画像情報特論 (3)

ハイブリッドTFRC

情報理工学専攻　甲藤二郎

E-Mail: katto@waseda.jp

NS-2 TCP-Linux

http://netlab.caltech.edu/projects/ns2tcplinux/

NS-2 TCP-Linux (1)

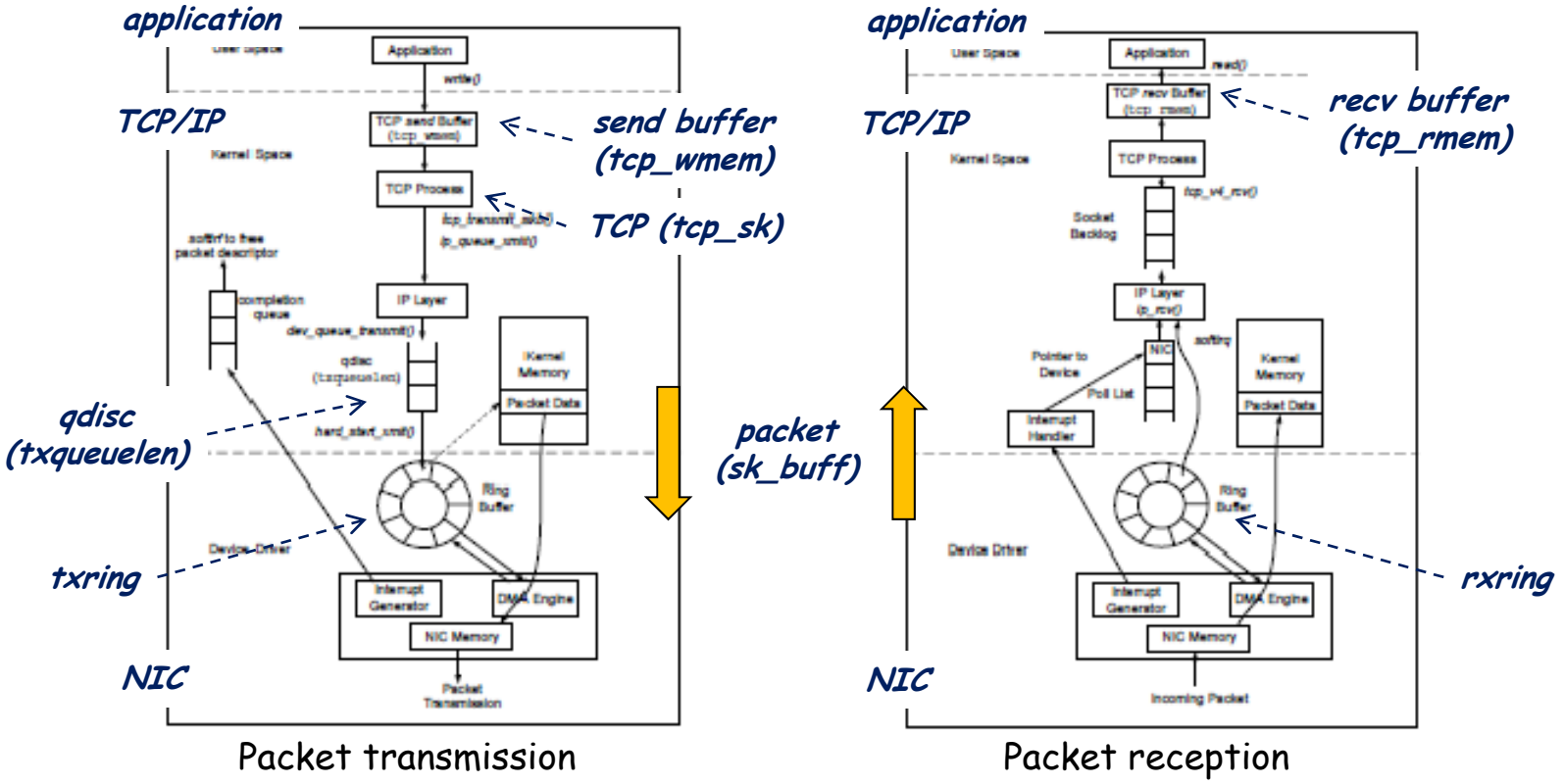
- Linuxカーネル内のTCP実装コードを使ったns-2シミュレーション
  - Linuxカーネル（実装）とns-2（シミュレーション）の橋渡し
    - 実装とシミュレーションの乖離の回避
  - 実装コードの検証
  - 新規アルゴリズムのLinuxカーネルへの容易な組込み

NS-2 TCP-Linux (2)

- Linux実装されているTCP (2.6.16-3現在)
  - TCP-Reno, TCP-Vegas, HighSpeed-TCP, Scalable-TCP, BIC-TCP, CUBIC-TCP (default), TCP-Westwood, H-TCP, TCP-Hybla
- 追加が検討されているTCP
  - TCP-Veno, TCP-LowPriority, Compound-TCP (Windows)

NS-2 TCP-Linux (3)

- TCP Implementation in Linux



Packet transmission

Packet reception

http://www.ece.virginia.edu/cheetah/documents/papers/TCPlinux.pdf

NS-2 TCP-Linux (4)

- Variables in tcp_sk

Name	Meaning	Equivalent in NS-2 *TCPAgent*
snd_ ssthresh	slow-start threshold	ssthresh_
snd_ cwnd	integer part of the congestion window	trunc(cwnd_)
snd_ cwnd_cnt	fraction of congestion window	trunc(cwnd_^2) %trunc(cwnd_)
icsk_ca_ priv	a 512-bit array to hold per-flow state for a congestion control algorithm	n/a
icsk_ca_ ops	a pointer to the congestion control algorithm interface	n/a

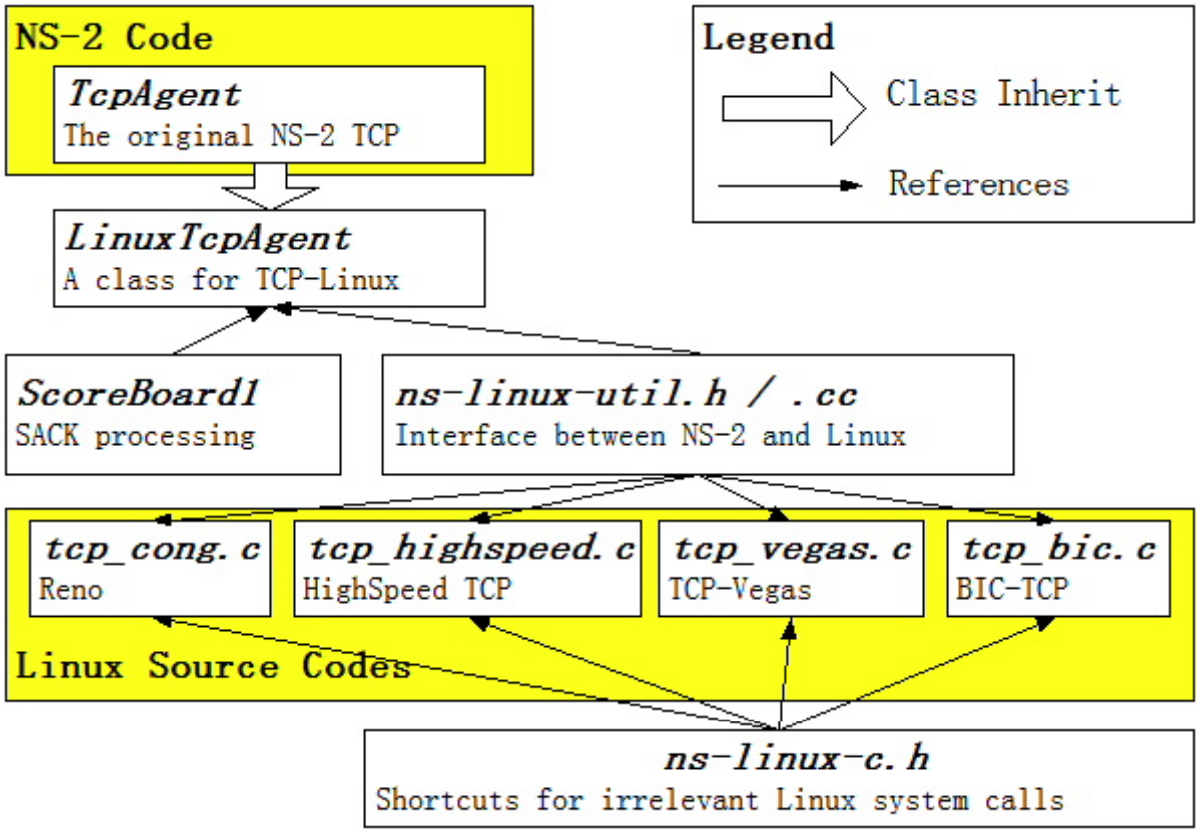
Congestion control modules:

cong_avoid: slow start & congestion avoidance
ssthresh: loss event handling
min_cwnd: fast retransmission

http://netlab.caltech.edu/projects/ns2tcplinux/

NS-2 TCP-Linux (5)

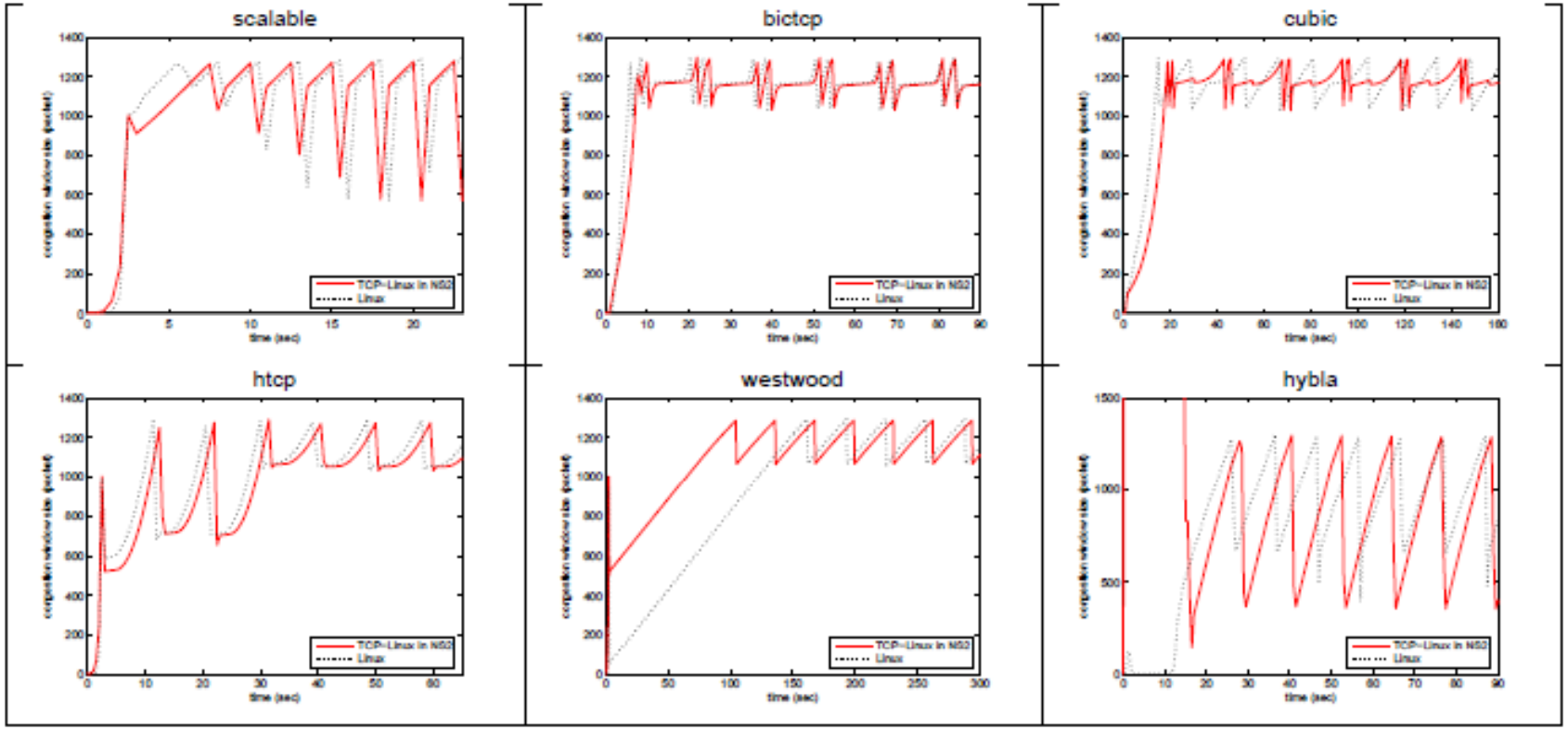
- Code structure



http://netlab.caltech.edu/projects/ns2tcplinux/

NS-2 TCP-Linux (6)

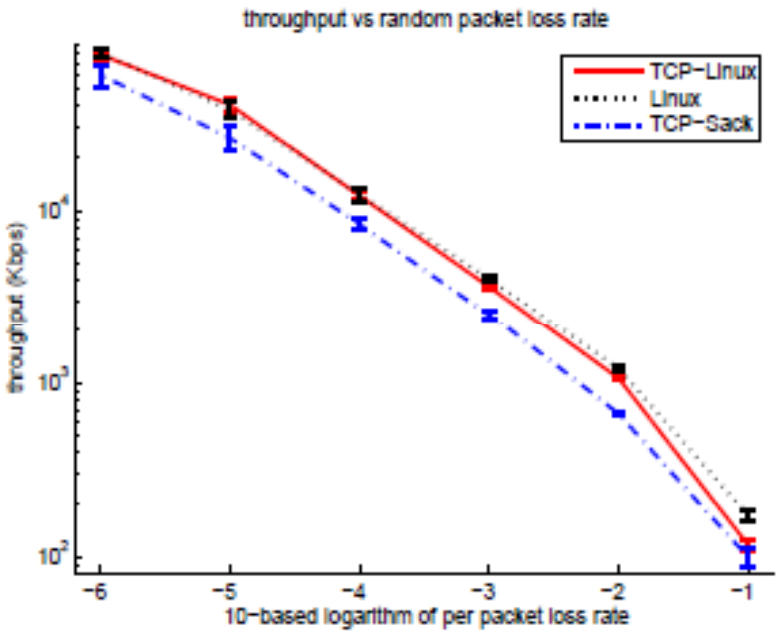
- Simulation (1) ns-2 & Linux



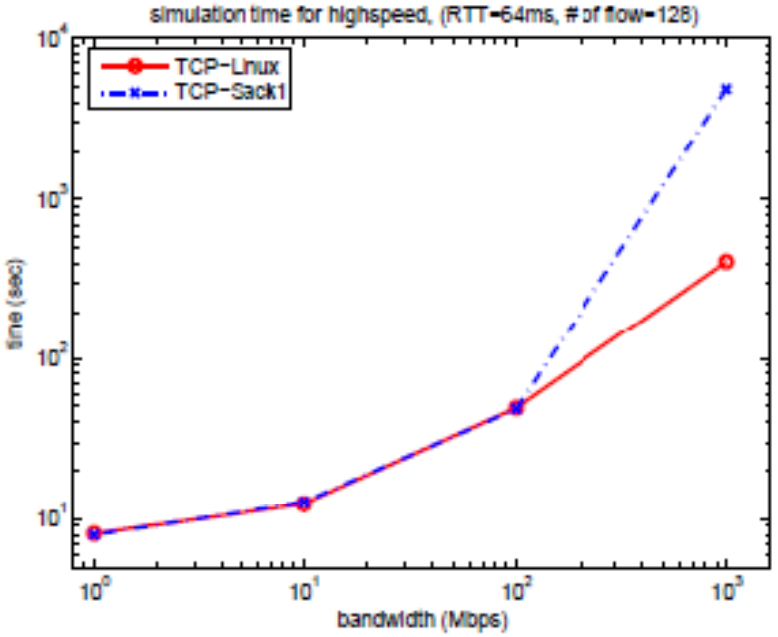
http://netlab.caltech.edu/projects/ns2tcplinux/

NS-2 TCP-Linux (7)

- Simulation (2) accuracy & speed



Accuracy



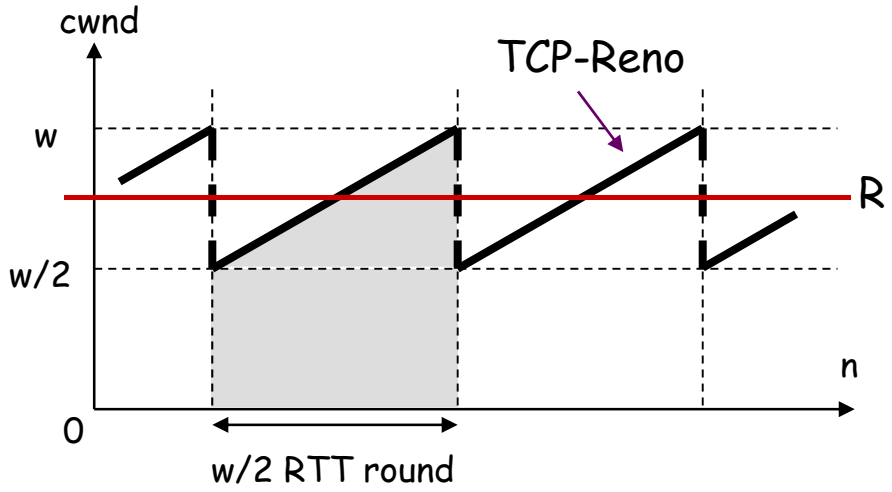
Speed

http://netlab.caltech.edu/projects/ns2tcplinux/

TCP Equations

TCP Modeling

- TCP-Reno Equivalent Rate



w: パケット廃棄発生時のcwnd
p: パケット廃棄率
RTT: ラウンドトリップ遅延
R: (等価) レート
b: delayed ACK の個数

$$\begin{cases} p = \dfrac{8}{3w^2} \\ R = \dfrac{PS}{RTT} \cdot \sqrt{\dfrac{3}{2p}} \end{cases}$$

タイムアウト考慮

$$R_{loss} = \frac{PS}{RTT\sqrt{\frac{2bp}{3}} + t_{RTO,loss} \cdot 3\sqrt{\frac{3bp}{8}} \cdot p(1+32p^2)}$$

J.Padhye et al: "Modeling TCP Throughput: A Simple Model and its Empirical Validation", ACM SIGCOMM 1998.

TCP Westwood

- Duplicate ACKs

FSE: Fair Share Estimates

$$ssthresh = FSE * RTT_{\min}$$

$$if\ (cwnd > ssthreh)\ cwnd = ssthresh$$

- Timeout

$$ssthresh = FSE * RTT_{\min}$$

$$cwnd = 1$$

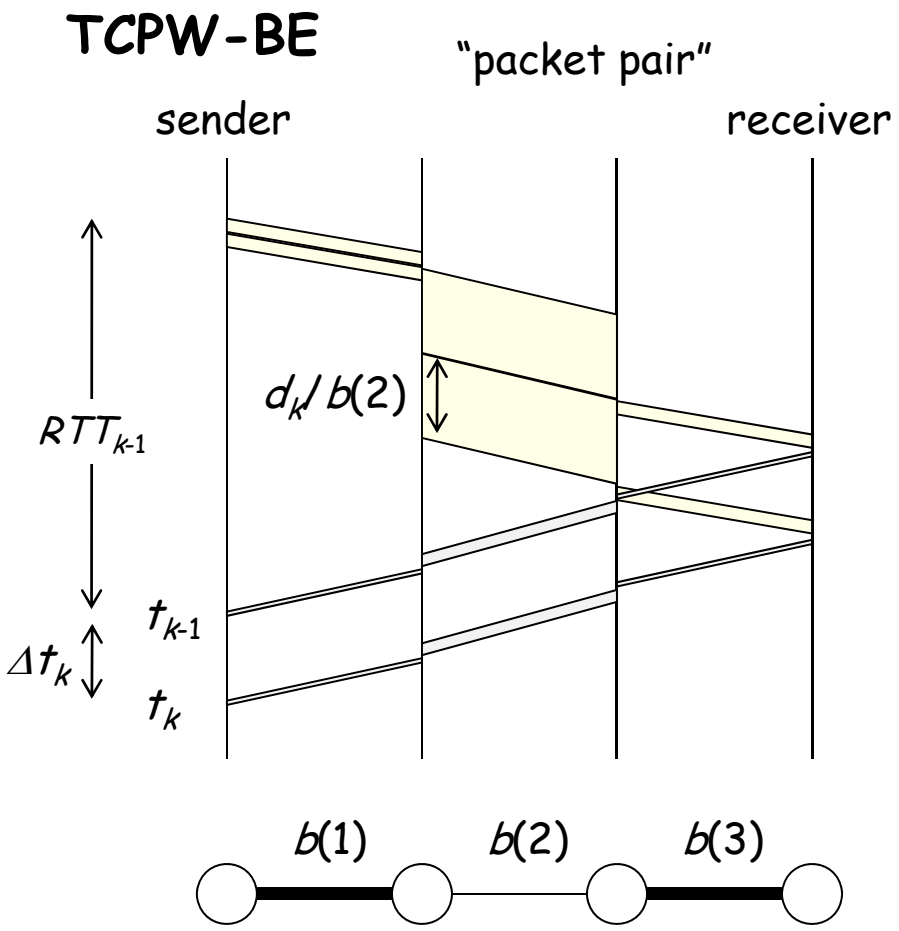
TCP-Reno の場合:

$$ssthresh = cwnd / 2$$

- FSEの求め方に応じて複数バージョン

C.Casetti et al: "TCP Westwood: Bandwidth Estimation for Enhanced Transport over Wireless Links", ACM MOBICOM 2001.

Bandwidth Share Estimation



Bandwidth share: $b = \min_j \left(b(j) \right)$

t_k: ack arrival time of the k-th packet

d_k: size of the k-th packet

$$\Delta t_k = t_k - t_{k-1} \approx \frac{d_k}{b}$$

$$b_k \approx \frac{d_k}{\Delta t_k}$$

移動平均を行い平滑化： $\hat{b}_k \to FSE$

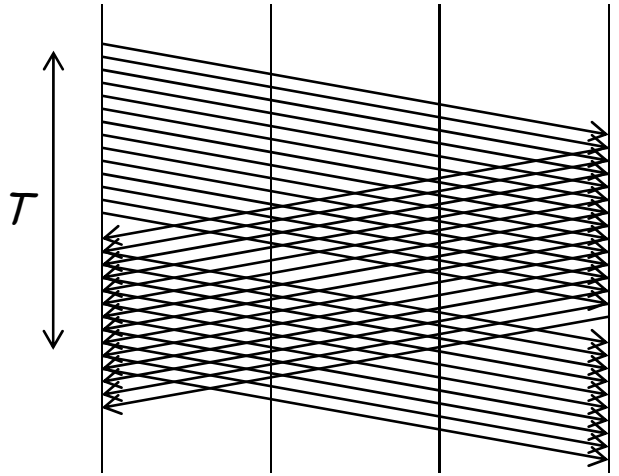
C.Casetti et al: "TCP Westwood: Bandwidth Estimation for Enhanced Transport over Wireless Links", ACM MOBICOM 2001.

Rate Estimation

(参考) TCP-Vegas

$$diff = \left(\frac{cwnd}{RTT_{\min}} - \frac{cwnd}{RTT} \right) \cdot RTT_{\min}$$

expect rate　　actual rate



TCPW-RE:

$$RE_k = \frac{\sum_{t_j > t_k - T} d_j}{T}$$

個数(cwnd) ⇒ バイト数(Σd_k)

RTT⇒ 観測時間(T= $\Sigma \Delta t_k$)

$$\hat{RE}_k \approx \frac{\sum d_k}{\sum \Delta t_k}$$

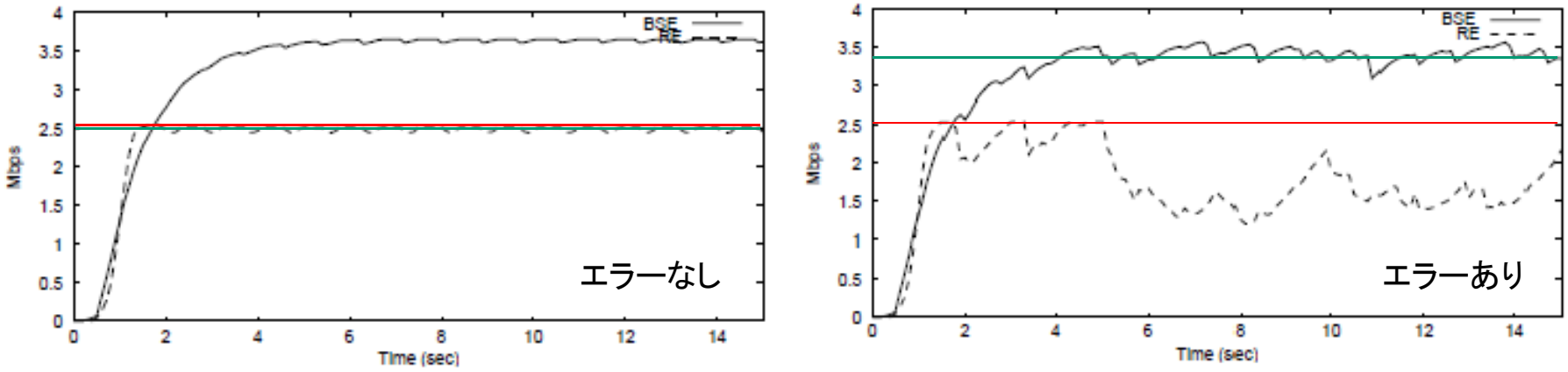
移動平均を行い平滑化： $\hat{RE}_k \rightarrow FSE$

$T = n \cdot RTT$ (e.g. n=4)

M.Gerla et al: "TCP westwood with adaptive bandwidth estimation to improve …", Comp. & Comm., 2004.

BSEとREの比較

実線: BSE、破線: RE、赤線: fair share、緑線： キャパシティー競合フロー

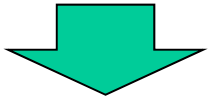


- BSEは値が高めになりやすい（バースト性のため）
- REはロスがあると値が低めになりやすい

M.Gerla et al: "TCP westwood with adaptive bandwidth estimation to improve …", Comp. & Comm., 2004.

Adaptive Bandwidth Share Estimation

- BSEは高め、REは保守的(低め)
- BSEとREの違い: サンプル間隔Tの大小



- 輻輳時はTを大きく、非輻輳時はTを小さく

TCPW-ABSE:

actual rate

$$T_k = \max\left(T_{\min},\ RTT \cdot \left(1 - \frac{\hat{T}h \cdot RTT_{\min}}{cwnd} \right) \right)$$

T_{min} : ACK到着間隔

$\hat{T}h \cdot RTT_{\min} < cwnd$ → 輻輳状態 → T_k を大きく(保守的なREへ)

多数のパケットを送っても実レートが上がらない

M.Gerla et al: "TCP westwood with adaptive bandwidth estimation to improve …", Comp. & Comm., 2004.

ハイブリッドTFRC

TFRC (RFC 3448)

- TCP-Friendly Rate Control

$$R_{TFRC} = \frac{PS}{RTT\sqrt{\frac{2bp}{3}} + 4 \cdot RTT \cdot 3\sqrt{\frac{3bp}{8}} \cdot p(1+32p^2)}$$

b=1: delayed ACK (推奨)

t_{RTO} TCP再送タイムアウト

  - RTTとパケットロス率pを観測して "TCP-Renoに等価" なレートを計算
  - RTP/UDP や DCCP によるリアルタイム系メディア (音声、音楽、映像、ゲーム) 応用を想定

M.Handley et al: "TCP Friendly Rate Control (TFRC): Protocol Specification", IETF RFC 3448, 2003.

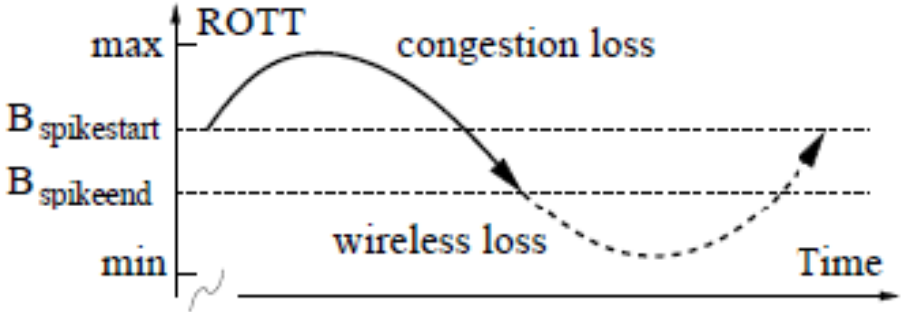
TFRC の弱点

- TCP-Renoの弱点を踏襲
  - バッファが BDP よりも小さい場合、頻繁に空き帯域が生じる
  - エラーが発生しやすい環境 (無線環境等) で、不必要にレートを下げる ⇒ LDA

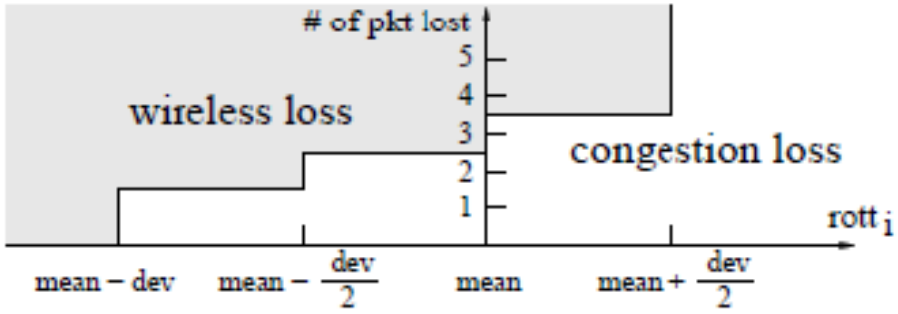
LDA (1)

"TFRC Wireless"

- Loss Differentiation Algorithm
  - 輻輳ロスとランダムエラーロスの区別



Spike algorithm



ZigZag algorithm

$$B_{spikestart} = rott_{\min} + \alpha \cdot (rott_{\max} - rott_{\min})$$

$$B_{spikeend} = rott_{\min} + \beta \cdot (rott_{\max} - rott_{\min})$$

$$\alpha = 1/2, \ \beta = 1/3$$

ROTT: Relative One-way Trip Time

S.Cen et al: "End-to-end differentiation of congestion and wireless losses", IEEE/ACM Trans. Networking, 2003.

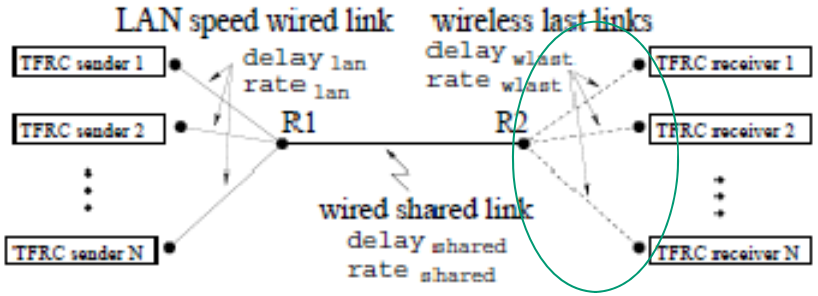
LDA (2)

"TFRC Wireless"

表のメトリック（上から）
- スループット
- 輻輳ロス
- ランダムロスと判断された輻輳ロス
- 輻輳ロスと判断されたランダムロス

- シミュレーション評価

Wireless last hop

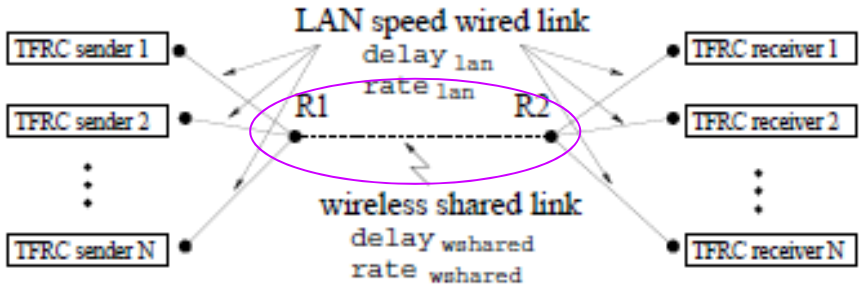


PERFORMANCE FOR WIRELESS LAST HOP, 1 FLOW

	TCP	TFRC	Omni	Biaz	mBiaz	Spike	ZigZag
thput	55	84	99	99	99	99	98
cong.	0.8	0.2	2.3	2.3	2.3	0.4	0.3
M_c	0	0	0	0.0	0.0	0.0	0.0
M_w	100	100	0	6.3	6.6	58	66

LDA

Wireless backbone



PERFORMANCE FOR WIRELESS BACKBONE, 1 FLOW

	TCP	TFRC	Omni	Biaz	mBiaz	Spike	ZigZag
thput	23	37	99	97	91	99	53
cong.	0.1	0.0	0.4	0.4	0.4	0.0	0.0
M_c	0	0	0	0.0	0.0	0.0	0.0
M_w	100	100	0	2.4	7.0	29	60

LDA

S.Cen et al: "End-to-end differentiation of congestion and wireless losses", IEEE/ACM Trans. Networking, 2003.

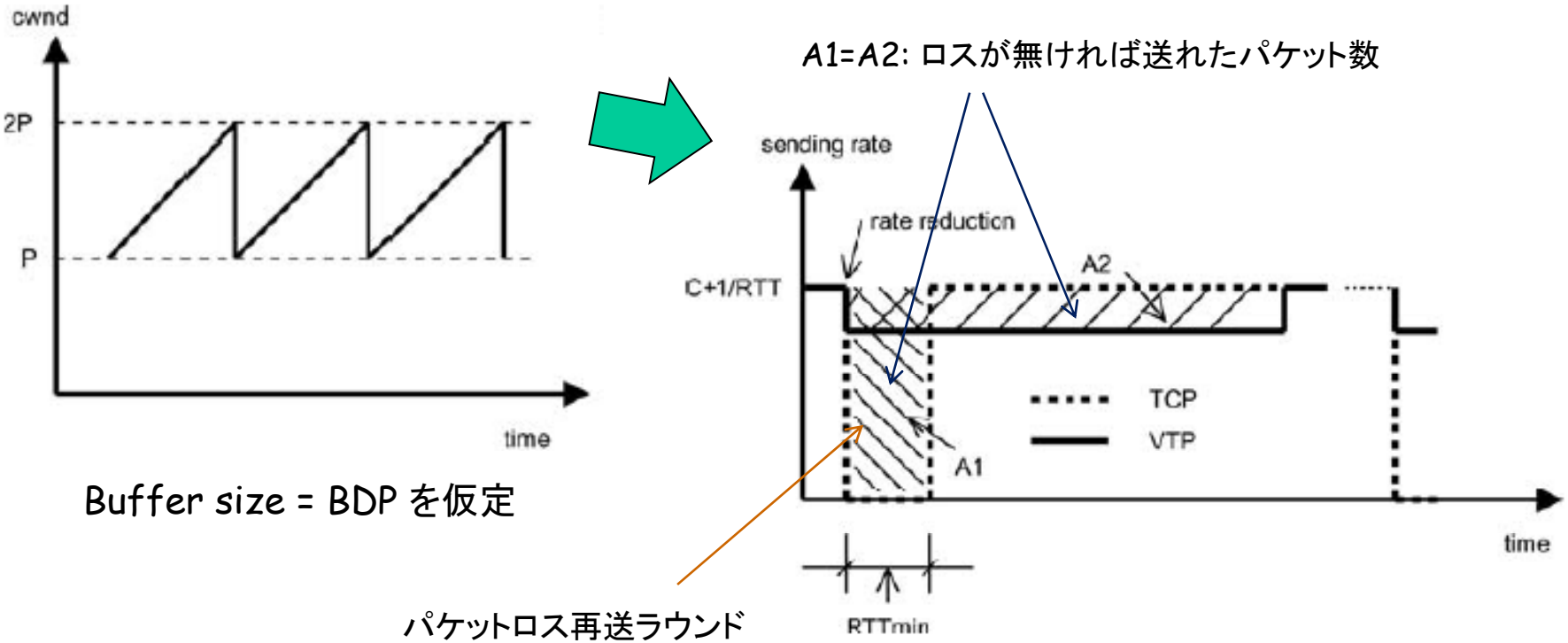
VTP (1)

- Video Transport Protocol
  - LDA（輻輳ロスとランダムロスの区別～TFRC Wireless）
  - TCPW-REと同様のレート推定（Achieved Rateの活用）
  - TCP-Renoのウィンドウ制御のエミュレート（レガシーTCPとの親和性）

G.Yang et al: "Smooth and efficient real-time video transport in the presence of wireless errors", ACM Trans. MCCAP, 2006.

VTP (2)

- VTPの概略



G.Yang et al: "Smooth and efficient real-time video transport in the presence of wireless errors", ACM Trans. MCCAP, 2006.

VTP (3)

- VTPのウィンドウ制御
  - 初期値: TCPW-REで求まるAchieved Rate
  - 更新: RTTごとに1パケット追加

$$R_0 = \text{Achived Rate}$$

$$ewnd_k = R_k \times RTT_k$$

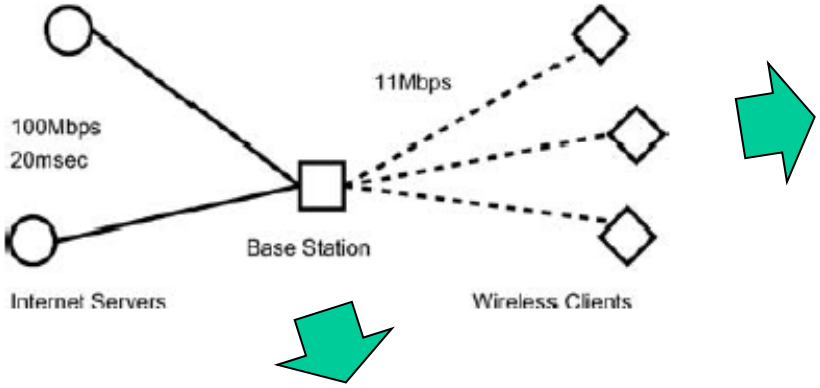
RTTが増加しなければ $R_{k+1} = R_k + \frac{1}{RTT_k}$

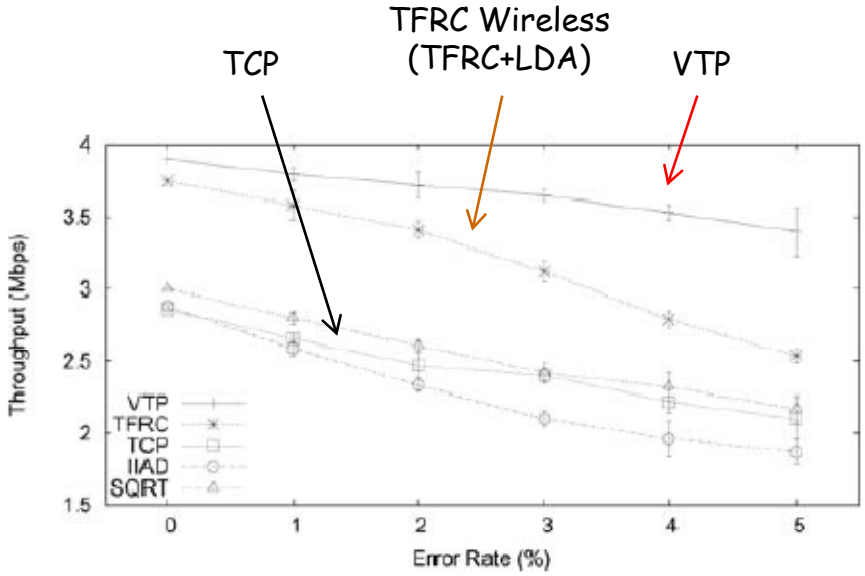
$$R_{k+1} = \frac{ewnd_{k+1}}{RTT_{k+1}} = \frac{ewnd_{k+1}}{RTT_k + \Delta RTT_k} = \frac{R_k \times RTT_k + 1}{RTT_k + (RTT_k - RTT_{k-1})}$$

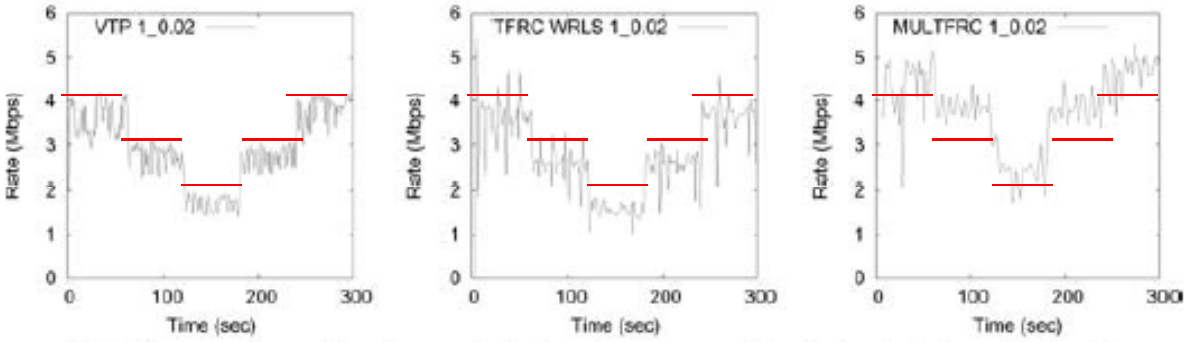
G.Yang et al: "Smooth and efficient real-time video transport in the presence of wireless errors", ACM Trans. MCCAP, 2006.

VTP (4)

- シミュレーション評価







(b) 2% errors (avg. time in good state = 1 sec, avg. time in bad state = 0.02 sec).

MULTFRC: 複数のTFRCコネクションを張り、帯域利用効率の改善を図る

G.Yang et al: "Smooth and efficient real-time video transport in the presence of wireless errors", ACM Trans. MCCAP, 2006.

VTP (5)

- VTPの弱点
  - Buffer size = BDP のみ想定
  - Buffer size < BDP の場合に生じる空き帯域有効利用の課題は想定外

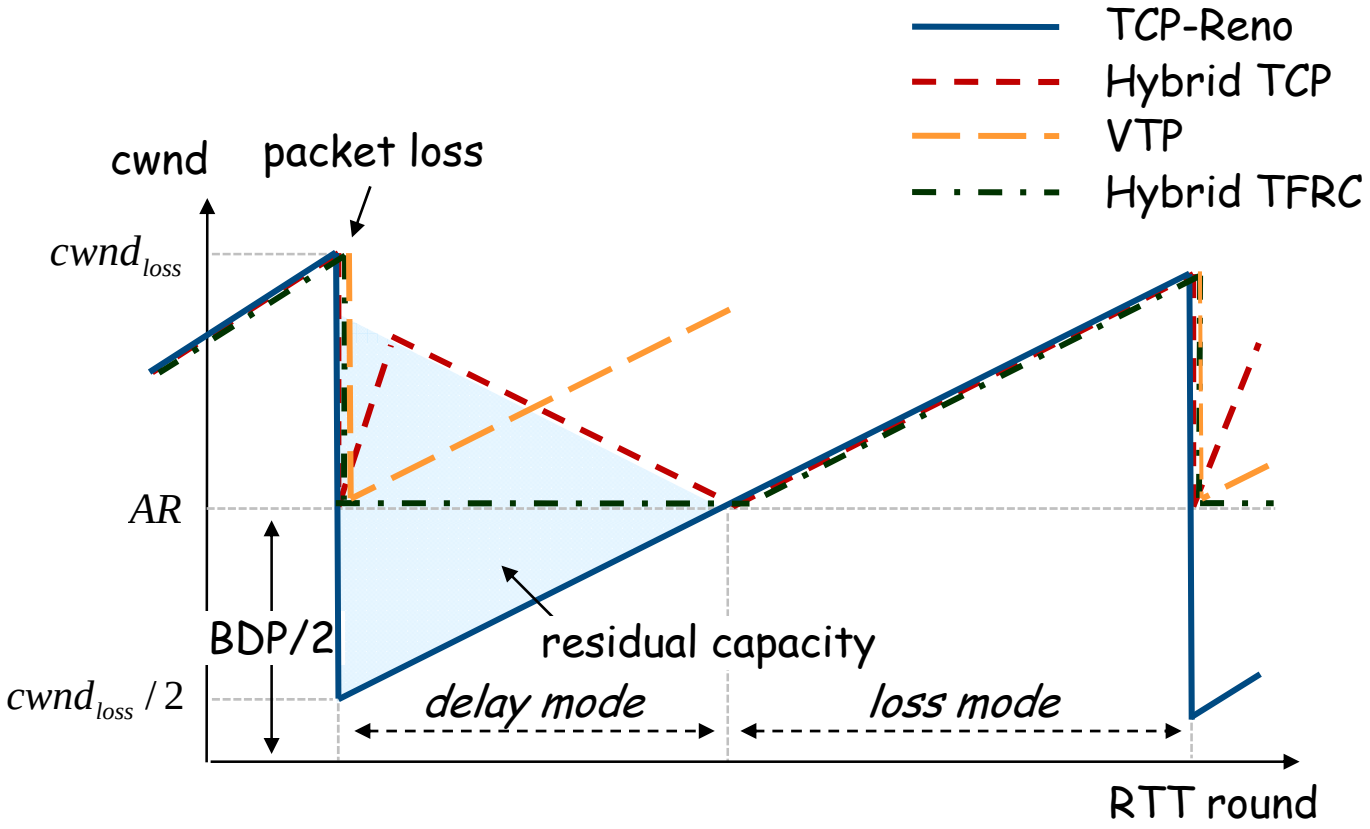
ハイブリッドTFRC (1)

- ハイブリッドTCPのTFRC版
  - RTTが増加しない場合はAchieved Rateで送信（効率性の改善）
  - RTTが増加したらVTPと同様のウィンドウ制御で送信（親和性の確保）
  - 小バッファサイズへの対応 ⇒ RTTの低減

G.Yang et al: "Smooth and efficient real-time video transport in the presence of wireless errors", ACM Trans. MCCAP, 2006.

ハイブリッドTFRC (2)

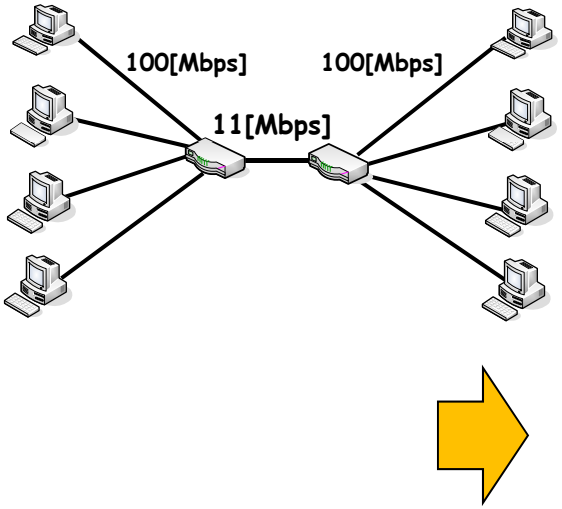
- ハイブリッドTFRCの動作 (Reno競合時)

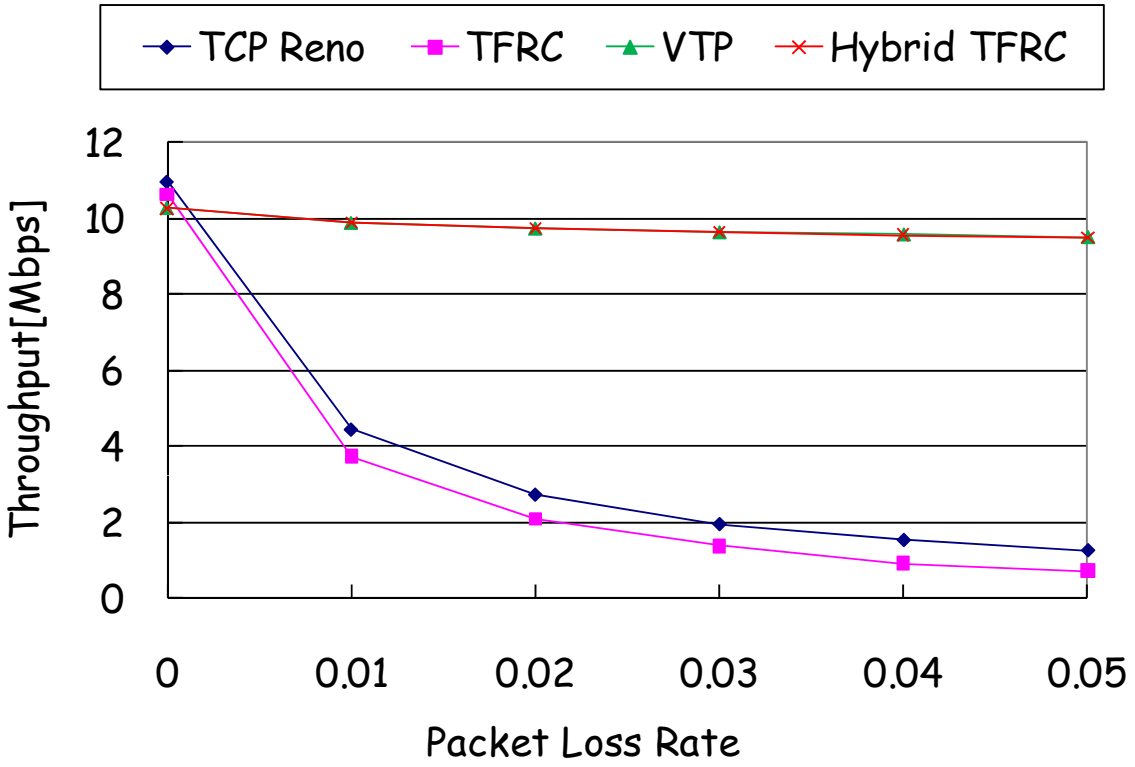


ハイブリッドTFRC (3)

- シミュレーション評価(1)

Buffer size = BDP

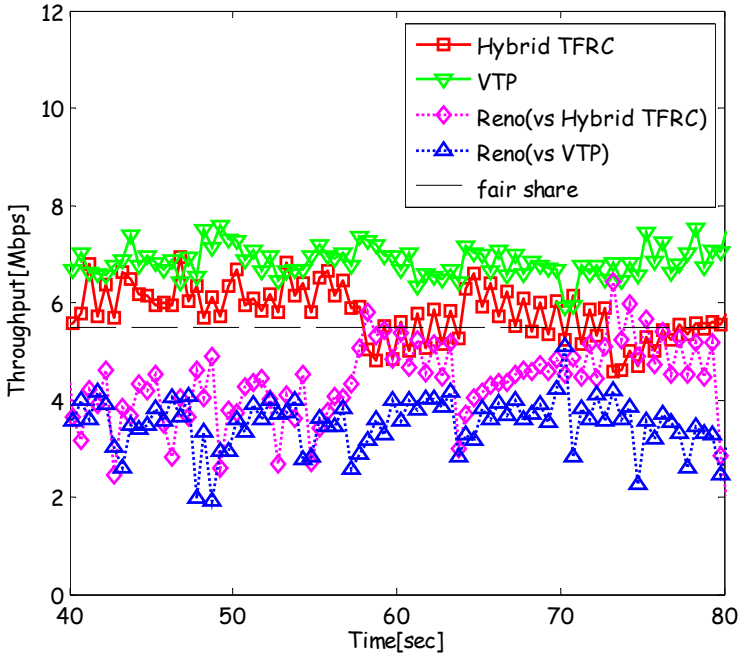




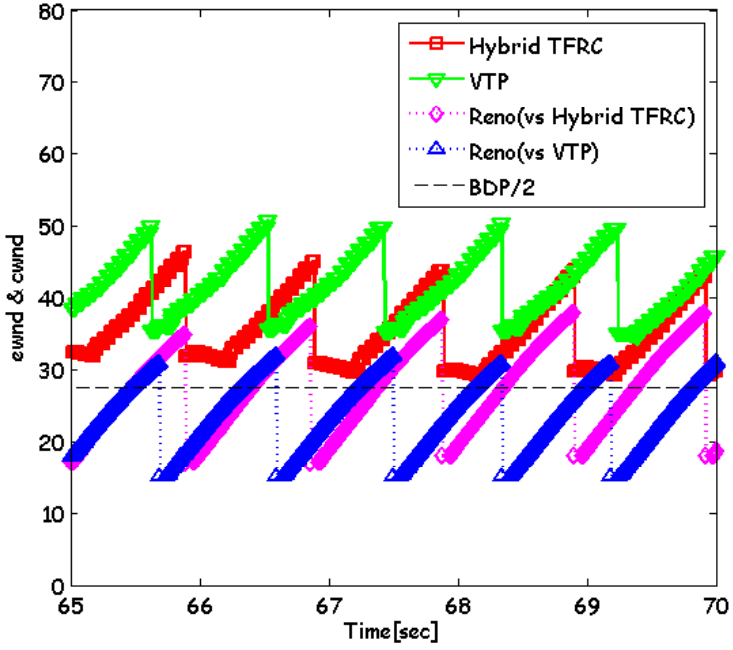
ハイブリッドTFRC (4)

- シミュレーション評価(2)

Buffer size = BDP/2



Throughput



ewnd & cwnd

TFWC と Relentless CC

TCP-Friendly Window-based Congestion Control (1)

- 従来のTFRCの弱点
  - Friendliness: 小バッファ時に不公平が生じる (競合TCPを追い出す)
  - Smoothness: 送信レートが大きく振動することがある (RTTが安定しないため)
  - Smoothness: RTTが小さ過ぎるとレート計算が安定しない
  - Responsiveness: フローのON/OFFに対する応答性は？
- Rate-based ⇒ Window-based

S.H.Choi et al: "Designing TCP-Friendly Window-based Congestion Control …", PFLDNet 2009.

TCP-Friendly Window-based Congestion Control (2)

- Ack Vector:
  - 受信者は、受信したパケットのシーケンス番号の並び(Ack Vector)を送信者に返す
  - 送信者は、Ack Vectorからパケットロスの平均発生間隔を計算し(1/p)、cwndを更新する

$$W_{TWRC} = \frac{1}{\sqrt{\frac{2p}{3}} + 12\sqrt{\frac{3p}{8}} \cdot p(1+32p^2)} \quad \Leftrightarrow \quad R_{TFRC} = \frac{PS}{RTT\sqrt{\frac{2p}{3}} + 12 \cdot RTT\sqrt{\frac{3p}{8}} \cdot p(1+32p^2)}$$

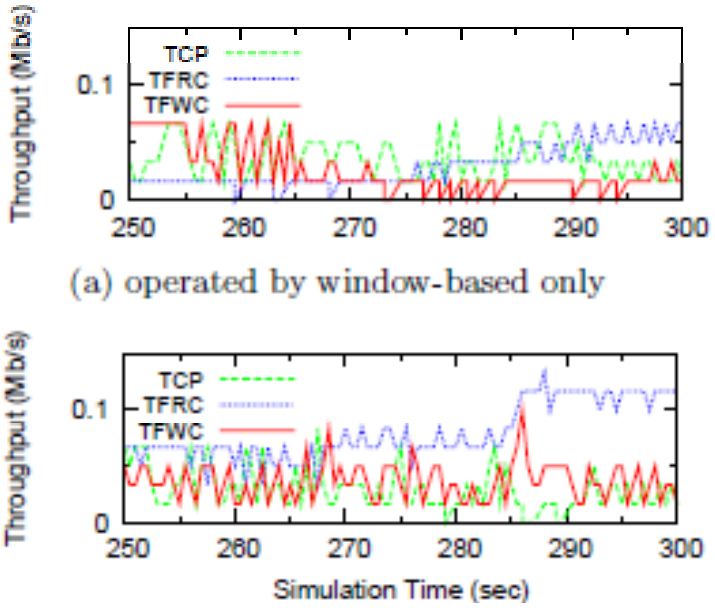
  - RTTが含まれないので数値的に安定

S.H.Choi et al: "Designing TCP-Friendly Window-based Congestion Control …", PFLDNet 2009.

TCP-Friendly Window-based Congestion Control (3)

- Hybrid Window & Rate
  - cwndが小さくなり過ぎるとタイムアウトが発生する

  - cwndが小さくなり過ぎた場合はTFRCを動作させる



(a) operated by window-based only

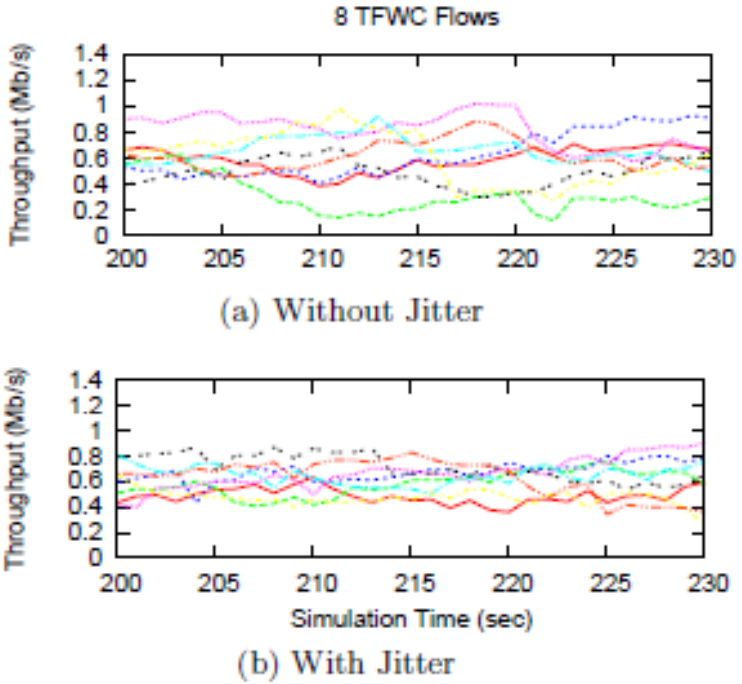
(b) operated by window/rate-based

S.H.Choi et al: "Designing TCP-Friendly Window-based Congestion Control …", PFLDNet 2009.

TCP-Friendly Window-based Congestion Control (4)

- Phase effect対策
  - drop tailでは競合フローのロス率が大きく異なることがある（特定フローのロスが"同期"）
  - 対策: RED (Random Early Drop) によるランダム廃棄

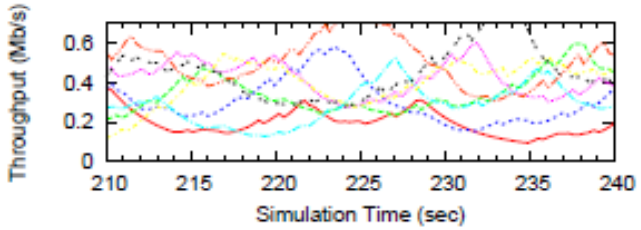
  - 類推: cwndに微小な"乱数"を加える



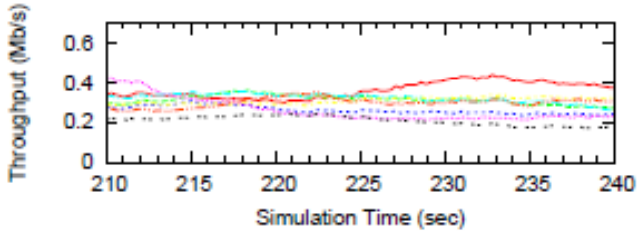
S.H.Choi et al: "Designing TCP-Friendly Window-based Congestion Control …", PFLDNet 2009.

TCP-Friendly Window-based Congestion Control (5)

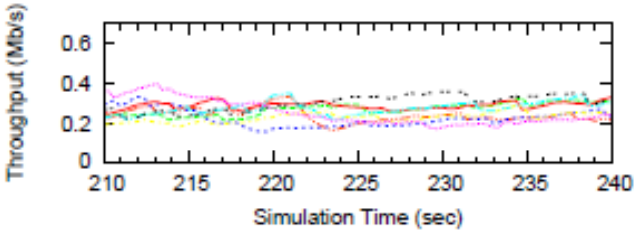
- Smoothness



(a) TCP's Abrupt Sending Rate Change

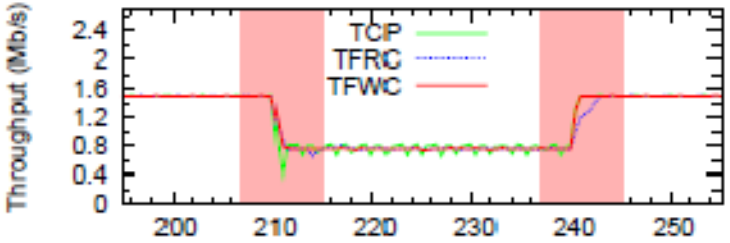


(b) TFRC Smoothness

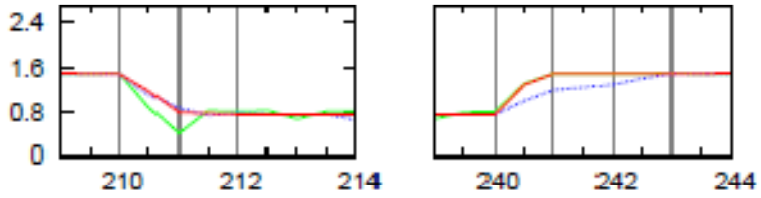


(c) TFWC Smoothness

- Responsiveness



(a) Impulse Response



(b) Zoom-In (high-lighted area)

S.H.Choi et al: "Designing TCP-Friendly Window-based Congestion Control …", PFLDNet 2009.

Relentless Congestion Control (1)

- Packet conservation
  - パケットが取り出されたら、新しいパケットを送信する (SIGCOMM 1988)

  - パケットがロスしたら、そのロス数分だけcwndを減少 (1/2は絶対にしない)

$$cwnd = cwnd - \text{loss counts}$$

$$ssthresh = cwnd - \text{loss counts}$$

  - 定常状態まではSlow Startで増加

M.Mathis: "Relentless Congestion Control", PFLDNet 2009.

Relentless Congestion Control (2)

- Scalability
  - 従来のAIMDは、パケットロスの時間間隔が帯域幅に応じて変化する
  - Relentless CCは、帯域幅に依存せず、パケットロス間隔を一定値にする (e.g. 3*RTT)
    - CUBIC-TCPの踏襲
  - End-to-end ⇒ Network assist (baseline AQM)

M.Mathis: "Relentless Congestion Control", PFLDNet 2009.

Relentless Congestion Control (3)

- Relentless CC + Baseline AQM (Active Queue Management)
  - ロスが無ければcwnd=cwnd+1
  - 3*RTT毎にパケットを廃棄 (by AQM)
  - cwndからロス数を引いて更新
  - 以上の繰り返し

- TCP-unfriendly paradigm

M.Mathis: "Relentless Congestion Control", PFLDNet 2009.