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A Wireless Channel Model Based Rate Control (WMRC) Scheme in RTP/UDP
for Real Time Multimedia Transmissions over Wired-Wireless Networks
(WMRC-RTP/UDP)
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Abstract

   This document introduces a Wireless Channel Model Based rate control
   scheme for improving the behavior of real time multimedia streams

with TFRC [1] in wireless-wired heterogeneous networks. Based on
wireless channel characteristics, the method can first identify the
level of packet losses of two different types by sending large and
small packets alternately, then adopt different adaptive rate control
strategies to increase the network throughput and decrease congestion
packet loss rate, to improve transmission quality of real-time
multimedia stream. The proposed method is compared with previously
reported algorithms [2-3] by simulation. It is shown from the
simulation results in different network topology environments, the
performance the proposed algorithm is better than existing algorithms
in the aspects of network bandwidth utilization and congestion packet
loss control. Parts of this method published in the IEEE Wireless
Communications and Networking Conference (WCNC2007) and in a pending
China patent (access number: CN101686100).

Table of Contents

1. Introduction

    This document describes a wireless channel model based packet loss
    discrimination method that can differentiate wireless random bit
    error packet loss from congestion packet loss over wireless-wired
    networks. This work focuses on how to obtain the dynamic change
    characteristics of wireless communication network from the
    transport/application layer, and to achieve a better end-to-end
    quality of service by making adaptive adjustment according to these
    changes. The proposed scheme shows to be more accurate than existing
    methods in estimating current network status by means of a wireless
    channel model and statistical analysis of large and small packets
    loss rates, and its performance basically not affected by the
    variation of network topology and the competition flows. The real-

time multimedia transmission protocol can carry out performance
optimization based on the packet loss discrimination results. For
example, if only the wireless losses are reported, the source and
channel coding ratio can be adjusted to increase the data protection
instead of reducing sending rate.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT",
"SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this
document are to be interpreted as described in RFC-2119 [RFC2119].

3. Packet Loss Discrimination

Previous research indicates that the packet loss probability caused
by random bit errors is related to the packet size in wireless
environment, namely, the larger the packet size, the higher the
packet loss probability. The congestion packet losses (drop-tail
router), however, are generally independent of the packet size. Based
on this observation, one can send small probing packets regularly to
the channel to distinguish from the large-size video packets, and
estimate current main cause of packet losses from their feedback
information. However, channel bandwidth utilization will be reduced
by these probing packets. Therefore, we use small-size video packets
instead of probing packet to improve channel bandwidth utilization. A
method for identifying packet loss reason from the packet loss rates
of large and small packets is developed below.

In order to identify between congestion packet losses and wireless
fading losses, the sender node sends large and small packets
alternately and the statistics of the lost large and small packets
over a period is calculated at the receiver end. For ease of analysis,
let us define the following variables:

Nps = the number of lost small packets;

Npsc = the number of lost small packets due to congestions;

Npsb = the number of lost small packets due to bit errors;

Npl = the number of lost large packets;

Nplc = the number of lost large packets due to congestions;

Nplb = the number of lost large packets due to bit errors.

According to traditional communication theory, the wireless packet
loss rate varies exponentially with the channel bit error rate r. In
some cases such as uniform distribution with a very small r, the
variations of packet loss rate with r can be approximated by a linear
function. For the wired channel, as mentioned above, the numbers of
large and small congestion packet losses in the given period would
approximately be equal, namely Nplc=Npsc. Thus, we have the following
equations:

Npl = Nplc + Nplb (1)

Nps = Npsc + Npsb (2)

Let B denote the ratio of Nplb to Npsb in certain wireless channel
condition. Obviously, B is a function of the channel bit error rate r
and can be expressed as:

B(r) = Nplb / Npsb (3)

Thus, equation (1) can be rewritten as:

Npl = Npsc + B(r) * Npsb (4)

At the receiver end, one can obtain the statistical values of Npl and
Nps over a time interval. If B is known, one can get the values of
Nplc (=Npsc), Nplb and Npsb by solving above equations, namely, the
congestion loss rate and random erroneous loss rate of large and
small packets respectively. One can then know current congestion
level of the wired networks from Nplc and Npsc, and fading condition
of the wireless link from Nplb and Npsb.

The problem now is how to know the value of B, and there seems very
few works have been done on this issue. In [4], B was assumed to be a
constant, dependent on the sizes of large and small packets. This
however is not always true according to our experimental results.
Therefore, by taking this assumption in solving equations (2) through
(4) for other variables, the applicable scope of the obtained values
will probably be rather limited.

To analyze the relations between packet loss rates and the packet
sizes, a group of experimental tests in wireless channels have been
carried out by using a modified Jakes Rayleigh fading model [5] in
different channel BERs (Bit Error Rates) and packet sizes. Linear,
quadratic and exponential curves are used to fit the obtained
simulation data where the exponential fitting has the largest fitting
errors.

Let G denote packet length, and assume the random bit error of a wireless channel is subject to uniform distribution and any two bit error events are uncorrelated. The packet loss error rate P of wireless link can thus be computed as

$$P=1-(1-r)^G \quad (5)$$

If r is very small, by Taylor expansion in r = 0 and omitting higher order terms, the linear (first-order) and second-order approximation of equation (5) can be obtained as follows,

Linear approximation (first order): $P=r*G$ (6)

Second-order approximation: $P=r*G*(1-r*G/2)$ (7)

If the linear approximation is used, we have,

$$Pl/Ps = Gl/Gs = b \quad (8)$$

Where Pl and Ps are the loss rate of large and small packets respectively; Gl and Gs are the lengths of large and small packet respectively; b denotes the ratio of the large packet length to small packet length. B in this condition is a constant (=b), which is frequently used in previous literatures. However, when r is not small enough, the above equation is not applicable. At this time, one can consider second-order approximation and the ratio can be obtained from equation (7) as,

$$Pl/Ps = b*\frac{2-r*b*Gs}{2-r*Gs} = b*A(r) \quad (9)$$

Where $A(r) = (2-r*b*Gs)/ (2-r*Gs)$ (10)

When b = 2, the above equation becomes,

$$A(r) = (1-r*Gs)/ (1-r*Gs/2) \quad (11)$$

From above analysis, we can see that, when r is not small enough, Pl/Ps is not a constant but determined by A. In most practical settings, the reasonable value scope of A in equation (11) is 0.6~1 on condition that $r<4/7*Gs$ (In fact, this is not a necessary condition). For example, when Gs=4000bits (500Bytes), then $r<1.4*10^{-4}$, and when Gs=800bits (100Bytes), $r<7.1*10^{-4}$. That is to say, the bit error rate is kept in a small or medium value range. Otherwise,

the second-order approximation (equation (7)) may not be applicable.
A(r) monotonically decreases for a given value G with the increase of
r.

On the other hand, let us define the following variable,

$\&N = Npl - Nps = Nplc + Nplb - Npsc - Npsb = Nplb - Npsb = \&Nb$ (Since
Nplc =Npsc) (12)

From equation (12), we can know that the difference between the
number of lost large and small packets is equal to the difference
between the number of lost large and small packets due to bit errors.
The wireless packet loss rate P increases linearly with packet size
under certain r ranges, namely,

$\&P=k*\&G$ (13)

Where $\&P=Pl-Ps$, $\&G=Gl-Gs$ and k (k>0) is an r-related constant. Since
&P can be calculated from equation (12), and &G is known in advance,
constant k can be obtained, that reflects the degree of link bit
errors.

The slope of packet loss rate curve decreases with packet length
increase when BER is relatively high; while the curve slope almost
keeps stable and uncorrelated to packet length when BER is small,
namely, the packet loss rate varies linearly with packet length under
small BER conditions. Therefore, one can estimate the current level
of r from k within the scope of $r<1.8*10^{-4}$.

As for congestion packet losses in wired networks, the adaptive rate
control mechanism TFRC[1], decreases the sending rate when high
packet loss rate is reported, and allows the loss rate to decrease
quickly (assuming the video stream shares the network bandwidth with
other TCP-friendly traffic) [6]. According to our NS2 simulation
results, the packet loss rate can normally drop by more than 50%
within 4 RTT (Round Trip Time) time.

Based on above analyses, we propose the following computation
strategy:

1) When the packet loss rate rises to an unacceptable level, the rate
control mechanism will decrease the sending rate. Thus, (1) If this
high loss rate is due to congestions of the wired link, then as
discussed above, the loss rate will drop considerably within several
RTTs; (2) If however, the loss rate doesn't show any obvious drop
within the time interval, one may regard present packet losses most

probably due to the bit errors of the wireless link rather than
network congestions.

2) If present packet losses are indeed mainly due to bit errors of
the wireless link, we may assume the number of congestion packet
losses Npsc in (2) and (4) much less than the number of erroneous
packet losses Npsb, and Nplc << Nplb. Equations (2) and (4) can then
be simplified as:

Nps = Npsb (14)

Npl = B(r)*Npsb (15)

From above equations, one can estimate the value of B, and at the
same time identify the main cause of current packet losses. Note that
in our method, there is no assumption of the linear correlation of
the erroneous packet loss rate with the packet size.

3) A linear prediction with error correction method is adopted to
estimate the value of B. If present packet losses are mainly due to
bit errors of the wireless link, then

Bt = d*Bt - 1 + (1-d)*&Bt (B0 is the initial value) (16a)

Otherwise, Bt=B0 (16b)

Where B0=Gl/Gs; &Bt=Npl/Nps is the prediction error corrective value;
d is a weighting coefficient (0<d<1); t=1, 2, ···, denotes sampling
time with sampling interval &t=q•RTT (q>0, a constant).

As mentioned above, B is no longer a constant when wireless random
bit errors are high. Therefore, we adopt the linear prediction with
error correction (corrective value &Bt) method to gradually track the
value of B as shown in equation (16a). The weighting coefficient d
determines tracking speed of Bt to the real value of B that is not a
constant, and the selection of d should compromise between tracking
speed and stability of B. Equation (16b) represents the case of low
BER of wireless link, where packet losses are mainly due to network
congestion and B can be approximated by a constant (see equation (8)).

4. Adaptation to Network State Changes

For timely response to the change of network status, a finite length
history record based large and small packet loss statistics is used.
In a sliding window, Nps and Npl record the lost number of small and
large packets respectively. When a new packet loss occurs, the oldest
recorded one will be removed from the record queue, and the

statistical value will be updated correspondingly. The main concern here is the selection of the queue length: on one hand, if the queue is shorter, the response to the change of network status will be quick, but may cause stability problem; on the other hand, if the queue is too long, the sending rate will be relatively stable, but the response to network state changes may be slow. Therefore, the length of lost packets record queue represents a compromise between response time and control stability.

In order to track the change of the network states effectively, we introduce another statistic to help determine the current network states. Let,

$$Dec = 2*(B*Ps-Pl) / ((B-1)*(Ps+Pl)) \quad (17)$$

And, $Pc = (Npsc+Nplc) / (Nps+Npl) \quad (18)$

From equations (1)-(4), one can see that the statistic Dec is equal to Pc in a statistical sense, and reflects the ratio of the number of congestion packet losses to the number of total lost packets. Therefore, the value of Dec also reflects the congestion/wireless packet loss status of current networks. The larger the value is, the more serious the congestion packet losses. The values of Pl and Ps in equation (17) are calculated by the reciprocal of the average length of intervals of packet loss events [1].

The advantage of calculating Dec by equation (17) is: if no packet loss happens for a long time, although no new packet loss event occurs, the interval between packet loss events gets longer, and the congestion reduction can be reflected timely by the weighting coefficient of sliding window. This means the weight of loss events that occur more recently is larger, and that of older loss events is smaller. In this way, we can track current network status more quickly, and at the same time achieve good control stability.

5. WMRC Behavior Description

5.1. Adaptive Rate Control Mechanism

To satisfy the TCP friendliness requirement, an adaptive rate control based on TCP throughput model is used in this paper. The used TCP throughput mathematical model is as follows [1]:

$$rate= \frac{TU}{RTT*sqrt(2*p/3) + (4*tout*(3*sqrt(3*p/8)*p*(1+32*p^2)))} \quad (19)$$

Where, TU denotes data packet size, RTT denotes round trip time, tout
denotes retransmission timeout time (tout = 4 RTTs), P denotes packet
loss rate. Equation (19) works well in wired IP environment, but in
wireless environment, the performance of conventional TCP/TFRC is
often unsatisfactory. Therefore, we modify this model a bit by
removing the number of lost packets due to wireless bit errors in the
calculation of P. Thus the capability to differentiate the packet
loss types will be essential.

In the WMRC algorithm, when the receiver detects a packet loss event,
it calculates the Dec first, then judges whether it is a congestion
packet loss by comparing the value of Dec with a given threshold Dth
(0<Dth<1). If it is judged as a congestion packet loss (Dec > Dth),
this packet loss event will be used in the calculation of packet loss
rate P in Equation (19); otherwise, it won't be. Thus, the packet
loss rate P obtained from the RTCP feedback packets at the sender
reflects only the congestion packet loss rate, not including the
wireless packet losses. And the calculated sending rate will be fit
the current network situation well. Then, the situation of excessive
restriction of the sending rate will not happen. The smaller the
value of Dth is, the greater the possibility of judging the current
packet loss as the congestion loss; conversely, the greater the
possibility of judging it as the wireless random packet loss. We
generally set Dth=0.8 after running a number of experimental tests.

5.2. WMRC Specific Implementation Steps

The proposed scheme is based on RTP/UDP transport layer protocol [7],
and works in following steps:

1) Initialization: after setting the initial sending rate, the
parameters of a, b, d, q, Gl, Gs, Dth, and the length of record queue
of packet losses, the algorithm enters a slow start stage.

2) When the media streaming gets into a stable state when packet loss
occurs, the receiver end computes the statistical values of Npl, Nps,
Pl, Ps, RTT, and P (recording the congestion packet loss rate only).

3) The receiver end calculates the value of Dec by equation (17) for
every packet loss event, and estimates the reason of the current
packet loss. If Dec > Dth, the packet loss is judged as the
congestion loss, otherwise, the wireless packet loss. If it is a
congestion packet loss, it will be used to update the packet loss
rate P; otherwise it will not be. At every sampling time point t, Bt
is updated by equation (16) according to the nature of the current
packet loss. At every certain time interval, the receiver informs the
sender the RTT and packet loss rate P by RTCP feedback packets.

4) If there are packet losses occurred during the feedback period, when the sender receives the RTCP feedback packets, rate control will be carried out according to the equation (19). Otherwise, a MIMD(a, b) congestion control scheme [8] will be adopted.

6. Security Considerations

WMRC is not a transport protocol in its own right, but a congestion control mechanism that is intended to be used in conjunction with a transport protocol. Therefore security primarily needs to be considered in the context of a specific transport protocol and its authentication mechanisms. Congestion control mechanisms can potentially be exploited to create denial of service. This may occur through spoofed feedback. Thus any transport protocol that uses WMRC should take care to ensure that feedback is only accepted from the receiver of the data. The precise mechanism to achieve this will however depend on the transport protocol itself.

In addition, congestion control mechanisms may potentially be manipulated by a greedy receiver that wishes to receive more than its fair share of network bandwidth. A receiver might do this by claiming to have received packets that in fact were lost due to congestion. Possible defenses against such a receiver would normally include some form of nonce that the receiver must feed back to the sender to prove receipt. However, the details of such a nonce would depend on the transport protocol, and in particular on whether the transport protocol is reliable or unreliable.

We expect that protocols incorporating large/small packet with WMRC will also want to incorporate feedback from the receiver to the sender using packet loss discrimination. The packet loss discrimination is a modification to TFRC that distinguishes the loss packets from congestion loss or wireless random error.

7. IANA Considerations

There are no IANA actions required for this document.

8. Conclusions

This document presents a wireless channel model based rate control scheme WMRC for wireless multimedia transmission control over hybrid networks. This scheme can detect the network status and differentiate packet loss types (wireless or congestion losses) by means of a wireless channel model and a special packet sending scheme with different packet sizes. It can adapt to the dynamic change of the networks and control the sending rate effectively. Theoretical

analysis and detailed implementation of the proposed scheme are given. At the same time, it should be noted that due to the use of large and small packets alternately in the proposed algorithm, the application layer needs to pack data into two different-size packets for transmission, which may increase the overhead of packing process.

9. References

9.1. Informative References

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