# Study on Significant Technology of H.264 in Mobile Network<sup>\*</sup>

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Abstract: - The characteristics of the mobile wireless network are limited bandwidth and high error rate, at same time the computing and storage capacity of the mobile devices are limited too. In order to ensure effective video compression and transmission, H.264/AVC standard provides the significant technologies and strategies in error resilience strategies, rate control, mode decision and scalable extension. On the basis of the detailed study in the strategies, a combination method is proposed, and experiment proves the quality of video transmission is improved at the same time the rate is reduced.

*Key-Words:* - Video transmission, H.264, Mobile network, Wireless transmission, Rate control, Error resilience, Mode decision, Scalable extension

# **1** Introduction

The computer and communication technologies have been a rapid development in the second half of the twentieth century. These scientific and technological innovations have brought tremendous changes to human life and work. The 4A (Anywhere, Anytime, Anyone, Any device) concept of access to information request is further raised in the beginning of the twenty-first century; the core is to achieve zero distance and barrier-free exchange of information and access to information.

The wireless communication links are essential for achieving the information access of 4A level. On the other hand, the 70% of human access to the information comes from the visual. The amount of the information achieved by the visual is much lager than those of the auditory stactile and other methods, so the video has become the most important information carriers. Therefore, the mobile video communication technology based on the wireless channel has become one of the hottest research topics nowadays. However, the use of wireless transmission of video signals are faced with many challenges, these are mainly in [1-4]: The contradiction between the huge video data and the limited bandwidth of wireless transmission; the contradiction between the high error rate of wireless channel and the sensitivity of lost data of the compressed video flow; the contradiction between the bandwidth fluctuation of wireless channel and the relatively constant bit-rate of compressed video.

It is necessary to find a more perfect video compression codec standard in order to solve these problems. Over the last decade, the video compression technology has gradually become a hot research, and the video codec technology makes many breakthroughs. Through the use of advanced video compression technology, the video quality is enhanced at the same bit rate. H.264/AVC is the new international video codec standard developed by VCEG and MPEG. The milestones in video coding are shown in Fig.1 [5].

As the next-generation video coding standard, compared to the previous coding standard, H.264/AVC codec performance has a great increase. Higher codec efficiency and stronger ability to adapt

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to network and to error resilience make it more suitable for wireless transmission. As wireless communications technology development, especially the emergence of the 3G communication system makes the video communication gradually becoming the major service and important part of the mobile telecommunications, and the services are more diversified, such as: wireless video broadcasting, wireless video dialogue, wireless VOD and wireless video surveillance etc.



Fig.1 Milestones in Video Coding

# **2** Problem Formulation

Bit error problem: An efficient compression technology must be adopted for video data because of the huge amount data and the limited wireless bandwidth. Because H.264/AVC standards have the high encoding efficiency and the good network adapter, it will become the 3G wireless video communications standards. There are still some difficult issues after high-compressed stream in the internet, especially the wireless transmission channel. But just because of the high compression efficiency, it makes H.264/AVC stream more sensitive to bit error, even a single primary error also may result in the sharp decline in video quality after reconstruction. The sequence of video compression makes a packet loss will cause serious transmission errors, the collapse will lead to the decoder when seriously.

Even if the packet error rate is very low in the receiving end, the decoded video quality has still

been severely impacted. On the other hand, there are a large number of random error and burst error which are introduced by multi-path reflection and decline of wireless channel, so that the normal transmission of bit flow is affected. In particular after the use of the variable length code, the stream is more vulnerable to the impact of the error, leading to the loss of synchronization in the decoding end. So the code flow cannot be correctly decoded before the synchronous code arrives. At the same time, the prediction code technology will spread the error to the entire video sequence, greatly reducing the quality of the image reconstruction.

As wireless channel has high error rate, it is necessary to use all kinds of error resilience strategies to ensure the video transmission reconstruction of image quality in the wireless environment. Therefore, in order to achieve a good quality of video transmission, the error resilience and hidden technology research has become a key problem needing to resolve in the mobile communication.

Rate problem: In order to make the everchanging video bit stream fit to the relatively fixed bandwidth requirements, so rate control is an essential and indispensable part in different video compression standards.

Rate control technology is one of the key factors which impact the video encoder coding efficiency. Its main task is to effectively control the video encoder to make the size of the output stream meet the actual transmission channel bandwidth limitations, and, as far as possible, to obtain the optimal decoding image. At present the famous algorithms mainly include MPEG-2 TM-5, MPEG-4 VM8 and H. 263 TMN-8 etc. In general, we can divide the main functions of rate control algorithm into two steps; the first step is the so called "bit allocation", that is, the limited bandwidth is assigned to the image unit of group of pictures (GOP), frame as well as macro block and so on. The second step is to calculate quantitative parameter (QP) in order to make the actual bit rate and object bit rate basically same.

The design of rate control is to achieve the optimal allocation bit as well as the precise control of output bit rate; the key is to establish a precise mathematical model of rate-distortion [6].

Traditional rate control method is to use of the buffer to adjust the video output bit stream, use of the time delay to make transmission channels adapt to the natural variable rate of compressed video bit stream. Now commonly used by the traditional rate control algorithms are VBR and CBR. The classical application of CBR rate control algorithm is in the high demand on transmission delay such as videophone etc, constant bit-stream is produced after video compression, and the two terminal need not provide a large buffer in the application. VBR is applied in time-varying channels. Giving full consideration to the instability of the transmission channels, a better protection to the channel bandwidth change is provided during the rate control. As the VBR rate control method adapts to a wide range, it also applies to the CBR channel. The traditional VBR rate control algorithm can bring the enough insurance to the video quality, but bring the unpredictable delay of the transfer bit stream. It also brings the video, audio synchronization difficult

Different form the traditional stable cable channel, wireless channel has a much higher bit error rate, and the bit error rate will soar by the increase of transmission packet [7]. So the packet size is restricted in the wireless channel. In general, the Maximum Transfer Unit (MTU) is about 1500 bytes in size in cable channel, and not more than 254 bytes in wireless channel [8]. Therefore, in the same channel bandwidth conditions, the wireless cable channel have channel and different transmission characteristics, wireless channel of transmission is characterized by short transmission time interval (TTI) plus small transfer protocol data unit (PDU), and cable channel of transmission is characterized by long TTI and big PDU. Because the more PDU are transferred in channel, the more they lose, it is especially important for wireless channel to control the number of PDU. This problem is not very obvious in cable channel because of the bigger IP packet, so the rate control strategies are different in the two kinds of channels [9].

Complexity problem: One key reason why H.264/AVC can get a great increase in encoding performance is that a variety of different coding models in motion estimation (ME) and models decision are introduced. However, the high computational complexity which is introduced for improving the compression efficiency is untenable to the real-time video applications. Compared with MPEG-4 simple configuration, the decoding complexity of H.264/AVC main configuration increases more than 3 times, and the coding complexity increases 10 times. Compared with the past two modes (16\*16 and 8\*8) which are generally used in the standards, H.264/AVC supports as many as 7 inter modes including 16\*16、16\*8、8\*16、8\*8、8\*4、4\*8 and 4\*4. And H.264/AVC introduces the intra mode; there are 9 prediction modes for 4\*4 luminance block, 4 prediction modes for 16\*16 luminance block, 4 prediction modes for 8\*8 chroma block.

But with the increase of mode choices, the strategy of mode selection becomes complex and the algorithm complexity of motion estimation has multiplied. The currently adopted encoding mode of H.264/AVC is that encoder must check up the encoding mode in every macro block so as to determine the optimal mode. The full search motion estimation applying the all models is estimated to have more than 80% of the whole calculation. This whole search method brings a heavy burden of calculation to encoder, and makes it difficult to attain real-time communication requirements [10].

Scalability problem: Also, the wireless network environment, which is made of various types of wireless network, is heterogeneous, such as 3G/GPRS/CDMA) 、 WL AN、 WPAN 、 wireless Ad Hoc network etc. And, the end-users of wireless access 、 processing capacity will vary, laptops, PDA, smart phones and other wireless terminals link with the entire wireless network through a variety of wireless access network. How to fully utilize the bandwidth and resources of the existing wireless network, provide the different video services based on the different demand and process capacity, realize the service flexibility and ensure the service quality will become an import issue [11].

For the problems mentioned above, there are all kinds of new research techniques and results to solve one side in a certain degree. However, it is few to combine all these improved methods for an experiment. In this paper, an innovation is that we choose and combine the advanced methods to test and analyze by 3G offline simulation environment and JVT reference software, and it provides reference for evaluation of video service.

In the following description, the structure of H.264/AVC video streaming transmission in the wireless network will be introduced first, then the key technology, the related discussion and solution will be presented, and the experiment is tested and analyzed, finally, we give a conclusion and refer the further research work.

# **3** Problem Solution

# 3.1 Structure of H.264/AVC Video Streaming Transmission

As shown in Fig. 2, H.264/AVC can be divided into two different semantic levels: VCL and NAL. VCL

and NAL are both included in the H.264/AVC standards, but their roles and division of labor are different. VCL is responsible for the effective video signal compression, and NAL gives a definition of the interface between video compression information and the outside. NAL is responsible for packaging each slice of VCL into a NAL unit. In the process of compression data synchronization and error identification are processed, and so on. After NAL decoder receives NAL units, it will check data synchronization and error identification to ensure the correct decoding.



Fig.2 H.264/AVC Semantic Hierarchical Structure

NAL unit is made of the unit head and the payload string, the size of the unit head is a byte, and the size of the payload string is decided by the size of output slice. According to the different slicing strategies of H.264/AVC encoder, the size of the payload string may be very different in the one video streaming. As shown in Fig.3, it is the head structure of NAL unit, there are 8 bits. And the first bit F is inhibition bit, normally the value is 0. When F is 1, it presents a syntax error, NAL unit cannot be decoded. NRI is 2 bits prediction bits, 00 presents the content of NAL unit is not used for inter prediction, vise verse. There are 5 bits in Type, its value is between 1 and 31 which decides the type of NAL unit.

0	1	2	3	4	5	6	7		
F	NRI		Туре						

Fig.3 NAL Unit Head Structure

In order to adapt different types of transmission systems (such as MPEG-2, H.324, wireless network etc.), NAL unit must be re-packaged in accordance with different transport protocols in the transport layer. As shown in Fig.2, in the circuit-switched wireless network business, NAL unit usually is packaged and delivered by the form of H.324 protocol which is defined in the Appendix B of H.264/AVC standard document. But when packets switching, NAL units usually are packaged and transferred in the form of RTP package. As the network of "All IP" trend, most multimedia application will apply in the form of packets switching in the next 3G/4G wireless network. In addition, the delay and loss of media transmitting in the wireless channel of circuit-switched and packetswitched are similar.

The data of H.264/AVC VCL are packaged RTP protocol packets and then transfer in IP network to CDMA2000 system as an example for real-time packet data service shown as Fig.4., for packet-switched business, IP protocol stack are similar, it is noteworthy that in order to reduce the relay of retransmission, UDP protocol is adopted. In the MAC layer and physical layer below the link layer, and the specific system is different, so here is simplified.



Fig.4 NAL Unit Package in CDMA2000 System

NAL unit first is packaged into RTP/UDP/IP packet as shown in Fig.4. According to robust header compression protocol ROHC, RTP/UDP/IP head is compressed, and PPP protocol head is added, become a RLP-SDU (Service Data Unit) unit in the radio link protocol layer. RLP-SDU is cut into RLP frame suitable for link transmission in the RLP layer. The same NAL unit possibly is cut into different frames. RLP frame added RLP protocol head forms the logical transmission unit LTU.

For multimedia broadcast/multicast service MBMS in 3G, the process of NAL package is a little different, because the ROHC is not used and the FEC method is adopted to improve the reliability of data transmission.

#### **3.2 Error Resilience Strategies**

Error on the quality of video decoding can be divided into the following three categories:

The first error is a single error of the video parameters, and it does not affect any of the video data except for the parameters of its own injury. Under such a circumstance error is limited in the single macro block which does not participate in any further prediction. When errors occur in DC or AC coefficients of intra macro block which are not used for motion prediction, it is the case. Due to the injury macro block is not applied to any follow-up prediction, the error loss is limited in the affected macro block. The decoder has skipped the right bits as it begins to read the next parameter. Because the synchronization will not lose, the damage of such errors is the lightest for the video quality after decoding.

The second category will cause more problems because of the time cumulative injury produced by prediction, as motion vector is encoded by the prediction difference, the errors in motion vector code will be transmitted to the end. And because the process of motion compensation introduces the time relationship, the errors will spread to the follow-up inter coding frames. Choosing the actual motion vectors encoding rather than the prediction residual part can reduce the effect [12], but obviously that will add the bit rate of video streaming. And because intra prediction is largely adopted, it will cause the same problem.

The third effect losing the synchronization is worst. At this point decoder can no longer judge which part of the frame the received information belongs to, such errors are caused by the change of bit rate. When decoder detects the errors in VLC, it will skip all the coming code bits. Whether it is right or not, it will search another correct synchronous code to recover the synchronous state. So a single bit of damage will become a series of channel errors.

There are two reasons on errors in wireless network, one is the bits inversion, the other is that RTP packets are discarded by the gateway or router. The latter data cannot recover, but the former data can recover by means of FEC etc in the applications which need not a high real time. Considering the delay issue in the real-time application, RTP packages are discarded directly [7]. In this way, the main errors are prediction loss caused by RTP packages loss and the problem of synchronization loss.

Judging from the current view of the latest study, the use of forward error correction (FEC) measures such as convolutional code or Turbo code can effectively reduce the bit error rate in wireless channel transmission. But too complex channel coding methods increase the complexity and cost of user terminals.3GPP Standardization Organization recently selects Digital Fountain's FEC technology as the mandatory component of 3G multimedia broadcasting and multicast transmission business MBMS.

Reference [13] refers a video error resilience transmission strategy in combination with the typical characteristics of 3G wireless channel and based on data division redundancy package and FMO. The method has some self-adaptable, and can adjust the length of redundancy package to maximize the saving rate and protect the important data when the channel error rate changes. Reference [14] refers a method based on the application of data partitioning and unequal error protection. The influence factor of inter MB and  $\gamma$  curve are introduced. According to their effects to error propagation, the C type partition is further divided into some subtypes, and then unequal error protection is given to all partition switches. The model of rate allocation and an iteratively improved bidirectional local search algorithm are also presented. Reference [15] puts forward a RTP packetization algorithm for H. 264/AVC video which is called Hybrid Mode Packetization (HMP). HMP considers video stream content correlation and unequal protection of important information with H. 264/AVC NAL video features. HMP can achieve good perceptual picture quality under packet-loss network conditions.

Reference [16] refers two improved methods: the first method changes the burst error in the network into statistically independent error, using convolution and interlaced technique on the packet encode; the second method divides video data into important data and unimportant data and checks the error in important data at receiving terminal, to control accordingly the retransmission mechanism

at transmit side in the wireless transmission system. Transmit terminal will be asked to retransmit only when the number of lost packets is larger than a threshold. Reference [17] first presents that mobile terminals support only part of error resilience tools of H. 264 /AVC because of the hardware restrictions. Then presents various error resilience tools and their usability in wireless environment, at the same time, some simple error resilience schemes for packet switched conversational services are proposed. Finally experiments data show that encoder with low complexity FMO method and appropriate intra block refreshing can get better error correction ability, because this method can make full use of spatial correlation between frames and reduce the requirement of bit rate. Reference [9] refers taking full advantage of the results of feedback coming from the reverse channel when bidirectional video transmission. Encoder can accordingly change the reference frames based on the result of feedback under the circumstances of multi-frame reference in the coder end so as to enhance the ability of error resilience.

#### 3.3 Rate Control

Different from MPEG-2, MPEG-4 and H.263 encoding system, the rate control algorithm of H.264/AVC standards is much more complex. This is because the quantitative parameter QP attained from rate control not only processes rate control, but also uses as the basis of selecting macro block code mode in rate distortion optimization model (RDO). When the macro block is processing the RDO optimization, the MAD (Mean Absolute Difference) of the macro block or the image is always needed to ensure the quantization parameter. But the MAD value only can be ensured after RDO is optimized and coding mode is decided. In order to resolve this contradiction that can not be avoided, H.264/AVC reference software adopts a number of methods based on the prediction to ensure the bits that each frame will take up and the MAD value, thus the quantitative parameter QP can be ensured.

The rate control algorithm of H.264/AVC has used HRD buffer reference model , fluid flow model , MAD prediction model , second ratedistortion model theories etc, these are important to the prediction of bit rate parameters in each frame.

JVT reference software provides different levels of rate control in GOP level, frame level and basic unit level, one basic unit can be number of macro blocks of your own choice. In order to facilitate the narrative here, we only consider each frame only has one basic unit.

EBR rate control algorithm aims at the wireless channel in particular [18, 19]. And EBR is the abbreviation of Explicit Bit Rate, that is, the bit number after encoding is made certain in advance by encoder according to the rated channel bandwidth, the output of encoder can not exceed the scope so as to meet the wireless network transmission need better. Compared with VBR bit control algorithm, the control mechanism of EBR is stricter with the output requirement of encoder. EBR rate control algorithm is designed specifically for the wireless channel, particularly in the 3G wireless channel. Because of the limitation of transmission frame size and delay in wireless channel, EBR rate control algorithm has the following advantages:

Lower end-to-end delay: as EBR adopts the idea of Source adapting to channel, each frame must complete transfer in the given time, thus, not only the buffer size of decoder can effectively be reduced, but also the end-to-end delay can be reduced too.

Improve the system's error resilience: compared with VBR methods, EBR strictly controls the bit number of each encoding frame, minimizing the number of PDU in a given bandwidth so as to minimize the lost number. It can effectively improve error resilience in the wireless channel environment.

Improve compression efficiency: as EBR can minimize the number of PDU and RTP package in the transmission, the additional expenses in the unit head reduce to the least too. So the compression efficiency improves.

In the EBR rate control algorithm, the compressed video frame is limit to the fix number of slices, the size of each slice is smaller than the max PUD unit in order to transmit the video frame in time. Because of the characteristic of EBR method, the needed buffer size in the receiving end can reduce to the minimum. EBR has the special requirement for the decoder and transmission channel. This is as follows: the synchronization of video frame. The video frame must be transmitted to the receiving end in the designated period of time as the receiving end doesn't supply the extra buffer. Another is the requirement of the PDU size. The wireless channel need meet different bit rate data stream to suit for different payload PDU unit as the coder outputs a series of different length bit rate.

The key to EBR rate control algorithm is the realization of slicing. JVT reference software

provides two methods: one is fixed macro blocks, the number of macro block in each slice is certain, as the number is specified by the user in encoding, the number of slice is certain after each frame is compressed. The other method is to fix the bytes of each slice, if the cumulative number of bytes when encoding exceeds a given value, a new slice is created. The two kinds of slicing methods has been widely used in the H.264/AVC encoder. The advantage of the former lies in well controlling the amount of the slice in the whole video frame, reducing the extra overhead of NAL unit head applicable to the low error rate of the cable network.

The advantage of the latter is able to control the size of the NAL unit. If the NAL unit is appropriately reduced in coding, it can decrease the loss which is brought by loss of NAL units. This is an effective method to improve the capability of error resilience in the wireless network. EBR algorithm requires combining the two algorithms mentioned above, both the amount of the slices in each image is fixed and the size of each compressed NAL unit is fixed. It is obvious that all present reference software algorithms are not competent.

Reference [9] first refers that the simplest method of EBR slicing is considered below: the image is sliced according to the amount of the macro block, and then each macro block of the slice is compressed in accordance with the objective bit rate. However the shortcoming of the average allocation is obvious. Because of the natural characteristic of the image, some of the region contains more texture, and some of the texture is uniform, some of the region contains the foreground information requiring more motion compensation bit rate, and some of the region contains the background information which even can adopt the skip mode when compressing. So the average method cannot reflect slicing the natural characteristic. Later a dynamically self-adaptive slicing strategy to realize EBR coding is referred to make the size of coded slice average distribution as far as possible. It first makes use of a feedback coding method to determine the reasonable slicing position in the first I frame and P frame of GOP, and then regards the result of slicing as the reference of remaining P frame in GOP. This slicing method can efficiently improve the utilization rate of channel resources and enhance the quality of the video encoding in relation to the fixed slicing method, and ensure the encoding efficiency at the same time.

The choice of quantization parameter of macro block and slice has to been considered except slicing. The slicing can only solve the optimizational distribution problem of output bandwidth, but the final output still must use choosing quantization parameter to control. Different from adopting one quantization parameter for each image, each slice must be strictly controlled to limit the output size in EBR algorithm, therefore the quantization parameter model has to been designed again to meet each slice bit rate requirement by setting different quantization parameter for each slice.

#### **3.4 Mode Decision**

In recent years, many fast algorithms on mode decision are present. In reference [20], macro block is classified based on the price of rate-distortion, and each mode type is tested according to a certain sequence, if the coding price of a certain mode is less than the threshold, the test to other remaining mode will stop. This method does not consider the correlation between the adjacent macro block modes.

In reference [21], the encoded macro blocks modes of the current frame and the former frame are used to predict the current macro block mode, the adjacent macro block mode which costs the smallest price is chosen as the current. This method only chooses one macro block of the former frame among the time-adjacent macro blocks, in which case the prediction mode is not always optimal, which may reduce the video encoding quality for the wrong mode prediction. In reference [10], the high complex problem of mode decision and the algorithms in reference [20] and [21] are analyzed, a fast mode decision algorithms is present. The concept of mode reference aggregate decision algorithm is proposed based on region forecast. This method can significantly reduce the number of candidate mode, thus simplifying the procedure of mode decision.

#### 3.5 Scalable Extension

A fundamental problem requiring to resolving on the video streaming transmission in internet is the network bandwidth fluctuation. Not only different people use different devices at different time, but also for the same one at the same time, even for the same video streaming, the data transmission rate from internet exists very widely difference. The video streaming generated by traditional encoding method is so hard to adjust to the complex fluctuation of the network bandwidth. There are two traditional methods to deal with this problem: transcoding and simulcast.

Transcoding is that the encoder first generates a high bit-rate stream, and then converts the bit stream to match the bandwidth before transmission. This method first increases the burden of the video server; secondly, converting from high rate stream to low rate stream will produce the extra video quality loss. Simulcast is to generate multiply different bit rate streams, choose an appropriate bit rate stream to fit the network bandwidth when transmission. This method is very difficult to implement the dynamic switching of the arbitrary bit rate stream in the process of transmission. The change of rate is also confined to a few specific rate, can not take full advantage of the capacity of the channel.Another problem is the package loss for video transmission in internet, in order to improve the coding efficiency in the coding process, every bit in the video stream has been strictly defined. The package loss even if a bit error can cause a big amount of vide stream not to decode correctly. So that the visual quality is decreased, and the error will affect many other later images quality by motion compensation.

At present, the optimal method to resolve the problems of network bandwidth fluctuation and package loss is to the scalable video coding. Scalable video coding is usually generating two streams: the basic layer and the enhanced layer. The basic level must been transmitted and has the low bit rate. The enhanced layer can carry out arbitrary cut-off of the bit stream in accordance with the bandwidth or not transmitted.

Since the scalable coding can be adaptive adjustment in a certain range of bit rate, it is suitable to the complex fluctuation of the network bandwidth. Besides, in addition to the basic layer stream need try its best to avoid the package loss, the package loss of the enhanced layer will not obviously decrease the visual quality and not affect other frames. So the scalable video coding has the better robustness.

SP/SI frame adaptively supports the bit rate adjustment in a big scale, can switch in different bit rate and different image quality, so it can make maximum use of the existing resources rather than induce the decoding error because of lack of reference frames. The basic principle of SP frame coding is similar to P frame, is still based on the motion compensation of inter prediction, but it can reconstruct the same frame image from different reference frame. Making use of this peculiarity can implement the random switch function of the bit stream, that is SP/SI frame can replace I frame in the use of bit stream switching, splicing, random access, skip forward/skip backward and error recovery etc. SI frame is corresponding to SP frame, use of intra prediction encoding technology. The shortcoming of SP/SI frame is that the more the stream is stored, the more the resource of the server is consumed.

There are tow usual ways to realize the scalable extension of video encoding: one is the use of the scalable coding methods, such as embedded bitplane coding; the other is the use of layered coding. The SVC extension of H.264/AVC combines layered coding and fine coding, realizes the complete flexibility of the space, the time and SNR.







Fig.6 SVC Encoder Structure

Simu lated Wireless System Type	Mode	Encoded Bit-rate (kbps)	Encoded Average PSNR(dB)	Encoded Size(KB )	RTP Packets Nu mber	Lost RTP Packets	Decoded Average PSNR(dB)
CDMA2000	Integrati -on of Key Technol -ogies	30.85	35.14	162	615	15	29.39
CDMA2000	No	51.24	37.54	276	1653	77	27.36

Table 1 Other Results Data Comparison of Wireless Error-prone Channel

The scalable stream of video transmission has another advantage, that is, different level is given the different protection strategy. The basic level can adopt the stronger protection strategy, such as: FEC and ARQ, a more flexible strategy to the enhanced level, such as: weak FEC or discarding directly.

#### 3.6 Experiment and Analysis

Not adopting FMO plus row intra-coded mode is chosen as error resilience strategy, rate control adopts self-adaptive EBR algorithm, mode decision adopts the concept of mode reference aggregate. Scalable coding is not considered. JVT H.264/AVC reference software JM10.2 is used as encoding and decoding software, the simulated RTP/UDP/ IP packet Packaging and wireless channel transmission are tested through VCEG-N80 reference software (MIP software). Test sequence is the foreman QCIF sequence sampling by 10frames/second, the transmission result is observed through the 2% error channel. The result is shown as Fig.7 and Table 1.

Spatial scalability is realized through up/down sampling filter changing the spatial resolution. Temporal scalability is realized through the internal temporal scalability in the coding structure of motion compensation and motion prediction. SNR scalability is realized by use of the embedded bitplane fine coding. SVC video stream transmission system structure for wireless channel is shown as Fig.5. The system includes: SVC encoder server, internet, wireless transmission network and all multiple mobile devices of SVC decoder. And SVC encoder structure is shown as Fig.6.

### 4 Conclusion

In this paper, after the study of the existing key strategies and technologies of H.264/AVC in the wireless channel application, a method of combination to compare and verify is proposed. Experimental results show that the method significantly reduces the rate and ensures the encoding quality at the same time in the higher error rate wireless channel, and the decoded average PSNR improves greatly. It is very valuable and meaningful for video services to apply in the wireless channel.



Fig.7 PSNR Comparison Results of Wireless Error-prone Channel

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