data Transport(PDT) which is a cell relay-based data transport layer to supports the prioritized delivery of video data. PDT also offers service flexibility and compatibility to broadband ISDN.

- ADTV applies spectral-shaping techniques to Quadrature Amplitude Modulation(QAM) to minimize interference from and to any co-channel NTSC signals.

We have simulated all aspects of the compression algorithm of the ADTV proposal. The compression algorithm as described in the *Advanced Digital*

Television, System Description submitted to FCC/ACATS and as implemented in the simulator is described below.

2.1 Compression Algorithm

The basic compression approach is the MPEG++ algorithm which upgrades the standard MPEG approach to HDTV performance level. The key components of this algorithm are described below.

2.1.1 Group of Pictures(GOP)

A GOP comprises up to three types of frames, the I, P and B frames. The I frames are processed using only intra-frame DCT coder with adaptive quantization; the P frames are processed using a hybrid temporal predictive DCT coder with adaptive quantization and forward motion compensation; the B frames are processed using a hybrid temporal predictive DCT coder with adaptive quantization and bidirectional motion compensation. The I and P frames are referred to as the anchor frames because of their roles in the bidirectional motion compensation of the B frames. The GOP structure offers a good tradeoff between the high efficiency of temporal predictive coding, good error-concealment features of periodic intra-only processing, and fast picture acquisition.

2.1.2 Input Sequencer

The GOP data structure requires some unique sequencing of the input video frames. Because of the backward motion compensation in B frames processing, the anchor frames must be processed before the B frames associated with the two anchors. The frames are transmitted in the same order as they are processed.

2.1.3 Raster Line to Block/Macroblock Converter

The basic DCT transform unit is an 8 x 8 pixel block called a block. The basic quantization unit is four adjacent blocks of Y, and one U and one

V blocks. Such a quantization unit is called a macroblock. The converter converts the raster line format to the block and the macroblock format.

2.1.4 I-frame Processing

An I frame is processed by an intra-frame DCT coder without motion compensation. A fixed quantizer is applied to the DC coefficient. The AC values are first weighted by a down- loadable quantization matrix before uniform adaptive quantization. The quantization step for the AC coefficients is controlled by a Rate Controller. The I frame coding is pretty much the same as JPEG scheme.

2.1.5 P-frame Processing

A P frame is first processed by forward motion compensation, motion is always referenced to the nearest past anchor frame. The search area is proportional to the number of B frames between two consecutive anchor frames. The prediction residue or original macroblock, depending on the motion compensation result, goes to DCT coder and quantizer. For intra-macroblocks, the DCT coefficient quantization is identical to that used for the I frames. For motion-compensated macroblocks, the DC and AC coefficients are quantized with same uniform quantizer.

2.1.6 B-frame Processing

Unlike the P frames, the B frames are subjected to bidirectional motion compensation. The motion references are the two anchor frames sandwiching the B frames. The search regions are proportional to the temporal distance between the B frame and the two anchor frames. Like P-frame macroblock, the B-frame macroblocks have a number of modes. In addition to all the modes for a P-frame macroblock, the B-frame macroblock further includes a bidirectional interpolative mode, using both forward and backward motion compensation, and a unidirectional mode. In the interpolative mode, an average of the forward and the backward motion- compensated macroblocks is used as the prediction macroblock. The B-frame macroblock is processed as a P-frame macroblock.

2.1.7 Differential, Run-Length and Variable-Length Coding(VLC)

The quantized DC coefficients of all the I-frame macroblock and P-, B-frame macroblocks in intra mode are coded with a DPCM coder. The quantized AC coefficients are coded with VLC after the zigzag scan ordering. Motion vectors are differentially coded. In addition, VLC is applied to all the coded information: motion vectors, macroblock addresses, block types, etc.

2.2 Data Prioritization Layer

The Prioritization Layer comprises the Priority Processor and the Rate Controller.

2.2.1 Priority Processor

Based on the information from the Rate Controller, the Priority processor pre-calculates the rate of HP(High Priority)/LP(Low Priority) for every frame. HP/LP fractional allocations may vary with the frame type. Every data element gets a priority assignment from the Priority Processor according to its importance. The header is always most important, followed by the motion vector, DC value, low frequency coefficients and high frequency coefficients.

2.2.2 Rate Controller

The Rate Controller monitors the status of the rate buffers in the Transport Encoder. It uses the buffer occupancy information to compute the necessary compression requirement and feeds the results in the form of appropriate quantization parameters to the Video Processor in the Compression Encoder. The Rate Controller also provides input to the Priority Processor regarding the initial allocation of HP/LP rate for the next Group of Pictures.

The algorithm used in this simulation for rate control is given by

$$QS = 2\lceil \frac{B}{200p} \rceil + 2$$

where QS is the quantizer step-size, and B is a measure of buffer fullness.

2.3 Transport Layer

The Transport Layer comprises the Transport Processor and the Rate Buffer.

2.3.1 Transport Processor

Data elements are supplied to the Transport Processor from the Prioritization Processor. The Transport Processor generates appropriate header fields for data group. The header fields are used in the construction of a basic transport unit called cell. A cell has a header and a trailer enclosing a payload area. Each cell has a fixed size of 256 bytes long. The header contains chaining and segmentation information which allows data groups to be segmented across cells. This feature limits the propagation of channel error from one cell to the next. The trailer field contains 16-bit error-checking CRC code.

2.3.2 Rate Buffer

Since the number of cells generated is not constant and the channel coding module interfaces with the Transport Processor at a fixed clock rate, we need a buffer to smooth out any rate variation. The maximum end-to-end delay is dependent on the size of the buffer.

3 Simulation Results

The ADTV system described above without the priority and transport processors was simulated in detail. The simulation programs were written in *C* and implemented on a *SUN* workstation. Along with the ADTV system we also simulated a video coding scheme based on the CCITT H.261 recommendations. The purpose was to have a benchmark for simulation.

In our ADTV simulator, the frames were arranged in the following sequence

I B B P B B P B B P B B I B B P...

The sequence used for testing the simulator was the *Susie* sequence. This sequence contains both low and moderate motion of the type to be encountered in most NASA applications. We present the results in the form of graphs tables and a videotape accompanying this report.

The coding rates and PSNR under different coding conditions are listed in Table 1. Two parameters are used to control the coding conditions. The parameter p controls the output rate and length of the rate buffer. The fullness of the rate buffer determines the quantizer step-size and therefore, the coding rate and quality. Thus the parameter p has an important impact on both coding rate and quality. The parameter t is used to decide whether the macroblock after motion compensation needs coding. Smaller values of t will lead to higher rates, while smaller values of t will result in lower quality.

The first three sequences were coded using the ADTV algorithm. In Sequence 1, with p=3, the average rate is is 0.22 bits/pixel. This rate is not sufficient to effectively code the I frames. As the B and P frames depend heavily on the I frames, this has a cascade effect on the entire sequence. As the quantizer stepsize depends on how full the buffer is, this low rate leads to the buffer getting filled up as the lower portions of the I frame are being coded. This means that when coding the lower regions of the I frames, the quantizer is coarse. This results in blocking artifacts which are very noticeable in the lower portions of the sequence. This effect can be seen in the first sequence on the videotape. When the coder has finished with the I frame, and the B and P frames are being coded, as the coding rate is lower for the B and P frames, the buffer situation gets partly remedied. However, this is not sufficient to get the quantizer step-size small enough to remove the blocking effect. When the buffer size is increased (p=5) the number of artifacts is reduced as seen in the second sequence on the video tape. An interesting effect can be seen in the third sequence. Here the buffer size was kept the same as in the second sequence, however, the motion compensation threshold t was kept high. This means that blocks that would have been coded in sequence 2 are left uncoded in sequence 3. This in turn emphasizes the blocking effects. One would think that given the fact that we are accepting more distortion, the rate would go down. However as we can see from Figure 1, the rates for sequence 2 and sequence 3 are almost identical. This could be attributable to the fact that the poor reconstruction of the P frames lead to poor prediction and hence an increase in bit rate which takes away any savings from the higher motion compensation threshold. Thus the parameter t while it effects the quality has little effect on the rate.

From Figure 1 we can see that the ADTV algorithm generates a very bursty traffic. This is in sharp contrast to the CCITT H.261 algorithm which produces relatively smooth output. We can see this from the results for sequences 4-7 which were coded using the CCITT H.261 algorithm using the same parameters p and t as the first three sequences. The rate and PSNR results for these sequences are given in Table 1 and Figures 3 and 4. Recall that the only significant difference between the CCITT H.261 algorithm and the ADTV algorithm are the sequencing and motion compensation. The use of the intra-frame coding every 13^{th} frame in the ADTV algorithm increases the bit rate. This is compensated for by using the different motion compensation approach giving the bursty traffic. In the CCITT H.261 algorithm, intra-frame coding is recommended once in every 134 frames, thus there is no significant variation in the rate from frame to frame. The disadvantage of the CCITT H.261 algorithm when compared to the ADTV algorithm is the decrease in the ability to randomly access any particular frame, and the decrease in the ability to react fast to sudden scene changes. Furhtermore it should be noted that the ADTV algorithm was proposed for HDTV sequences, while the sequences we are using have significantly less resolution.

Due to the importance of I frame, which serves as the anchor frame for both P and B frame, we decide to put more coding efforts in such frame to try to eliminate the blocking effect. In Sequence 7-9, the ADTV algorithm has been modified to keep the quantization stepsize QS constant while coding the I frame. One effect is that the buffer becomes really full during coding the I frame, and the subsequent frame gets very little of the coding resources. This results in an increase in burstiness as can be seen from Figures 5 and 6. However, this approach does result in the reduction/elimination of the blocking effect. That such a simple strategy can result in such dramatic improvement shows that should blocking effects appear in the HDTV sequences, attention should be paid to the encding of the I frames.

Figures 7-14 show various comparison results between the ADTV, The modified ADTV and the CCITT H.261 algorithms. While these comparisons show an advantage for the CCITT H.261 algorithm, subjective comparisons tend to show the reverse. We invite the reader to examine the videotape and draw their own conclusions.

4 HDTV and CCSDS

To speculate how the HDTV service would be accomodated by the NASA network, we briefly review some of the relevant features of the CCSDS recommendations.

4.1 CCSDS Principal Network

A "CCSDS Principal Network" (CPN) serves as the project data handling network which provides end-to-end data flow in support of the Experimental, Observational and interactive users of Advanced Orbiting Systems. A CPN consists of an "Onboard Network" in an orbiting segment connected through a CCSDS "Space Link Subnetwork" (SLS) either to a "Ground Network" or to another Onboard Network in another orbiting segment. The SLS is the central component of a CPN; it is unique to the space mission environment and provides customized services and data communications protocols. Within the SLS, CCSDS defines a full protocol to achieve "cross support" between agencies. Cross support is defined as the capability for one space agency to bidirectionally transfer another agency's data between ground and space systems using its own transmission resources. A key feature of this protocol is the concept of a "Virtual Channel" which allows one physical space channel to be shared among several data streams, each of them may have different service requirements. A single Physical space channel may therefore be divided into several logical data channels, each known as a Virtual Channel.

Eight separate services are provided within a CPN. Two of these services ("Path" and "Internet") operate end-to-end across the entire CPN. They are complementary services, which satisfy different user data communications requirements: some users will interface with only one of them, but many will operate with both. The remaining six services ("Encapsulation", "Multiplexing", "Bitstream", "Insert", "Virtual Channel Data Unit" and "Physical Channel") are provided only within the Space Link Subnetwork for special applications such as audio, video, high rate payloads, tape playback, and the intermediate transfer of Path and Internet data. Our interest is with the services provided by the Space Link Subnetwork.

4.2 Space Link Subnet Services

The Space Link Subnet supports the bidirectional transmission of data through the space/ground and space/space channels which interconnect the distributed elements of the CCSDS Advanced Orbiting Systems. It also provides "direct connect" transmission services for certain types of data which requires timely or high-rate access to the space channel. During SLS transfer, different flows of data are separated into different Virtual Channels, based on data handling requirements at the destination. These Virtual Channels are interleaved onto the physical channel as a serial symbol stream. A particular Virtual Channel may contain either packetized or bitstream data, or a combination of both.

The Space link Subnetwork consists of two layers; the Space Link layer and the Space Channel layer which correspond to ISO-equivalent Data Link layer and Physical layer respectively. Efficient use of the physical space channel was a primary driver in the development of these protocols. The Space Link layer is composed of the Virtual Channel Link Control sublayer(VCLC) and the Virtual Channel Access sublayer(VCA).

The main function of VCLC sublayer is to convert incoming data into a protocol data unit which is suitable for transmission over the physical space channel. Four type of protocol data units may be generated by the VCLC sublayer: fixed length blocks of CCSDS Packets, called "Multiplexing Protocol Data Units" (M-PDUs); fixed length blocks of Bitstream data, called "Bitstream Protocol Data Units" (B-PDUs); fixed length blocks of mixed packetized and isochronous data, called "Insert Protocol Data Units" (IN-PDUs); and fixed length blocks of data for use by retransmission control procedures, called "Space Link ARQ Procedure Protocol Data Units" (SLAP-PDUs).

There are several procedures in VCLC sublayer to perform the function. The Encapsulation procedure provides the flexibility to handle virtually any packet structure. It puts a primary header to delimited data units (including Internet packet) and make it to be a CCSDS Packet. Multiplexing procedure multiplexes those CCSDS Packets on the same virtual channel together. The length of multiplexing protocol data unit is fixed since it is required to fit exactly in the fixed length data space of VCDU/CVCDU. There maybe some packets which overlap two or more M-PDU and "first packet pointer" points out where the first packet starts. Some user data, such as audio, video, playback and encrypted information, will simply be presented to the SLS as a stream of bits or octets. The Bitstream procedure simply blocks these data into individual Virtual Channels and transmits it. When the transmission rate is high, Bitstream data may be transmitted over a dedicated Virtual Channel. Alternatively, if the transmission rate is low, it can be inserted at the front of other packetized or bitstream data. This is called the Insert procedure. Through this procedure, bandwidth can be used more efficiently.

The last procedure of VCLC is Space Link ARQ Procedure (SLAP) which is used to provide guaranteed Grade-1 delivery of data links that interconnect the space and ground elements of a CPN. The SLAP-PDU carries "Link ARQ Control Words" (LACWs) which report progress on receipt of data flowing in the opposite direction. Upon arrival at the receiving end, the LACW is extracted from the PDU, and the sequence number is checked to assure that no data has been lost or duplicated. In the event of a sequence error, the LACW carried by PDUs traveling in the opposite direction is used to signal that a retransmission is required. This retransmission begins with the first PDU that was not received in sequence, and all subsequent PDUs are retransmitted in the order in which they were originally provided to the LSAP from the layer above.

The VCA sublayer creates the protocol data units used for space link data transfer: these are either "Virtual Channel Data Units" (VCDUs) or "Coded Virtual Channel Data Units" (CVCDUs), and are formed by appending fixed length Header, Trailer and (for CVCDUs) error correction fields to the fixed length data units generated by the VCLC sublayer. The VCA sublayer is composed of Virtual Channel Access(VCA) and Physical Channel Access(PCA) procedures. VCA procedure generates VCDU for protocol data units which come from VCLC sublayer or accepts independently generated VCDU from reliable users. A VCDU with a powerful outer code of error-correcting Reed-Solomon check symbols appended to it is called a CVCDU. Relative to a VCDU, a CVCDU contains more error-control information and, hence, less user data. "Virtual Channel ID" which is field of VCDU/CVCDU

Header can enable up to 64 VC to be run concurrently for each assigned Spacecraft ID on a particular physical space channel. Since space data is transmitted through weak signal, noisy channel as serial symbol stream, a robust frame synchronization process at the receiving end is required. Therefore, fixed length VCDU/CVCDU is used and PCA procedure prefixes a 32 bits Synchronization Marker in front of VCDU/CVCDU to form a "Channel Access Data Unit" (CADU). A contiguous and continuous stream of fixed length CADUs, known as a "Physical Channel Access Protocol Data Unit" (PCA-PDU) is transmitted as individual channel symbols through the ISO-equivalent Physical Layer of the Space Link Subnet, which is known as the "Space Channel Layer".

4.3 Space Link "Grades of Service"

Three different "grades of services" are provided by the Space Link Subnet, using a combination of error detection, error correction and retransmission control techniques. We have to note that each virtual channel can only support a single grade of service.

4.3.1 Grade-3 Service

This service provides the lowest quality of service. Data transmitted using Grade-3 service may be incomplete and there is a moderate probability that errors induced by the Space Link Subnet are present and that the sequence of data units is not preserved. A VCUD is discarded if an uncorrectable error is detected at the destination. Grade-3 service should not be used for transmission of asynchronous packetized data, because it provides insufficient protection for the extensive control information contained in the packet headers.

4.3.2 Grade-2 Service

CVCDU is the unit of transmission that support Grade-2 service. The Reed-Solomon encoding provides extremely powerful error correction capabilities. Data transmitted using Grade-2 service may be incomplete, but data sequencing is preserved and there is a very high probability that no data errors have been induced by the Space Link Subnet.

4.3.3 Grade-1 Service

Data transmitted using Grade-1 service are delivered through the Space Link Subnet complete, in sequence, without duplication and with a very high probability of containing no errors. It is provided by using two paired Reed-Solomon encoded Virtual Channels, in opposite directions, so that an Automatic Repeat Queueing (ARQ) retransmission scheme may be implemented.

5 HDTV Transmission on the CCSDS Network

As described in the previous section, some user data, such as audio, video, playback and encrypted information, can simply be presented to the Space Link Subnet (SLS) as a stream of bits or octets. The SLS merely blocks these data into individual Virtual Channels and transmits them using Bitstream Service. Some bitstream data, such as digitized video and audio, will have stringent delivery timing requirement and are known as "isochronous" data. For the transmission of ADTV coded information, the channel transmission rate is high enough to dedicate a specific Virtual Channel. Although the coding output rate is quite bursty, there are two mitigating circumstances

1. the pattern of burstiness is relativey "uniform". That is, the data rate peaks every 13^{th} frame.

2. the variations occur very fast, that is high traffic persists for only a single frame followed by low traffic.

Because of (2) the traffic can be smoothed out using a moderate sized buffer, and (1) implies that the size of the buffer can be ascertained with some confidence.

The delay constraints on the transmission preclude the use of the Space Link ARQ procedure, while the delay constraint coupled with the high rate argue against the use of the Insert and Multiplexing procedures. This leaves the Bitstream procedure as the only viable candidate for HDTV transmission. This conclusion coincides with that of the CCSDS Red Book *Audio, Video, and Still-Image Communication Services*

The Bitstream service fills the data field of the B-PDU (Bitstream- Protocol Data Unit) with the Bitstream data supplied at user's request. Each B-PDU contains data for only one VC, identified by the VCDU-ID parameter. Each bit is placed sequentially, and unchanged, into the B-PDU data field. When the Bitstream data have filled one particular B- PDU, the continuation of the Bitstream data is placed in the next B-PDU on the same VC. Due to the delay constraints of the PDU release algorithm, if a B-PDU is not completely filled by Bitstream data at release time, some fill pattern has to be filled into the remainder of the B-PDU.

As far as the grade of service is concerned, one could use the error protection service provided by the ADTV algorithm with grade 3 service, or discard any error protection from the ADTV signal and use grade-2 service. Given the sketchy amount of information available abouthforward error correction in the ADTV algorithm we would suggest the use of the Grade-2 service in the CCSDS recommendations. Some kind of forward error correction is imperative because of the need for data sequencing along with the general need for video integrity. Therefore, Grade-2 service which adopts Reed-Soloman encoding is a logical choice. According to the minimum predicted performance of Grade-2 service, the probability that a Coded Virtual Channel Data Unit (CVCDU) will be missing is 10^{-7}. If we assume a CVCDU contains 8800 bits of data, from our simulation, about 95 macroblocks of video data (for ADTV format, a frame is formed by 90Hx60V macroblocks) will get lost in a duration slightly over one and half hour. This shouldn't hurt the quality too much in motion compensation scheme. The probability that a CVCDU contains an undetected bit error is 10^{-12}, only one bit error will occur in a transmission period over 11 hours. If this error bit occurs in video data, it won't be easy to notice the degradation. But if the error bit occurs in control data, some degree of damage is inevitable. It may therefore be desirable to provide some more protection to the control data before it enters the network. We are still looking at this particular issue.

Table 1. Performance of coding rate and PSNR using ADTV and H.261 technique
(Susie Sequence 150 frames)

	Coding Rate (Mbits/s)	STDR* (Mbits/s)	PAR**	Average PSNR	STDP*
Sequence 1	9.27	2.12	1.68	35.14	4.24
Sequence 2	15.34	2.26	1.33	37.64	4.52
Sequence 3	15.33	2.31	1.33	37.67	4.52
Sequence 4	9.39	0.61	1.65	36.83	0.81
Sequence 5	15.54	0.48	1.31	39.05	0.79
Sequence 6	15.54	0.49	1.31	38.42	0.49
Sequence 7	31.23	8.17	1.86	41.31	1.43
Sequence 8	17.27	10.72	3.37	38.77	1.68
Sequence 9	17.24	10.86	3.37	38.85	1.61

* STDR : Standard deviation of coding rate

STDP : Standard deviation of PSNR

** PAR : Peak to average ratio

frames/second : 29.97

Act. video pixels : (Luma) 1440Hx960V, (Chroma) 720Hx480V

Sequence 1 : ADTV, p=3, t=1

Sequence 2 : ADTV, p=5, t=1

Sequence 3 : ADTV, p=5, t=3

Sequence 4 : H.261, p=3, t=1

Sequence 5 : H.261, p=5, t=1

Sequence 6 : H.261, p=5, t=3

Sequence 7 : ADTV, p=10, t=1, q=4 for intra-mode frame

Sequence 8 : ADTV, p=5, t=1, q=4 for intra-mode frame

Sequence 9 : ADTV, p=5, t=3, q=4 for intra-mode frame

Fig. 1 Coding Rate using ADTV Technique

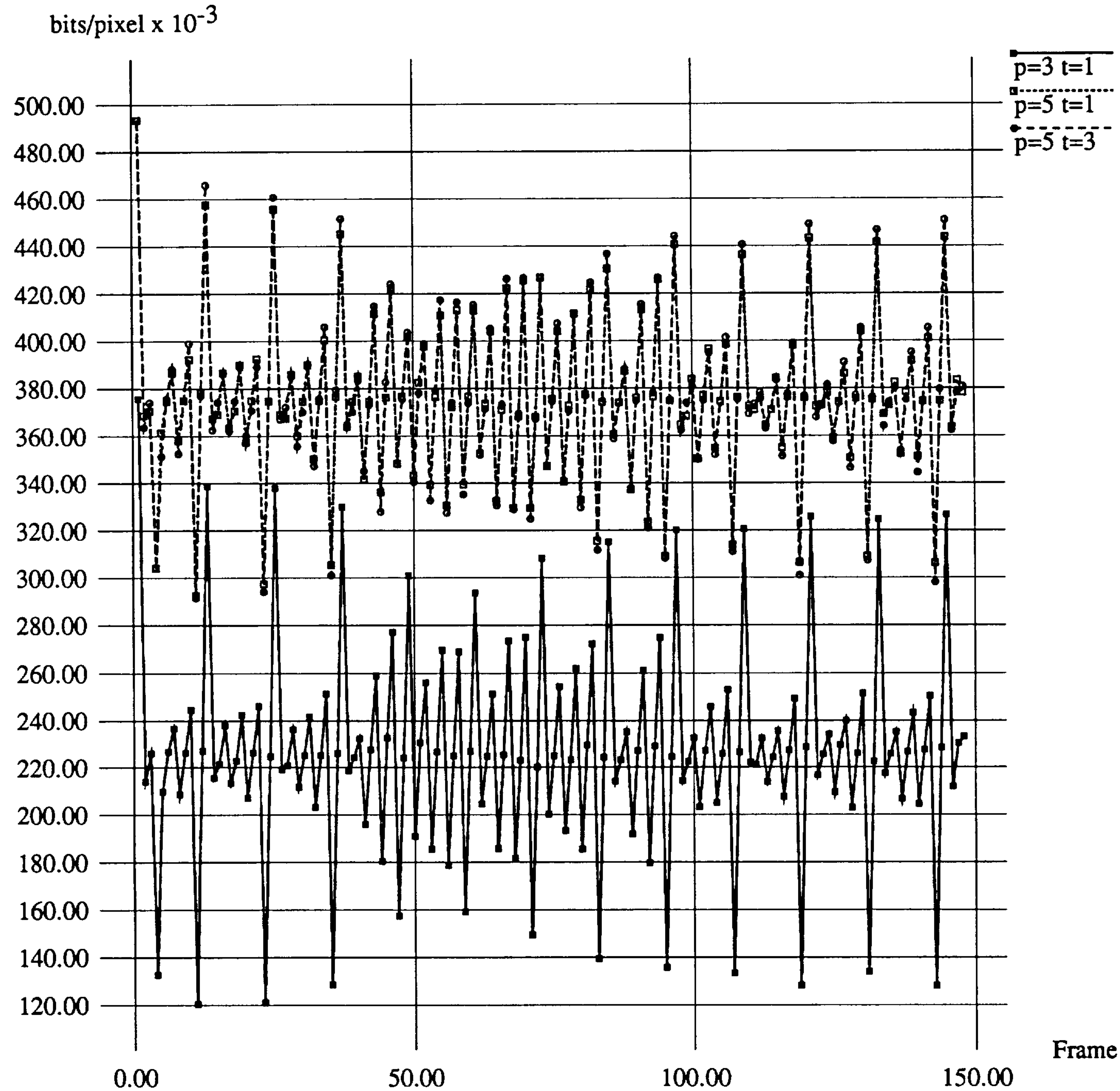


Fig. 2 PSNR using ADTV Technique

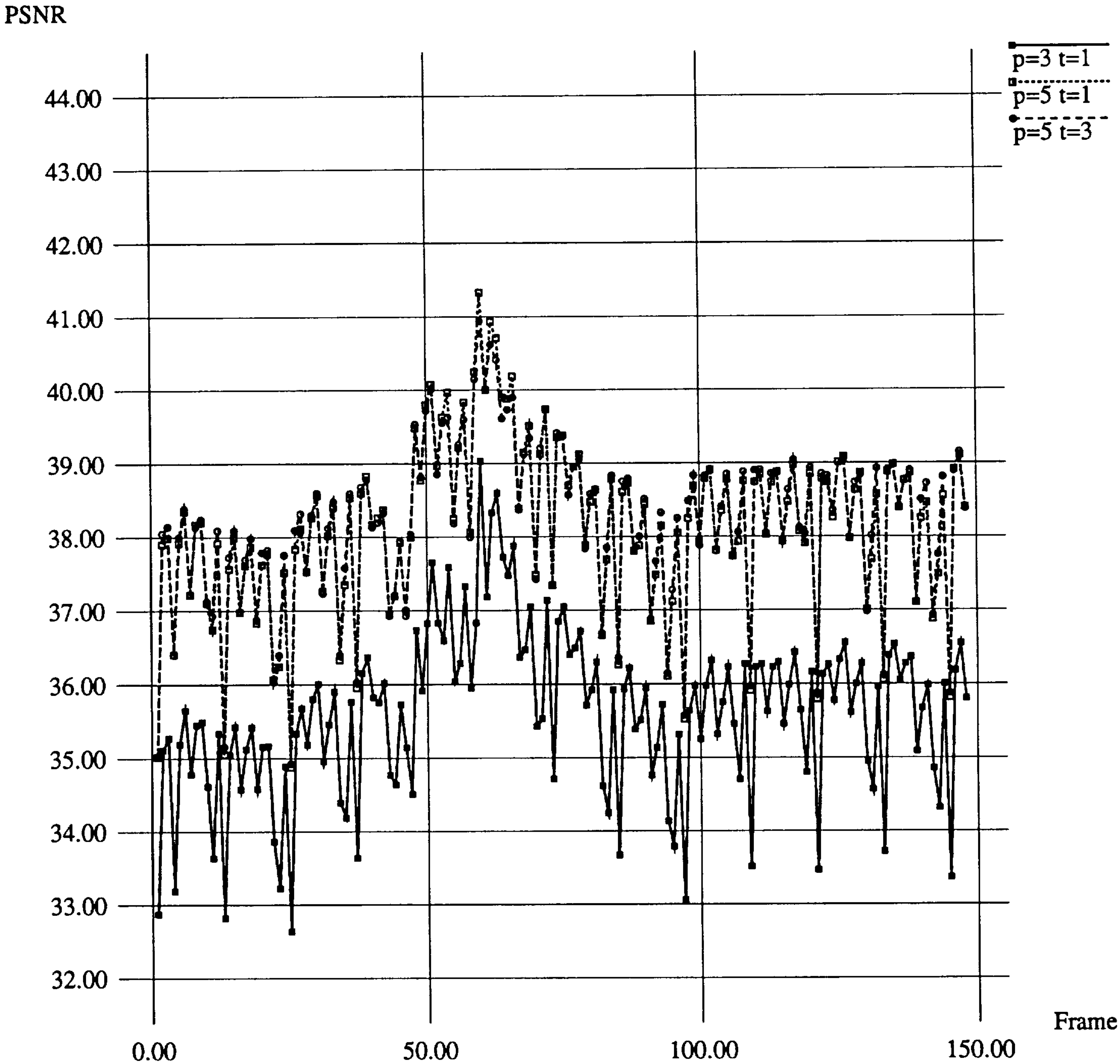


Fig. 3 Coding Rate using H.261

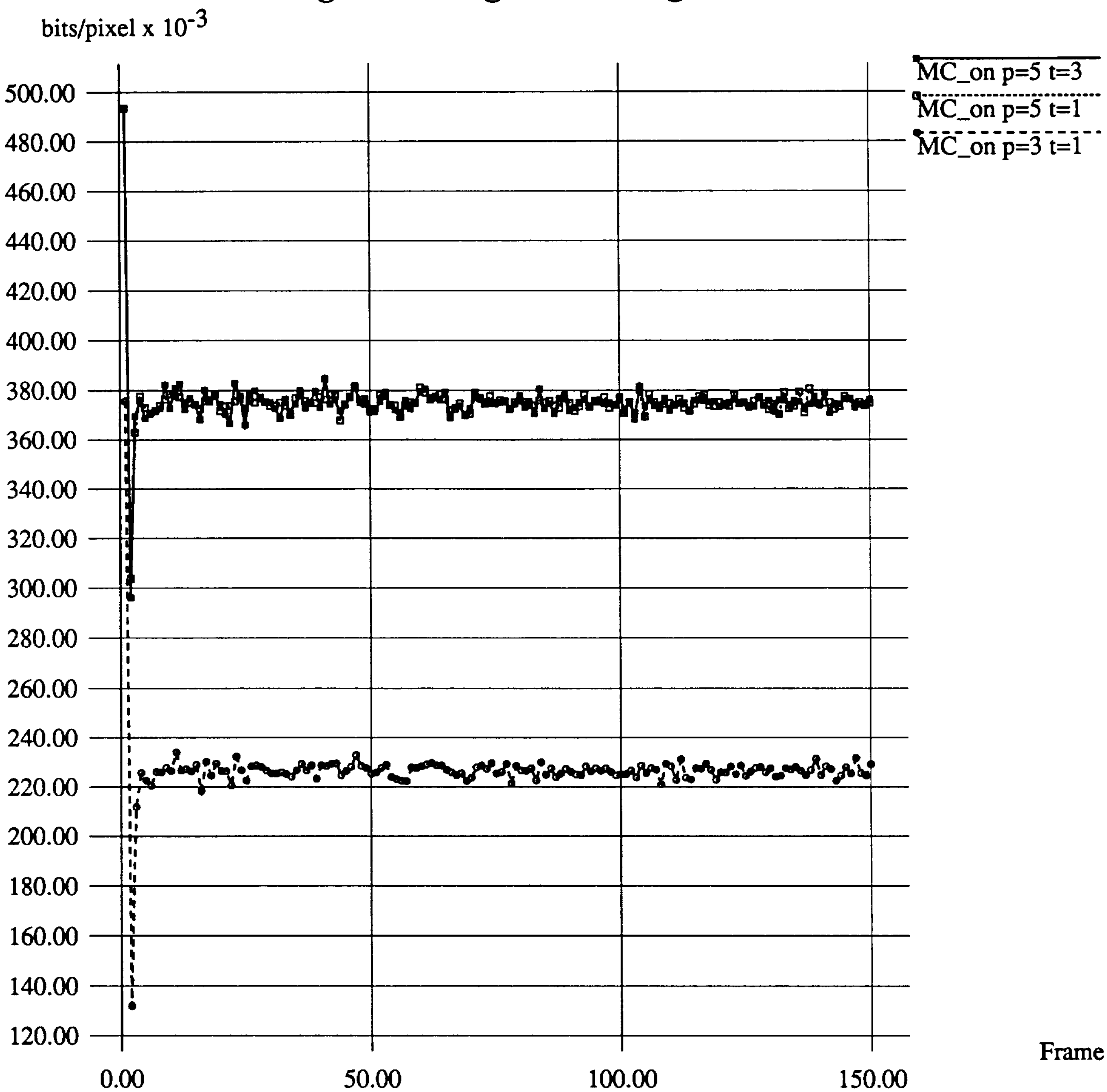


Fig. 4 PSNR using H.261

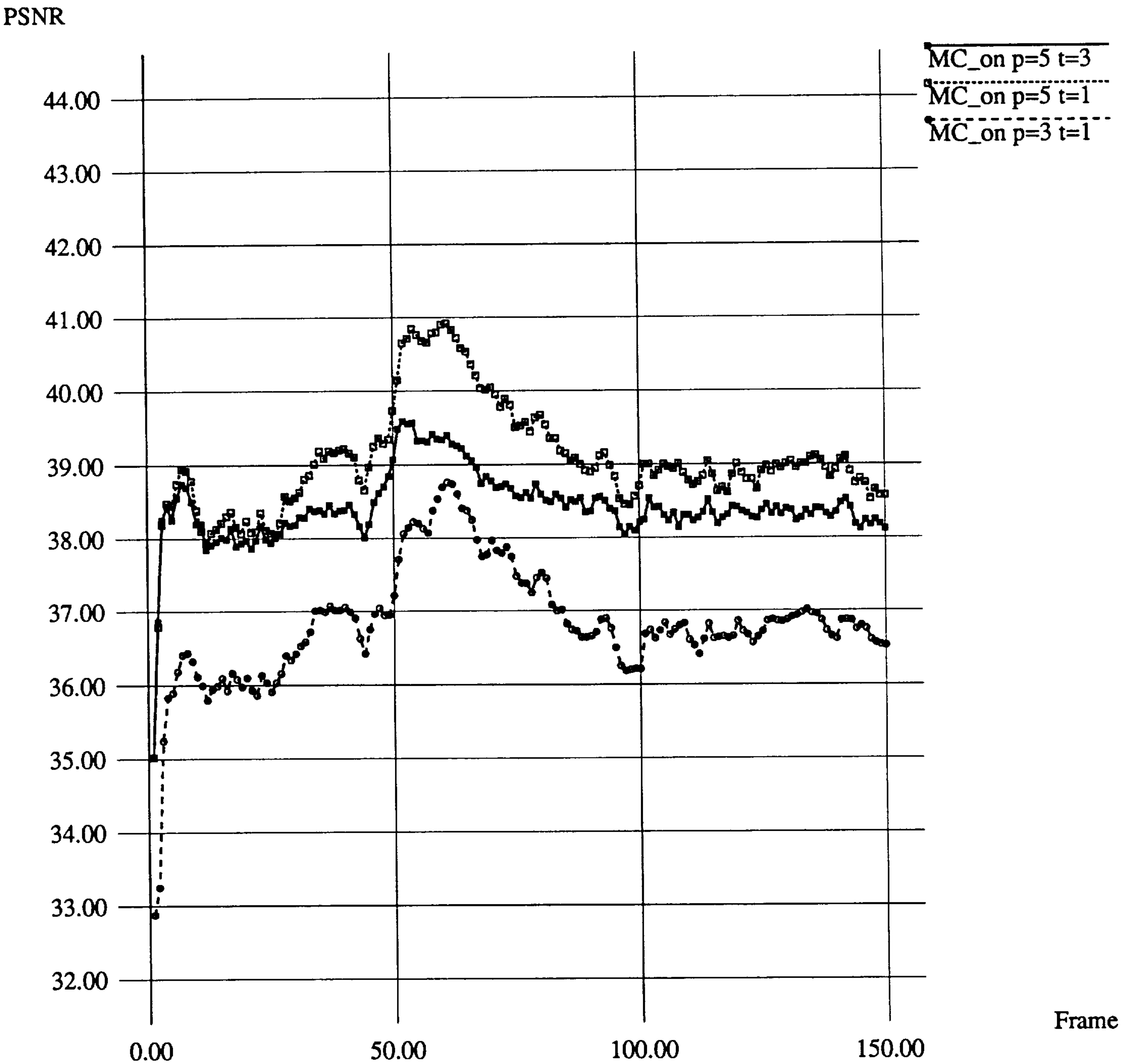


Fig. 5 Coding Rate using Modified ADTV Technique

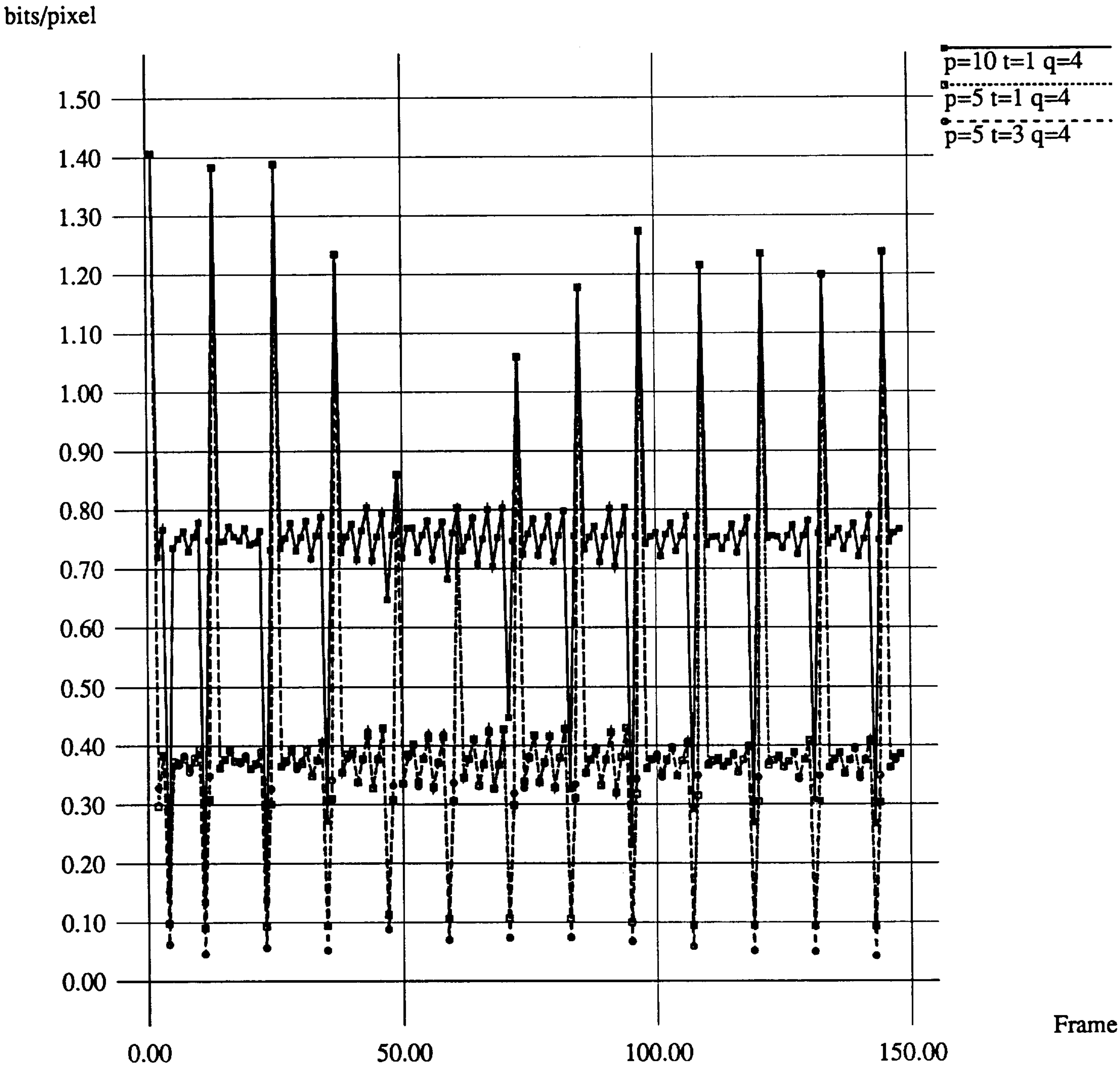


Fig. 6 PSNR using Modified ADTV Technique

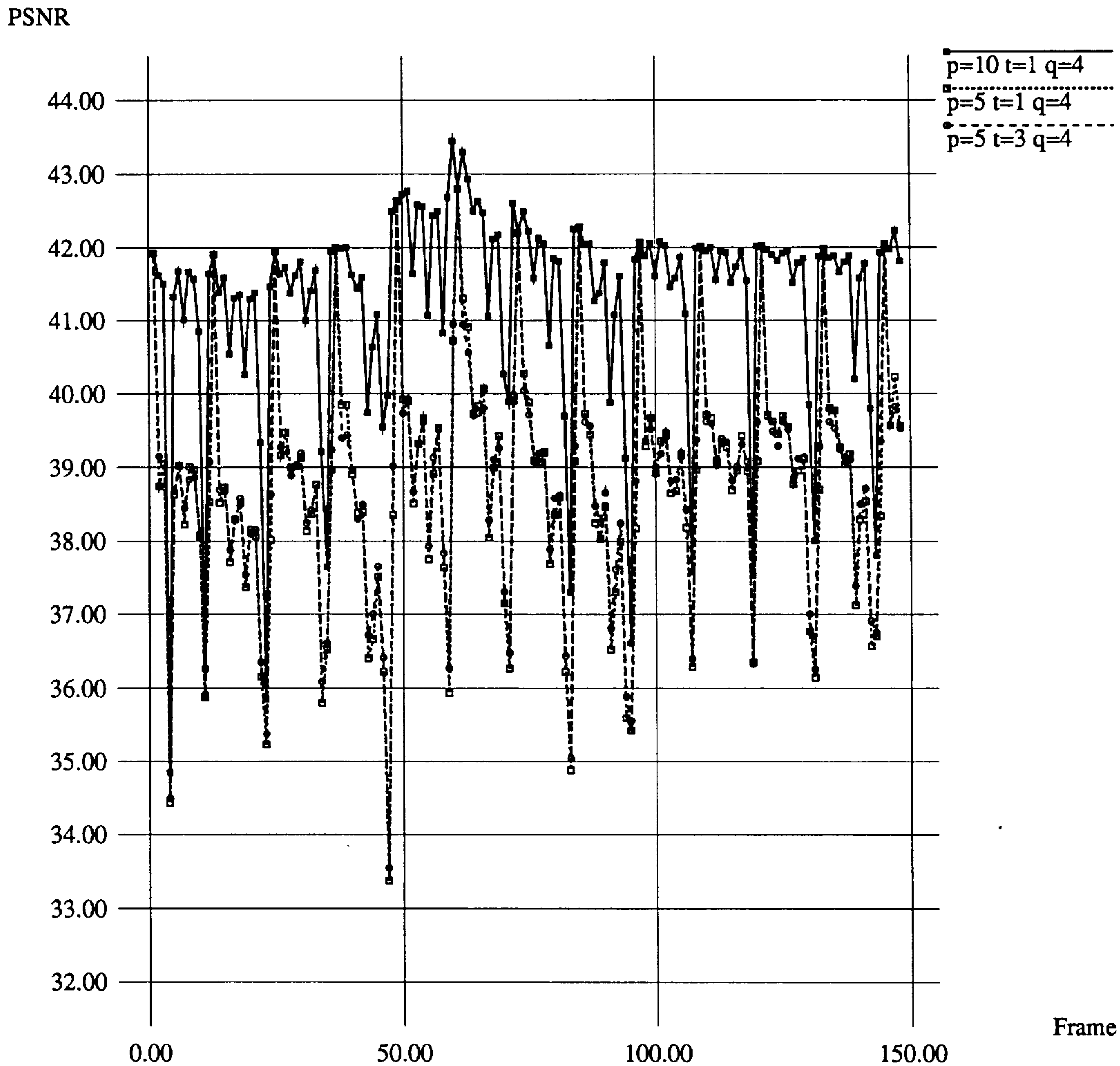


Fig. 7 Comparison of Coding Rate (p=5 t=1)

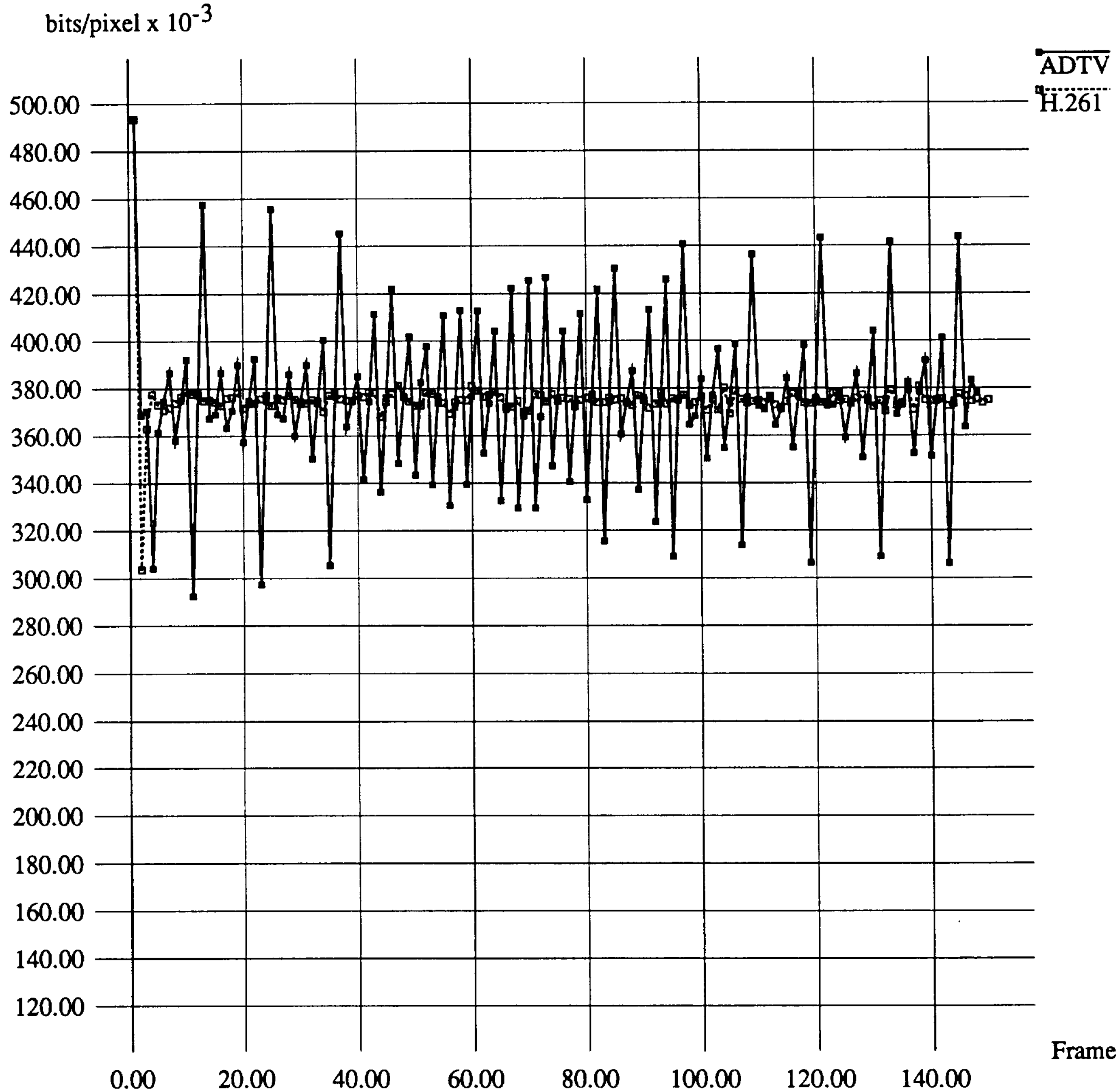


Fig. 8 Comparison of PSNR (p=5 t=1)

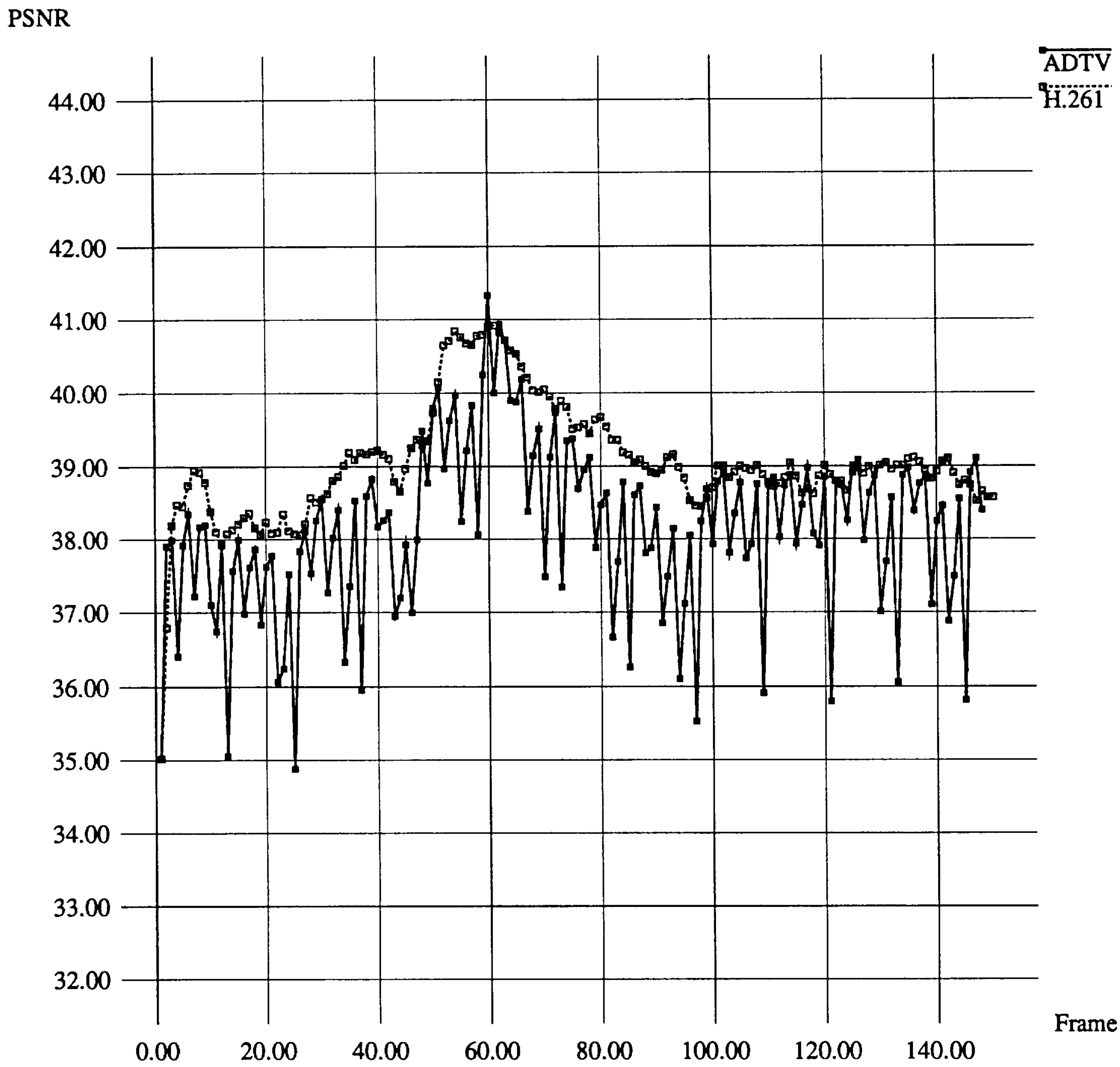


Fig. 9 Comparison of Coding Rate (p=5 t=3)

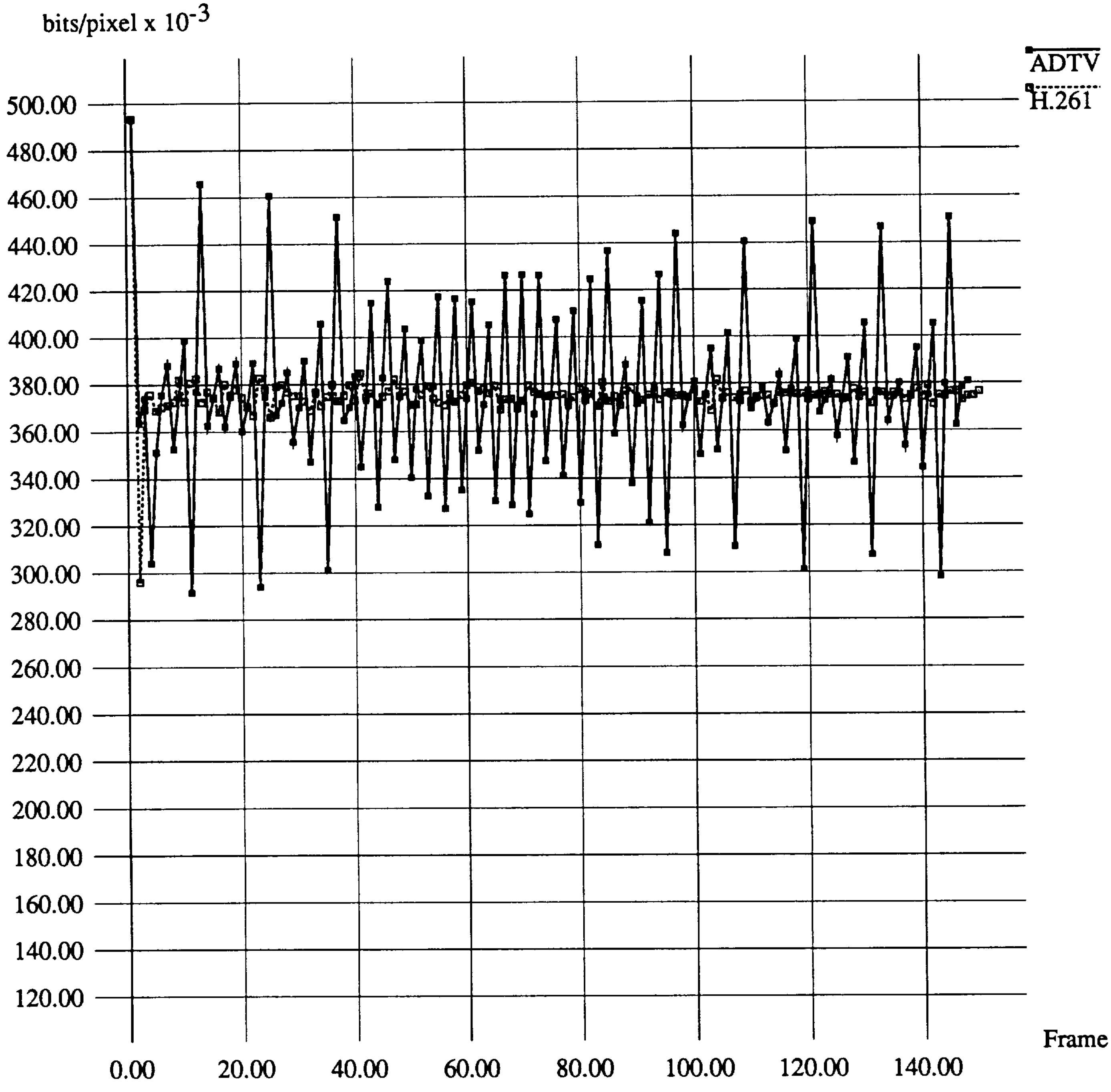


Fig. 10 Comparison of PSNR (p=5 t=3)

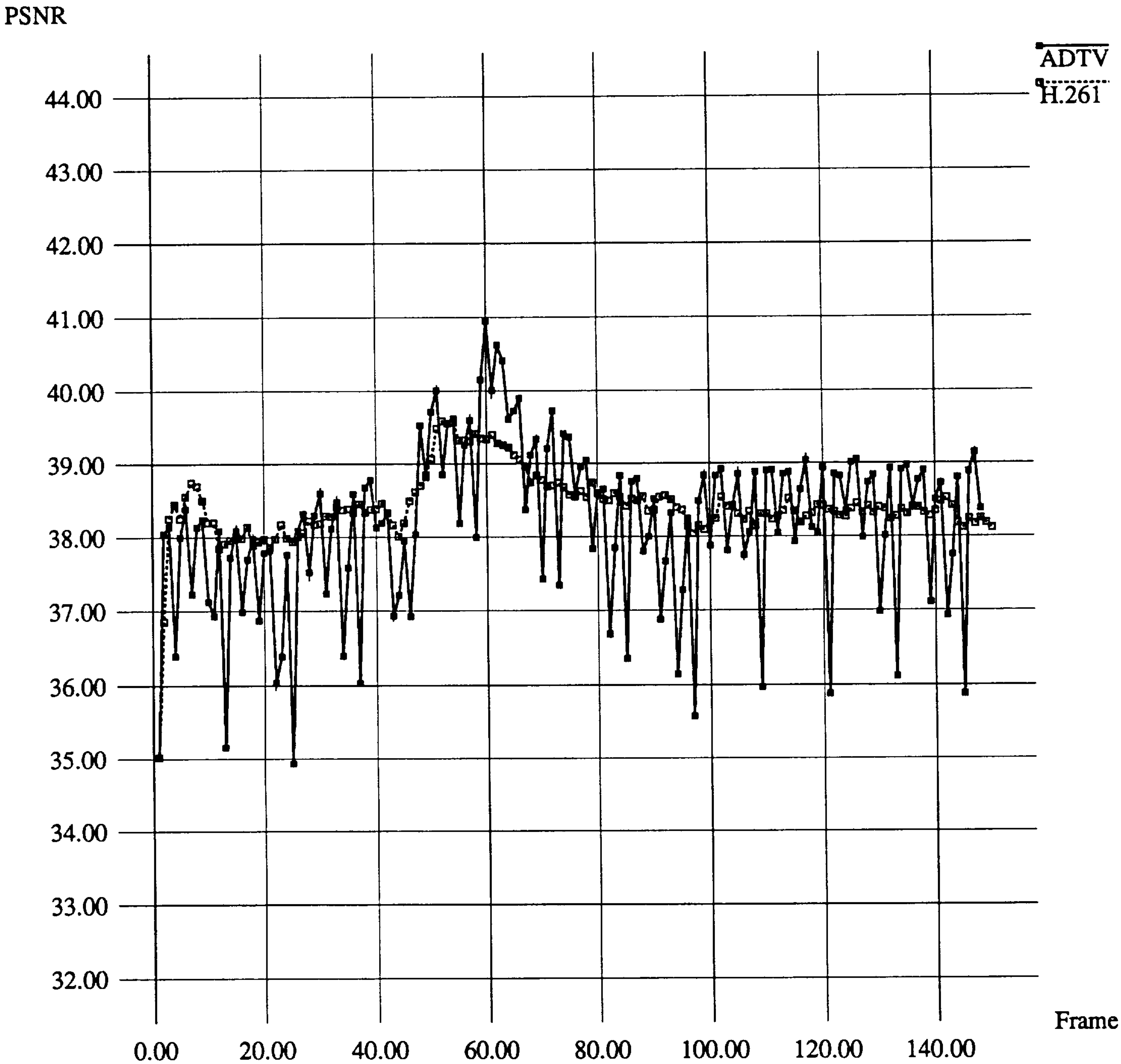


Fig. 11 Comparison of Coding Rate (p=5 t=1)

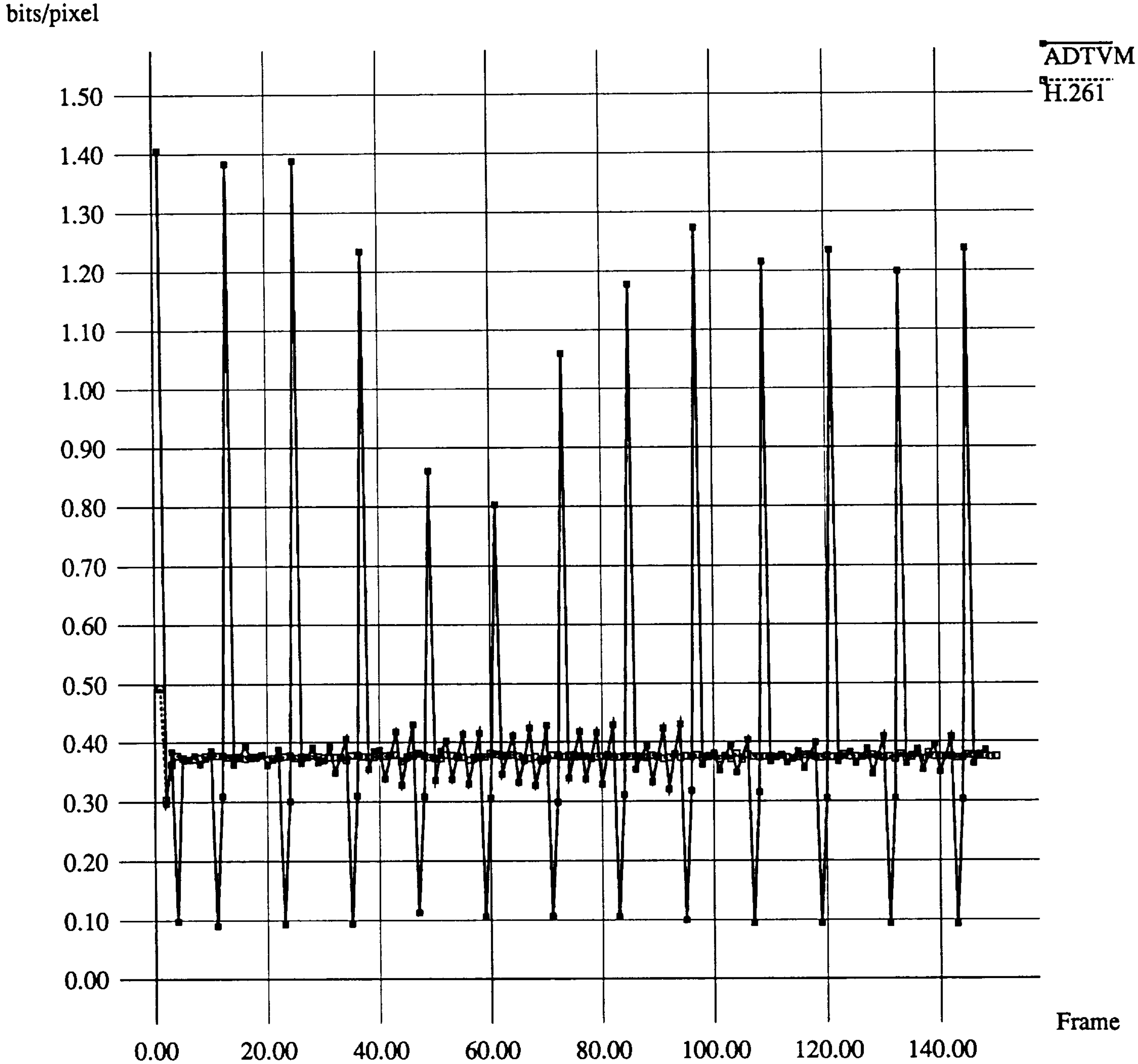


Fig. 12 Comparison of PSNR (p=5 t=1)

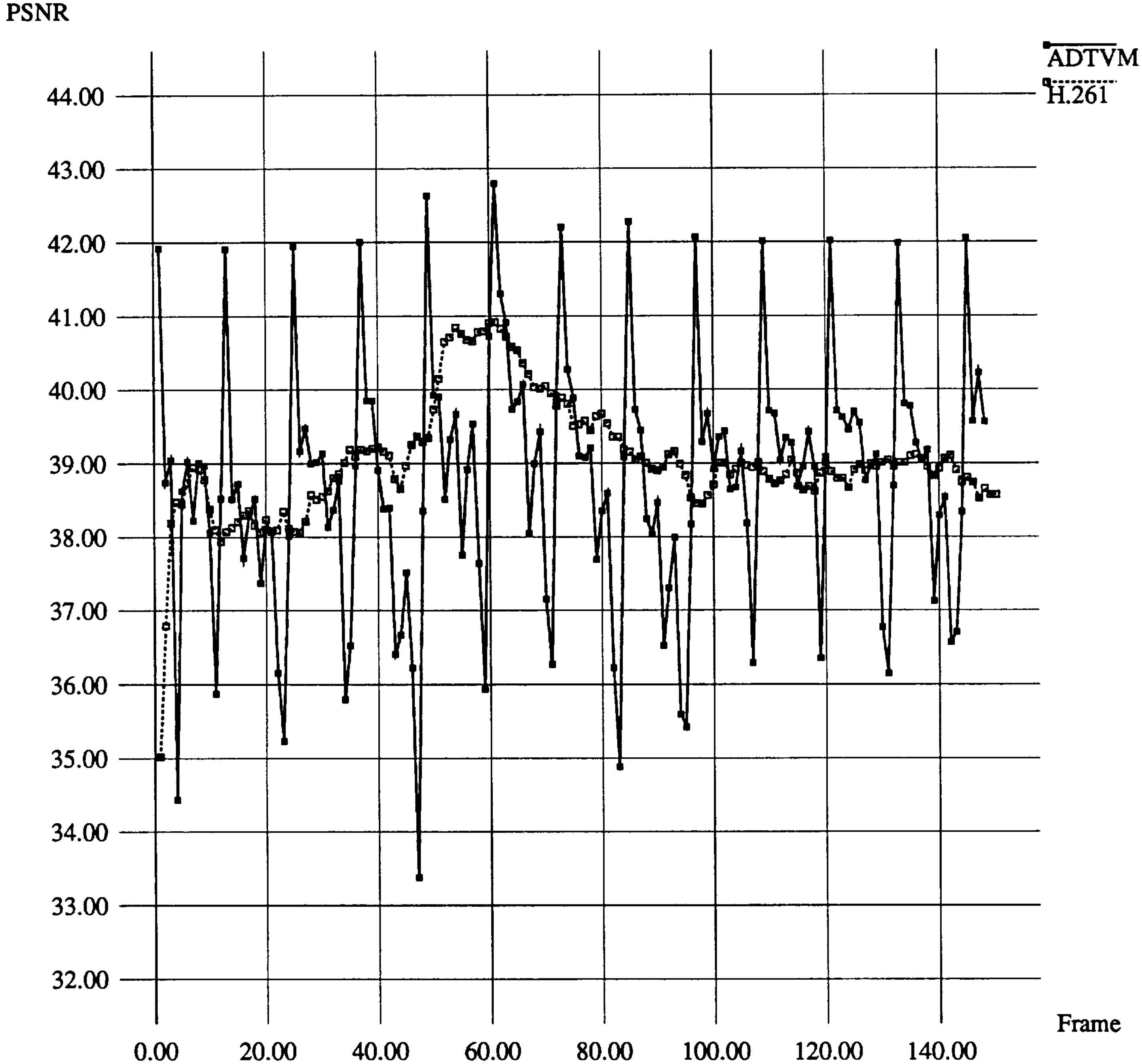


Fig. 13 Comparison of Coding Rate (p=5 t=3)

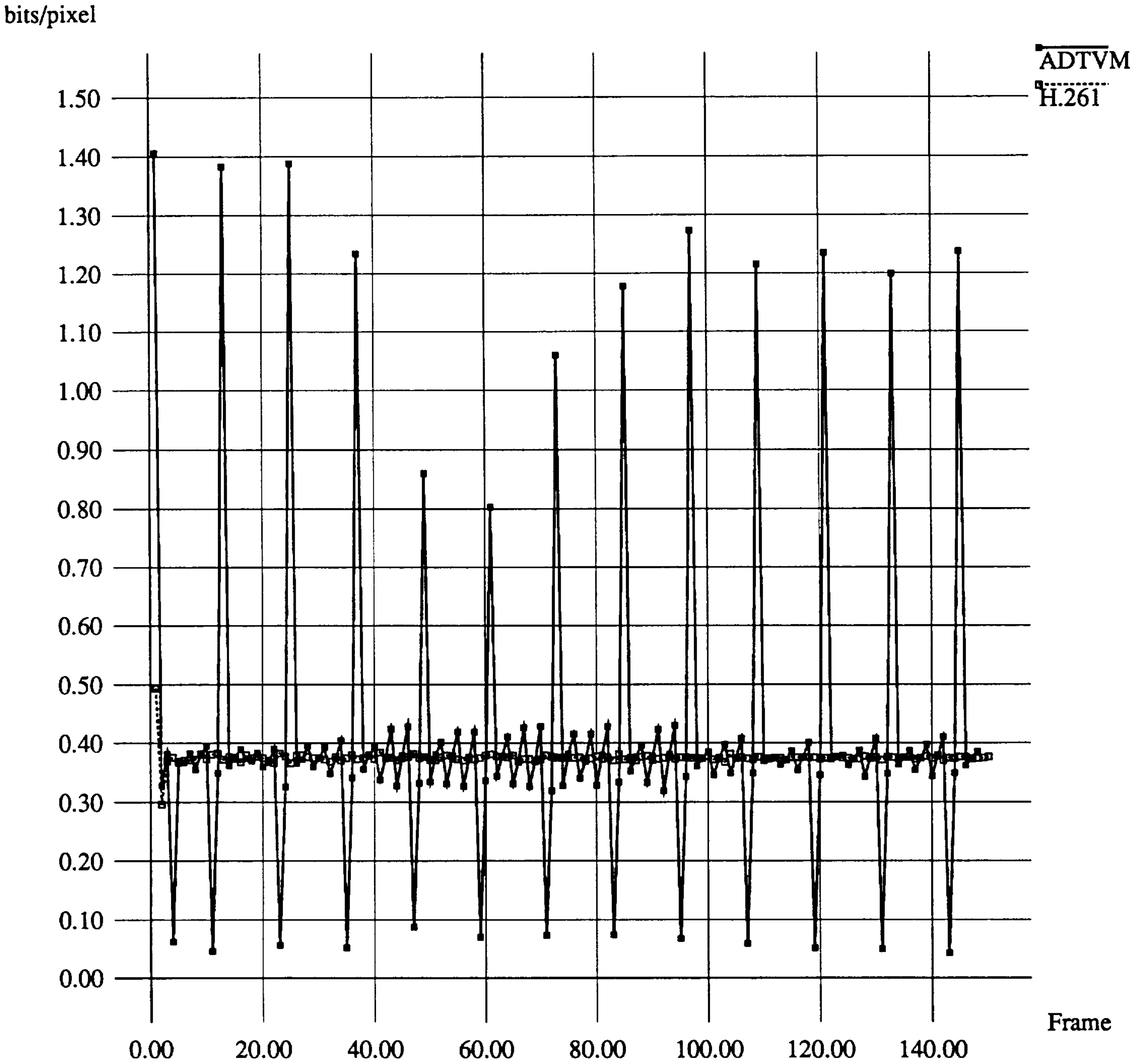
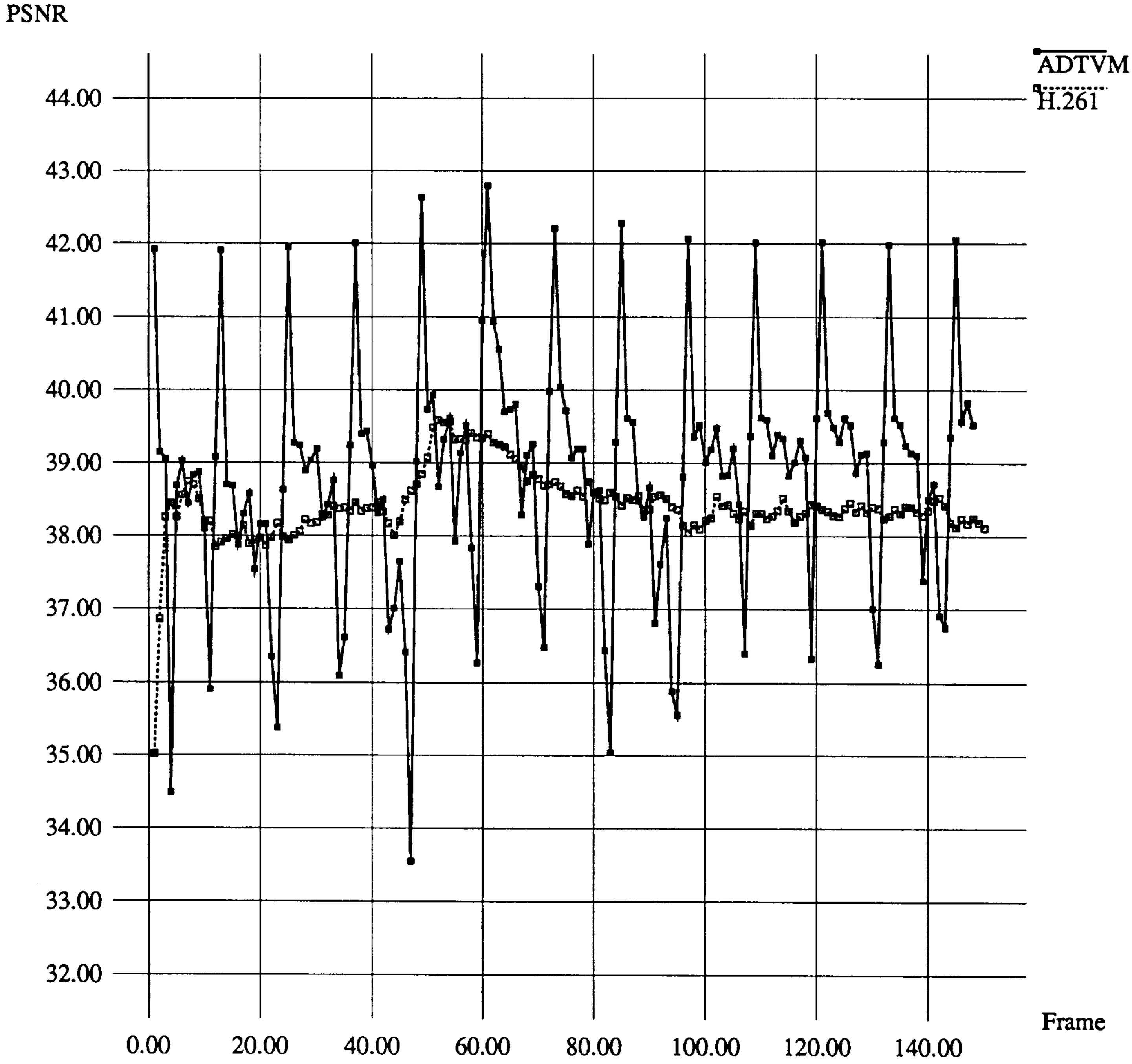


Fig. 14 Comparison of PSNR (p=5 t=3)



APPENDIX