无线 IP 网络中一种针对实时流的报头压缩算法*

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A Header Compression Algorithm for Real-Time Streams in Wireless IP Networks

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Abstract: In wireless IP networks, it is necessary to adopt header compression schemes for the reduction of the protocol header size in order to save the scarce wireless bandwidth resource and make real-time services economically feasible and physically realizable. However, all the existing header compression schemes have not taken wireless channel state into consideration when designed. For a better usage over wireless links, a new channel state based robust header compression algorithm (CSB-ROHC) for real-time streams in wireless IP networks such as 3G platforms is proposed and analyzed in this paper. Through adjusting the dimension of VSW in W-LSB encoding (a key parameter in header compressor) with the accurate estimation of wireless channel state, this new algorithm can achieve a good balance of compression ratio and error-resistant robustness for its adaptive usage over wireless links. The authors present simulation results that demonstrate the effectiveness of the CSB-ROHC algorithm over wireless links.

Key words: robust header compression; channel state based; W-LSB encoding; wireless IP networks

摘 要: 无线 IP 网中,为了能够有效利用带宽资源,使提供实时业务具有经济上的可行性和现实性,必须采用报头压缩技术减小协议报头带来的额外开销.然而,现有的报头压缩方案在设计时没有考虑无线信道状态.为了能够更好地适应无线链路特性,提出并分析了一种无线 IP 网络中针对实时流的基于信道状态的健壮报头压缩算法.该算法通过对无线信道状态的精确估计来调节报头压缩器中 W-LSB 编码的可变滑动窗口的大小,在压缩率和抗差错健壮性

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之间能够实现好的平衡,并适应特性经常变化的无线链路.最后,通过仿真结果证明该算法在无线链路上的有效性.
关键词: 健壮报头压缩;信道状态;W-LSB 编码;无线 IP 网络
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1 Introduction

It is expected that the new telecommunication systems will be based on the convergence between mobile telephony and Internet, providing a wide variety of services independently of user location. In this scenario is being developing the third generation (3G) mobile system[1]: a true mobile service convergence of voice, data and image. Actually, the only way to make the 3G mobile system convenient seems to be a network platform totally based on the IP protocol, called "all-IP network"[2], that includes wireless links whose bandwidth is the most precious resource of the whole wireless system. However, the encapsulation process of the hierarchical transmission protocols wastes a substantial part of the wireless bandwidth for the transmission of control information (header) that does not have any specific function for the management of wireless channel itself.

This problem becomes even worse for real time voice services which will continue to be dominant in future mobile systems. For such services, speech data will most likely be carried by RTP[3]. A packet will then have an IP[4] header (20 octets), a UDP[5] header (8 octets), and an RTP header (12 octets) for a total of 40 octets. With IPv6[6], the IP header is 40 octets for a total of 60 octets. However, the size of the payload depends on the speech coding and frame sizes being used and may be as low as 15−20 octets. It is obvious that 67%−80% of the wireless channel bandwidth is wasted for headers transmission and only 20%−33% of the bandwidth is used for the transmission of real information for the final users.

For wireless systems, the scarce wireless resources must be utilized efficiently in order to provide mass-market services at reasonable prices. The header overhead introduced by transmission protocols must be reduced by header compression. Through the significant reduction of the header size for the same application, we have a lower demand of bandwidth (lower cost) as well as a greater efficiency and responsiveness.

The error-proneness (the bit error rate as high as 10e-3, even 10e-2) and the large delays (the round trip time as high as 100-200ms) of wireless links make the header compression defined in Ref.[7] (Compressed RTP) perform less than well. RObust Header Compression (ROHC)[8] scheme has been proposed by the IETF working group, which aims at providing a compression scheme that has high compression efficiency and high robustness when used over wireless links.

On the other hand, the multipath fading on a wireless channel is considered to follow a Rayleigh distribution. The issue of modeling a burst error process on a correlated Rayleigh fading channel has been addressed in Refs.[14,15], where it was shown that, for a broad range of parameters, the sequence of data block success and failure can be approximated by means of simple finite state Markov models. These models accurately represent the type of burst error behavior of the wireless channel by transitioning with a different probability between every state, each having a different error probability and a different duration.

However, all of existing header compression schemes have not taken the wireless channel state into consideration when designed. When used over wireless links, they act in an invariable way when wireless channel enters into different states. In addition, they just adopt variations of the encoding methods and repairing mechanisms to minimize the number of packets lost when error happens over wireless links. These known header compression schemes can't work very well over wireless links because they lack of flexibility to handle different wireless channel states. If the current channel state can be predicted, header compression schemes may take some effective actions and mechanisms according to the current channel state before compressing headers when the wireless channel enters into different states. Intuitively, this type of header compression scheme may suit much better for the use over wireless links.

In this paper, a channel state based robust header compression (CSB-ROHC) algorithm for real-time streams in wireless IP networks such as 3G platforms is proposed and analyzed. By adjusting the dimension of VSW (Variable Sliding Window) of the W-LSB (Window-based Least Significant Bits) encoding in header compressor with the accurate estimation of wireless channel states, this new algorithm can not only compress the RTP/UDP/IP headers robustly, but also achieve rather high efficiency. We present experimental results that demonstrate the effectiveness of the CSB-ROHC algorithm over simulated wireless links.

This paper is organized as follows: Section 2 briefly introduces state of the art of the RTP/UDP/IP header compression schemes, Section 3 describes the design of CSB-ROHC algorithm in detail, Sections 4 and 5 show the simulation environment and performance evaluation for CSB-ROHC, and Section 6 gives concluding remarks.

2 RTP/UDP/IP Header Compression: State of the Art

Header compression is possible for the presence of a significant redundancy among header fields, both within the same packet header and in particular among the consecutive packets belonging to the same packet stream. Thus, by sending static field information only initially and utilizing dependency and predictability for other fields, the header size can be significantly reduced for most packets.

Generally, header compression algorithms maintain, at both the compressor and the decompressor sides, a shared state called context, including relevant information to which compression and decompression are done relatively. Normally, the decompressor will keep synchronized with the compressor, thus it can decompress the compressed packet header correctly. But when packets are lost over the link, the decompressor context will be brought out of sync with the compressor context and the decompressing of subsequent headers will fail. This effect, so-called error propagation, which will last until the contexts are brought into synchronization by some means, will degrade the performance of header compression, especially on relative high BER links such as wireless channels.

CRTP is suitable for compressing RTP/UDP/IP headers over low speed serial links, which can compress the 40 or 60 octets RTP/UDP/IP headers to 2-4 octets. On lossy links with long round trip times such as most wireless links, CRTP does not perform well[9]. A viable header compression scheme for usage in wireless systems must produce very small headers to enable efficient usage of the scarce spectrum while still being robust to the error patterns.

Many schemes have been proposed to make header compression suit able for use in a wireless environment. ACE (Adaptive Header ComprEssion)[10] and ROCCO (Robust Checksum-based header COmpression)[11] are two significant ones among these schemes. ACE introduces an innovative W-LSB encoding algorithm for the changing fields that uses many reference values contained in VSW (Variable Sliding Window) in the compressor, while the decompressor can use any value of these references to decompress correctly. ROCCO uses a CRC included in the compressed headers to verify the correct reconstruction in the decompressor and to avoid the error propagation.

The IETF RObust Header Compression Working Group (ROHC WG) is studying new header compression schemes that are having good performances over wireless links of 3G cellular systems. The schemes should also be applicable to other future link technologies with great loss and long round-trip times. RObust Header Compression (ROHC) scheme which uses and combines all the techniques developed by ACE and ROCCO has been just proposed, which aims at providing a compression scheme that has high compression efficiency and high robustness when used over wireless links.

3 Channel State Based Robust Header Compression: CSB-ROHC

In this paper, we propose a channel state based robust header compression algorithm (CSB-ROHC) for real-time streams in wireless IP networks such as 3G platforms. The block diagram of CSB-ROHC is illustrated in Fig.1. In fact, in our realization, the function of channel estimator is included in header decompressor.

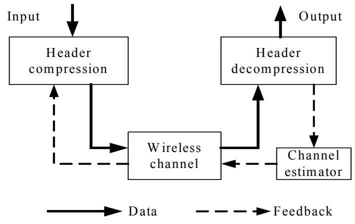


Fig.1 Function block diagram of CSB-ROHC

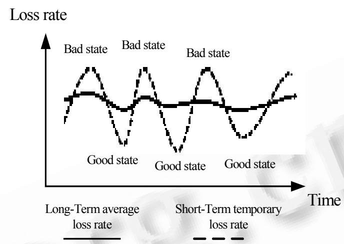


Fig.2 Good state and bad state

In CSB-ROHC, header compressor adopts a robust encoding technique which is called window-based least significant bits (W-LSB) encoding algorithm. We propose to use the accurate estimation of wireless channel state to control the size of Variable Sliding Window (VSW) of W-LSB encoding in order to achieve a good balance between compress ratio and robustness. CSB-ROHC may be adaptable to the burst error and delay characteristics of wireless links over which it is used.

Before introducing CSB-ROHC algorithm, it is important to brief the functionalities of W-LSB encoding that are most related to our work.

3.1 W-LSB encoding

CRTP does not perform well on wireless links which are characterized by high bit error rate and transmission delay because the decompressor is required to use always the same reference base value that is used by the compressor in order to decompress the encoded value correctly. However, the packet including this reference base value could be lost on the channel because of the bad channel conditions. The innovative Window-based Least Significant Bits encoding (W-LSB encoding) does not impose this restriction because it tries to encode a value based on a group of reference base values included by the sliding window (VSW). The decompressor can decompress the encoded value correctly once any value in VSW can be delivered successfully. By using W-LSB encoding, the compressor would not come into the inconsistency with the decompressor unless all values in the VSW are lost.

W-LSB encoding is used for header fields whose values are always subject to small changes among consecutive packets (*changing fields*). With W-LSB encoding, the k least significant bits of the field value are transmitted instead of the original field value because the Most Significant Bits (MSB) remain relatively constant during many sessions. The value of k is calculated in the compressor using N reference values included in the VSW: for each reference value, v_ref, k_ref is determined so that the value to compress belongs to the interpretation interval $f(v\_ref,k\_ref)$

$$f(v\_ref,k\_ref)=[v\_ref-p, v\_ref+(2\hat{}k\_ref-1)-p] \tag{1}$$

where p is an integer.

<------- interpretation interval (size is 2^k) ------->

|-------------+--|

v_ref-p v_ref $v\_ref+(2\hat{}k\_ref-1)-p$

For any value k_ref, the k_ref least significant bits will uniquely identify a value in $f(v\_ref, k\_ref)$. The parameter *p* is introduced so that the interpretation interval can be shifted with respect to *v_ref*. Right value of *k* is the maximum among the *N* values of the *k_ref* calculated above. After receiving *k* bits, the decompressor derives the original value using a previously received value as reference.

It should be clear how the dimension of VSW influences k and therefore the compression ratio too as *k* is typically a monotonous increasing function of the number of reference values *N*, i.e. the dimension of VSW. When *N* is small the scheme gives a high compression ratio, but losses of the packets are not allowed on the channel. Otherwise, when VSW is large, the scheme has strong robustness, low compression ratio, and low bandwidth efficiency. When the compressor knows what values in VSW have been received by the decompressor, VSW can be shrunk to obtain a rather high compression ratio. To shrink the VSW, the compressor needs some means to get feedbacks that indicate what value has been received by the decompressor.

Considering the features of W-LSB encoding, the header compression mechanism we are going to propose introduces an innovative algorithm estimating optimum VSW dimension in wireless links: the channel state based robust header compression.

3.2 Description of CSB-ROHC algorithm

Many different variations of the methods to estimate the wireless channel states have been proposed in the literature. In CSB-ROHC, we adopt the method used in TCP-ELSA[12] to predict channel states. The state of link is defined in terms of the ratio of its temporary loss rate to its long-term average loss rate as depicted in Fig.2 instead of defining the state of a link according to the absolute value of the loss rate. The solid line in Fig.2 represents the long-term average loss rate while the dashed line represents the short-term temporary loss rate. If a link's temporary loss rate is higher than its average loss rate, the link is considered to be in a bad state; otherwise, it is considered to be in a good state.

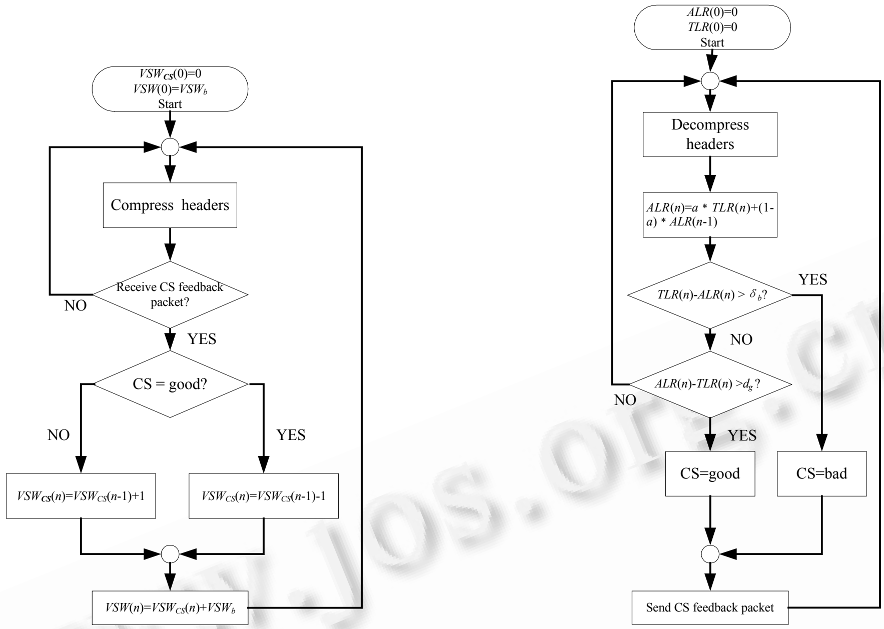
CSB-ROHC estimates the link state by maintaining two parameters: the average packet loss rate (ALR) and the temporary packet loss rate (TLR), using a weighted moving average algorithm. By assigning different parameters to the weighted moving average, ALR can represents the long term quality of the wireless link and TLR can be set to be more sensitive to transient packet loss rate changes and indicate the current packet loss rate.

According to this information, CSB-ROHC adjusts the dimension of VSW in header compressor, which is the key parameter for compression performance. When VSW is small the scheme has a high compression ratio but it cannot avoid many packet losses. Instead, when VSW is large the scheme has strong robustness, but weak compression ratio and weak bandwidth efficiency.

The header decompressor determines the state of wireless link according to the values of ALR and TLR, and notifies the compressor the estimation result by sending a specially defined type of feedback packet. Every time receiving such type of feedback packet, the compressor adjusts the dimension of VSW accordingly. When TLR is larger than ALR for a certain threshold, which signals that the wireless link is in a temporarily bad state, the compressor increases the dimension of VSW. In this state, CSB-ROHC can tolerate more packets lost over the wireless link by increasing the number of reference base residing in VSW. After several rounds, TLR gets lower than ALR for a certain threshold, so the wireless link is in a normal good state. In this state, the compressor decreases the dimension of VSW accordingly to provide a greater compression ratio. These thresholds can be set to be different values depending on the specific system and wireless channel.

3.3 Implementation of CSB-ROHC algorithm

According to the algorithm described above, we give the following program flow chart of CSB-ROHC implementation as shown in Fig.3. It includes two parts: one part is the VSW dimension adjustment in header compressor and the other is the channel estimator in header decompressor.



(a) VSW dimension adjustment in header compressor (b) Channel estimator in header decompressor

Fig.3 Implementation flowchart of CSB-ROHC

In Fig.3, CS means channel state. We indicate the actual value with the expression similar to $A(n)$ and the previous value with $A(n–1)$. At current time, the dimension of VSW is expressed as:

$$VSW(n) = VSW_{CS}(n) + VSW_b \tag{2}$$

where $VSW_{CS}(n)$ depends on the channel state, while VSW_b is a basic constant larger than 1. $VSW(n)$ is dynamically adjusted according to the algorithm depicted in Fig.3(a).

In channel estimator, the current ALR is calculated according to the weighted moving average filter algorithm:

$$ALR(n) = \alpha \times TLR(n) + (1-\alpha) \times ALR(n-1) \tag{3}$$

where the value of the filter constant, α, dictates the degree of filtering, i.e. how strong the filtering action will be. Its value belongs to (0,1). δ_b and δ_g represent the thresholds respectively when the channel is in a bad or good state. Their value also belongs to (0,1).

4 Simulation Environment

In order to evaluate the performance of CSB-ROHC algorithm, we apply this algorithm to the existing header compression schemes CRTP and ROHC separately. The new schemes implemented are called CRTP+ and ROHC+. In CRTP+ and ROHC+, CSB-ROHC algorithm is applied to the fields: IP identification, RTP timestamp and RTP sequence number.

Our simulations have been conducted according to the block diagram shown in Fig.4. The implementation of channel estimator function is included in header decompressor.

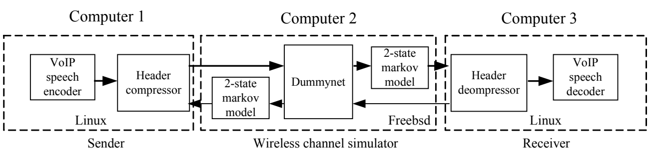


Fig.4 Simulation scenario

The VoIP speech encoder generates a packet with a fixed size, 264 bits, every 20 ms (13.2kbps), corresponding to the GSM speech codec. The length of the talk spurts and the silence intervals between them are both exponentially distributed with an expected length of 1 second. Silence suppression is used.

The wireless channel simulator includes two parts: dummynet[13] and 2-state Markov model[14,15].

Dummynet is used to simulate the bandwidth and delay management of wireless channel. In our experiments, three types of wireless links are simulated, as shown in Table 1.

Table 1 Three types of wireless links

Link type	Bandwidth (kbits/s)	Delay (ms)	BW*Delay (KB)
GSM[16]	9.6	150	0.2
UMTS[17]	384	70	3.3
WLAN 802.11b[18]	11000	10	13

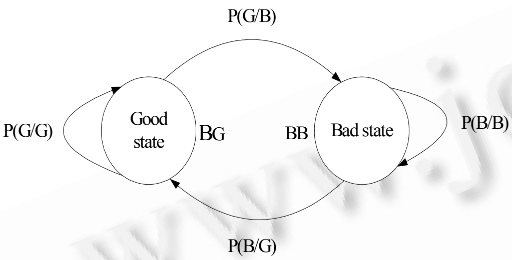


Fig.5 2-State Markov model

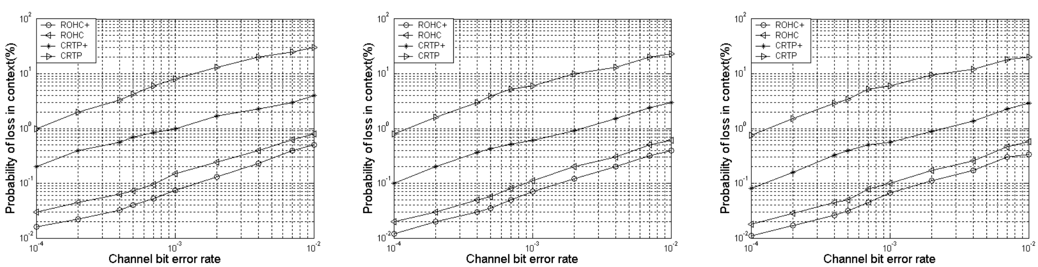
For simplicity, the 2-state Markov model illustrated in Fig.5 is used to simulate the burst error characteristic of wireless channel.

The good state represents the error free behavior of the link, while the bad state represents a fade. In each state, bit errors are considered to be uniformly distributed, whose mean bit error rates are respectively $\boldsymbol{B_B}$ and $\boldsymbol{B_G}$. The durations of each state are also uniformly distributed with average durations $\boldsymbol{T_B}$ and $\boldsymbol{T_G}$. The bit error rate can be got through setting the different values of $\boldsymbol{B_B}$, $\boldsymbol{B_G}$, $\boldsymbol{T_B}$ and $\boldsymbol{T_G}$. In addition, the feedback link used in simulation is supposed to be independent of wireless link although they have the same link characterization.

5 Simulation Result and Analysis

5.1 Probability of loss in context

Figure 6 indicates the variation of the robustness of each scheme, quantified as probability in context loss versus channel BER. The probability of context loss for CRTP is much higher than these for other three schemes even using the algorithm twice. Since the robustness of W-LSB encoding, CRTP+ is much more robust than CRTP, and the context is rarely brought out of sync for ROHC and ROHC+. Furthermore, ROHC+ is more robust than ROHC. In ROHC, when the number of packet lost is larger than the predefined dimension of VSW, the context between header compressor and decompressor will be out of sync. However, in ROHC+, through using CSB-ROHC algorithm to enlarge the dimension of VSW in advance, such case will be avoided.

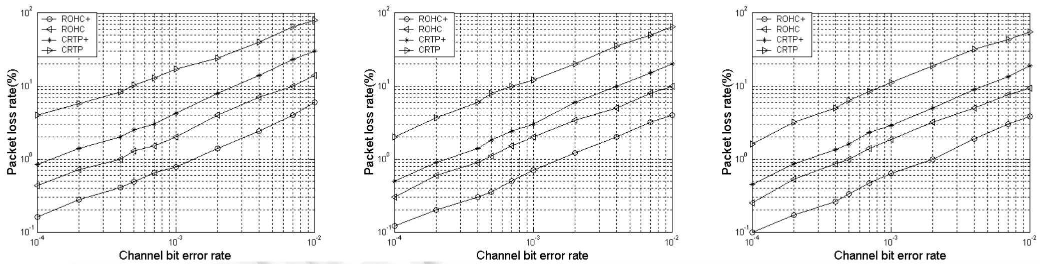


(a) GSM (b) UMTS (c) WLAN 802.11b

Fig.6 Probability of loss in context versus channel BER

5.2 Packet loss rate

Figure 7 illustrates the packet loss rate versus channel BER for various schemes. A packet loss between CRTP compressor and decompressor results in a burst of additional packet loss due to CRTP's repair mechanism. Due to CSB-ROHC algorithm, the packet loss rate of CRTP+ is decreased significantly when compared with that of CRTP. Since ROHC+ can adjust the dimension of VSW according to the current channel state, resulting in less probability of context loss, its packet loss rate is approximately 2 times less than that of ROHC. Figure 7 also shows that when using ROHC+ scheme over each type of the simulated wireless links, the packet loss rate is all less than 1% when channel BER is not excessively high ($<10^{-3}$).



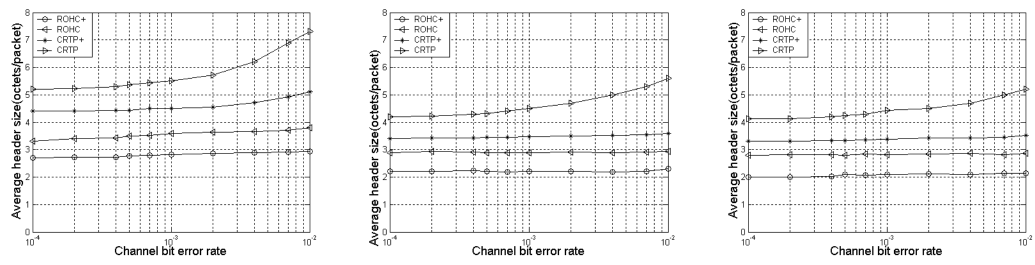
(a) GSM (b) UMTS (c) WLAN 802.11b

Fig.7 Packet loss rate versus channel BER

On the other hand, on GSM link, since channel bandwidth (9.6kbps) is less than the voice date rate (13.2kbps), many packets lost on channel mainly for the reason of queue, the packet loss rate is larger than for those UMTS and WLAN 802.11b. The packet loss rate on WLAN 802.11b link is similar to that on UMTS link except for CRTP, which suggests that the channel bandwidth and round trip time impose little impact on CRTP+, ROHC and ROHC+ schemes while the performance of CRTP is heavily impacted by the round trip time of link.

5.3 Average header size

Figure 8 shows the average header size plotted against channel BER over three types of channel. CRTP needs a context update packets (17 octets) for each loss, so the average header size increases as the link BER increases. Since using W-LSB encoding, the context is rarely brought out of sync for CRTP+, ROHC and ROHC+: the average header size for them is slightly larger than the minimum header size. With the help of CSB-ROHC algorithm, CRTP+ and ROHC+ can adjust the dimension of VSW quickly when channel enters into different states. Their average header sizes are respectively nearly 1 octet smaller than those of CRTP and ROHC. Furthermore, on GSM link, since many packets lost in channel due to queue, the average header size is larger than that on UMTS link or WLAN 802.11b link. The average header size for WLAN 802.11b link is little less than that for UMTS link.



(a) GSM (b) UMTS (c) WLAN 802.11b

Fig.8 Average header size versus channel BER

6 Conclusions

This paper proposes a new channel state based robust header compression algorithm (CSB-ROHC) for real-time streams in wireless IP networks such as 3G platforms. Through adjusting the dimension of VSW in W-LSB encoding (a key parameter in header compressor) with an accurate estimation of wireless channel state, this new algorithm can achieve a good balance of the header compression ratio and error-resistant robustness for its adaptive usage over wireless links. With CSB-ROHC, we implement two new header compression schemes: CRTP+ and ROHC+.

Simulation results show that the performance of CRTP+ is much better than that of CRTP both in header compression ratio and error-resistant robustness when used over wireless links. The performance of ROHC is improved significantly when adding CSB-ROHC algorithm. When using ROHC+ scheme in wireless VoIP, the packet loss rate is less than 1% when channel BER is not excessively high ($<10^{-3}$) and voice quality is sufficiently better. Simulation results also show that when used on a narrow bandwidth channel, the performance of header compression schemes decreases for the reason of packet lost because of queue. When used on broadband channel, the channel bandwidth and round trip time of link impose little impact on CRTP+, ROHC, and ROHC+ schemes while the performance of CRTP is heavily impacted by the round trip time of link.

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