

# Quality-of-Service Provisioning System for Multimedia Transmission in IEEE 802.11 Wireless LANs

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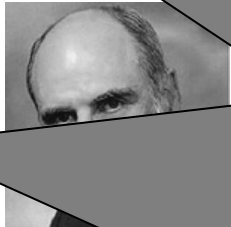
# The next wave of the Internet

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The Net's founders predict its future:

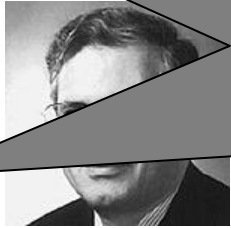


"Nomadic computing, providing access when you're on the road so that the Internet services you see when you're in any place else are no different than what you have back in your office."  
--Leonard Kleinrock



**Towards Multimedia Oriented  
Mobile Systems and providing  
"Anytime Anywhere Anyform"  
Information Systems**

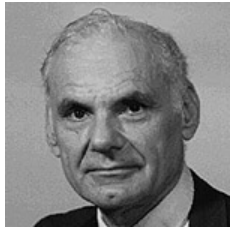
"If you have a question, you can find the answer and pull it out of the ether."  
--Vinton Cerf



"The speed of the Internet will increase exponentially."

"The speed of the Internet will increase exponentially."

"You're going to see a lot of new applications that we're going to see."  
--Robert Kahn



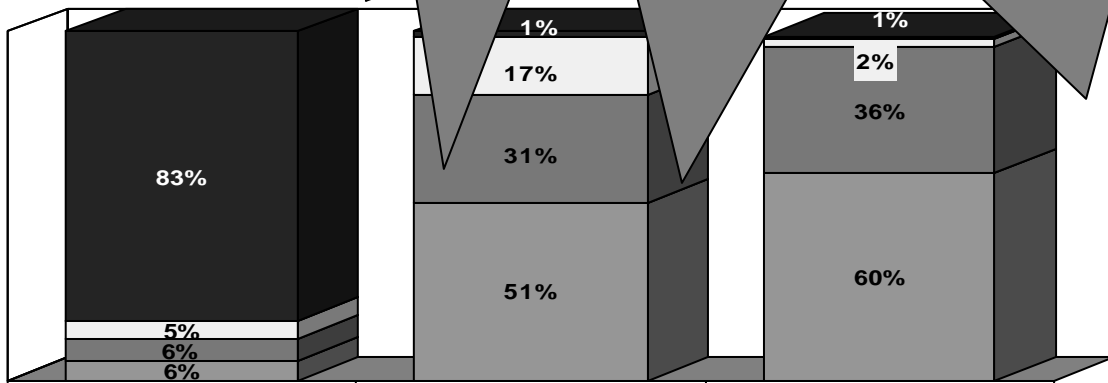
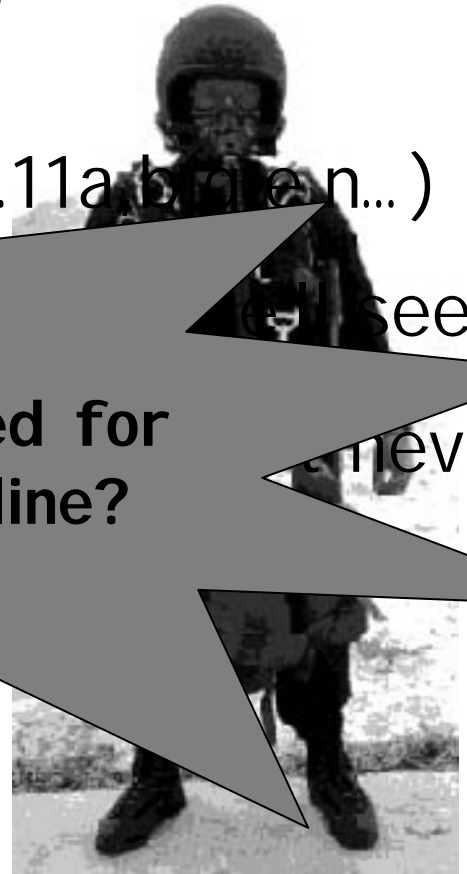
"The Internet will become the peering network for the world's telecom traffic. Voice and video will transfer over to it in the next five to 10 years. Clearly, you're going to have video on demand, radio or TV, that can have millions of different sources or special subjects that (small numbers) care about."  
--Lawrence Roberts

# Status of specified wireless networking technologies

IEEE 802.11 (wired networking) (802.11a/b/g/n...)

Bluetooth (I'll see)

What applications are suited for current status and time line?

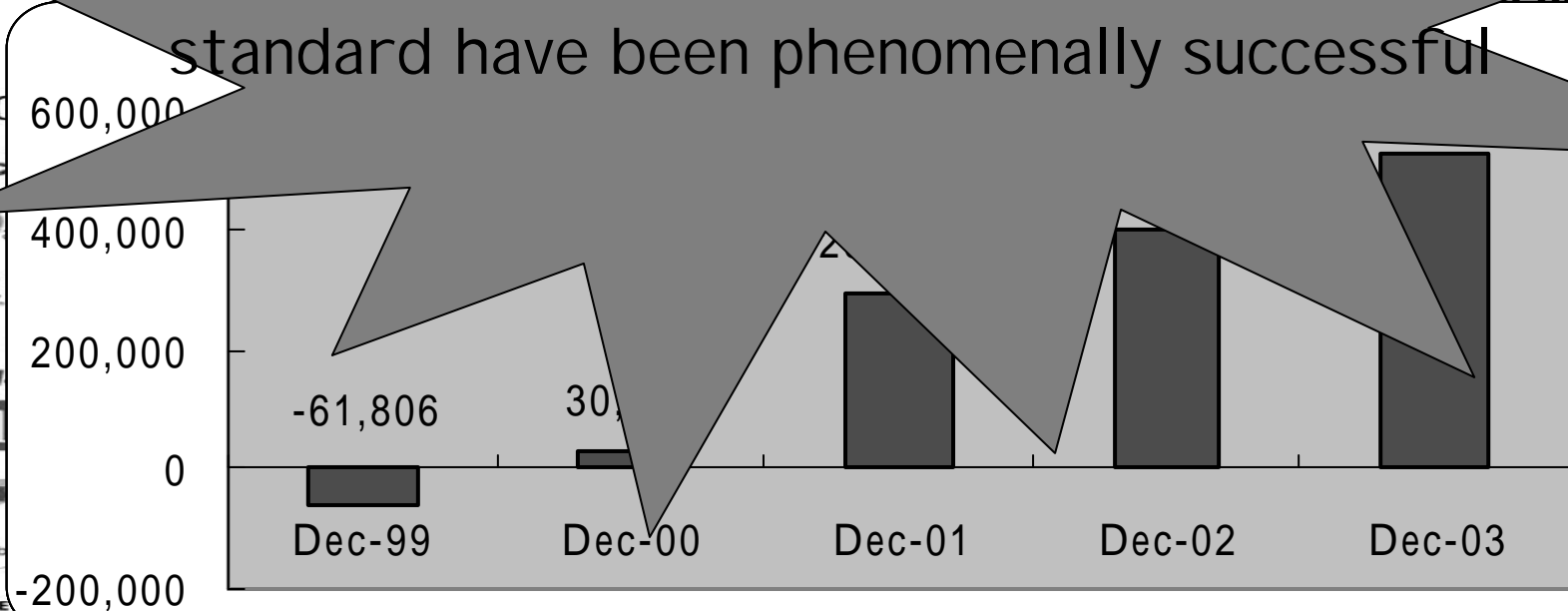


- Use today
- Will use within 18 mos.
- Will not use within 18 mos.
- Don't know

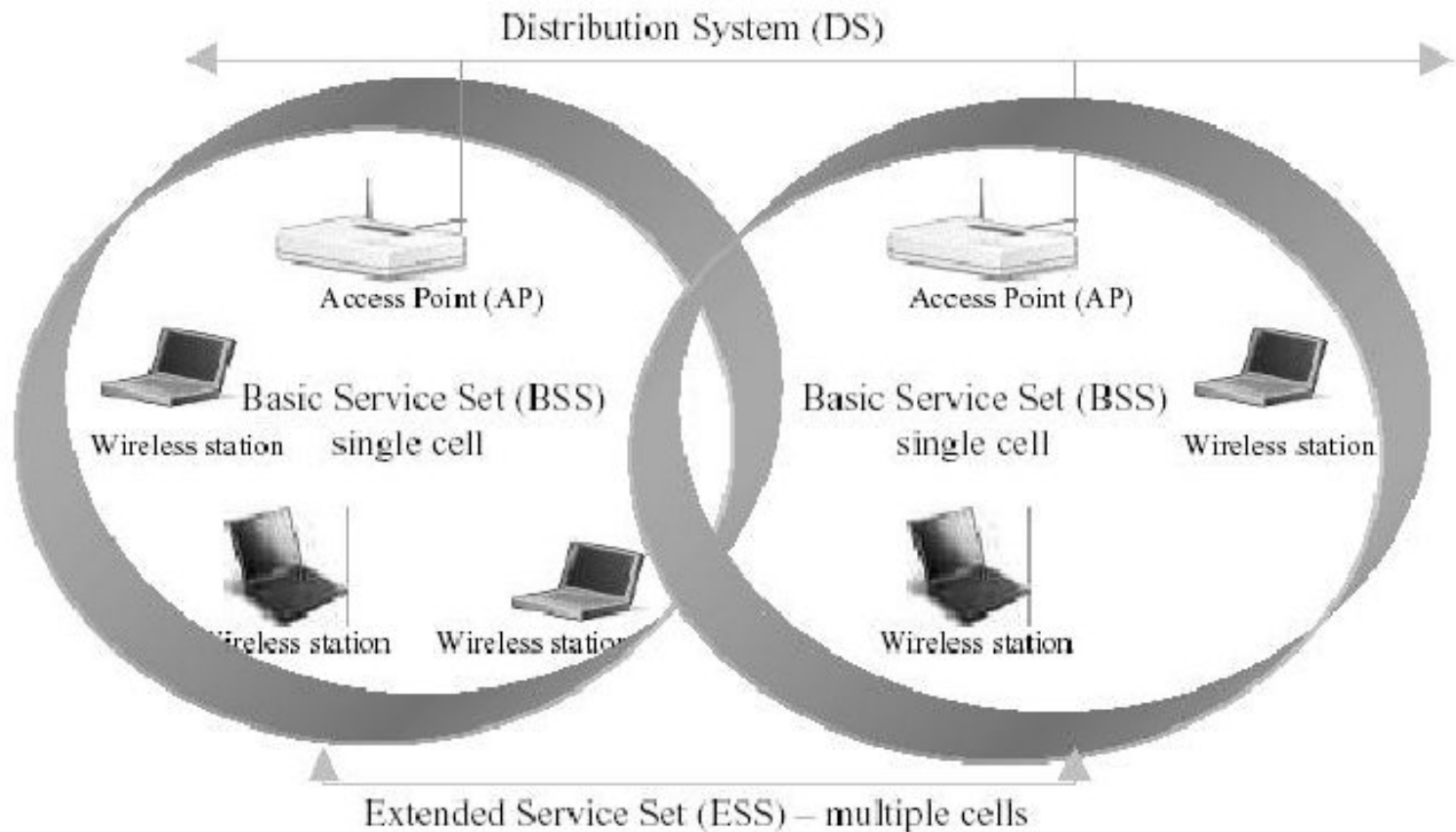
# IEEE 802.11 Market

- ❑ Emergence of the Wireless ISP
- ❑ Currently, 71,500 hotspots deployed worldwide, expected to grow to around 477,000 by 2007 (Frost & Sullivan Group)
- ❑ Intel launched its Centrino wireless platform in March 2003. The next generation of Centrino is expected to be introduced in 2005 and Napa in 2006.

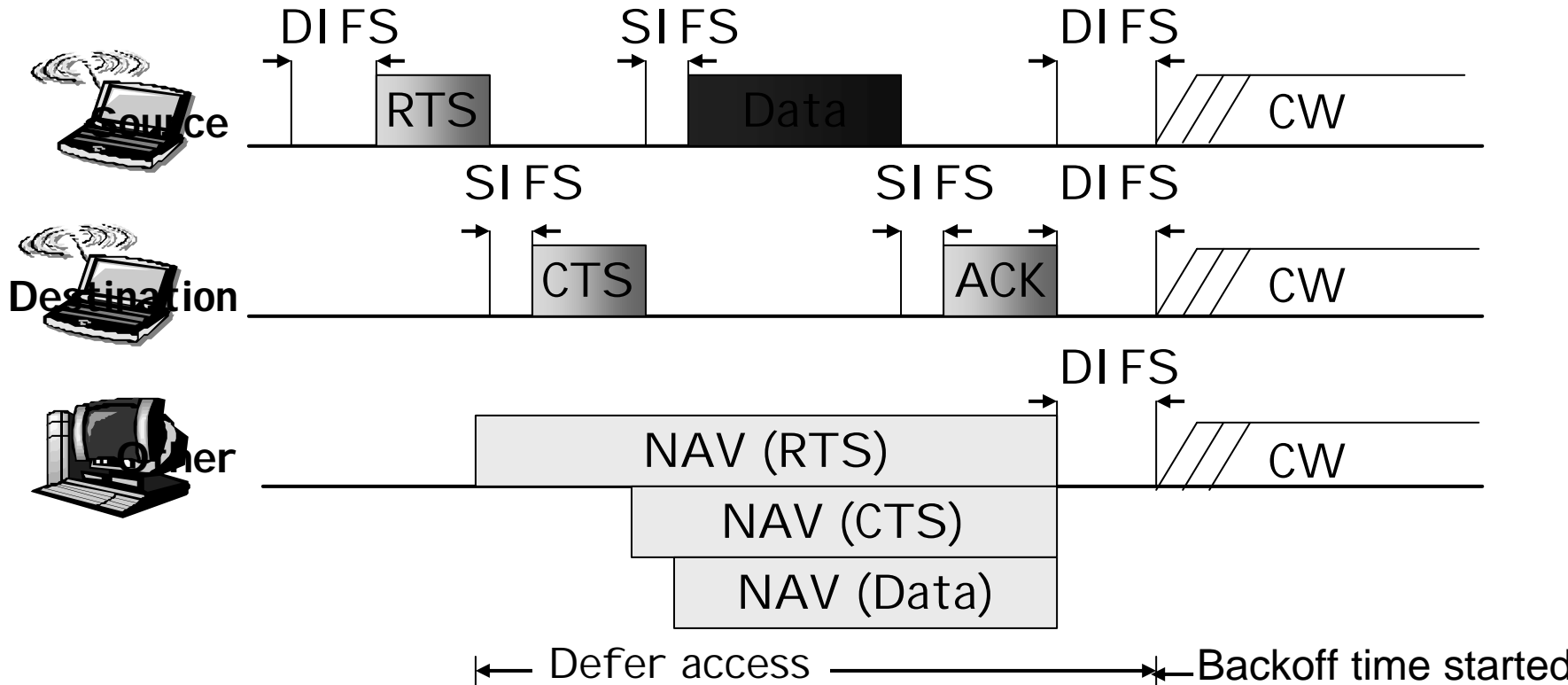
Wireless LANs based on the IEEE 802.11x standard have been phenomenally successful



# Generic 802.11 WLAN Architecture

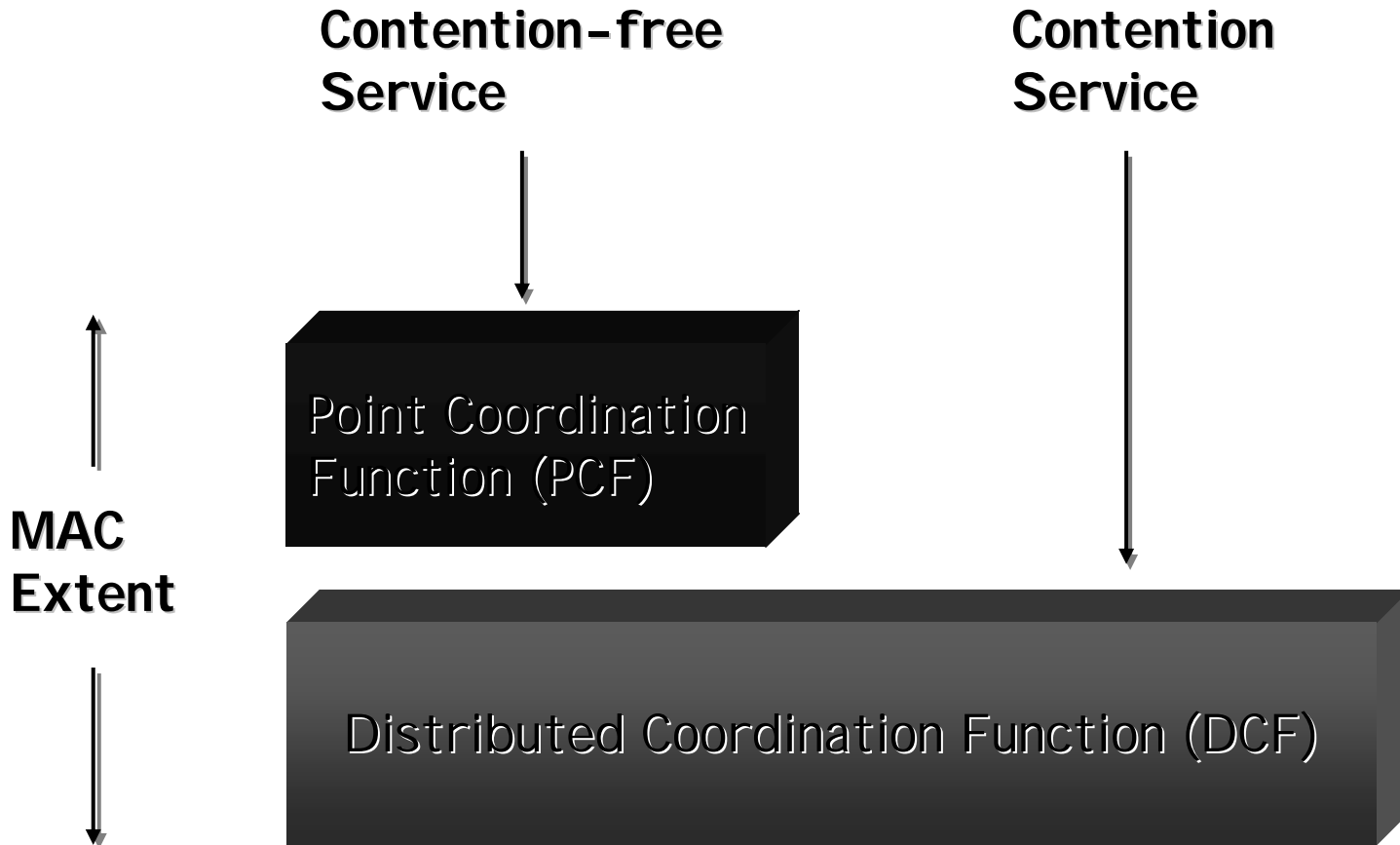


# DCF (CSMA/CA)

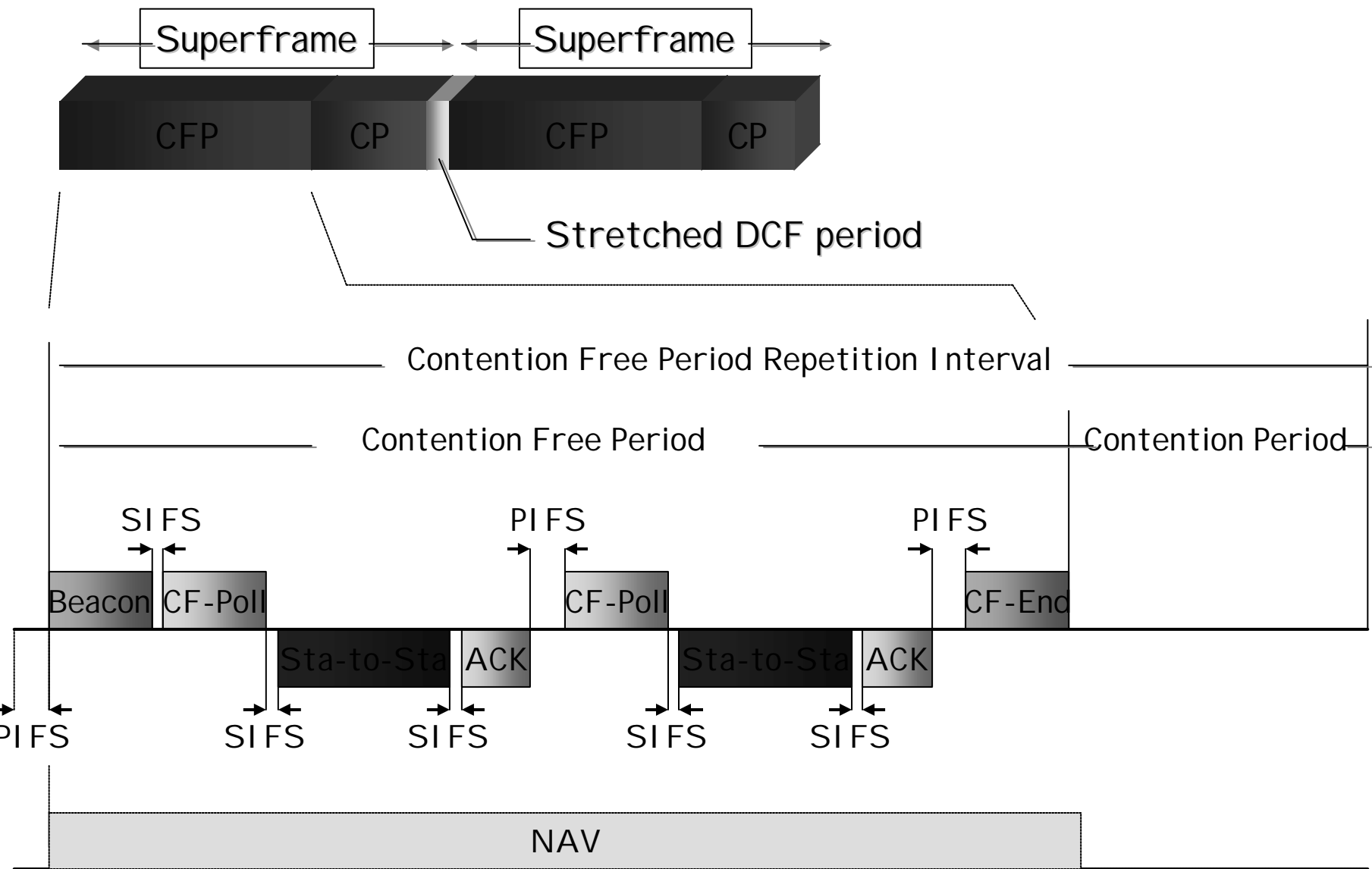


backoff time:  $\left[ \text{ranf}() \cdot 2^{2+i} \right] \cdot \text{Slot\_Time}$

# MAC Architecture



# PCF





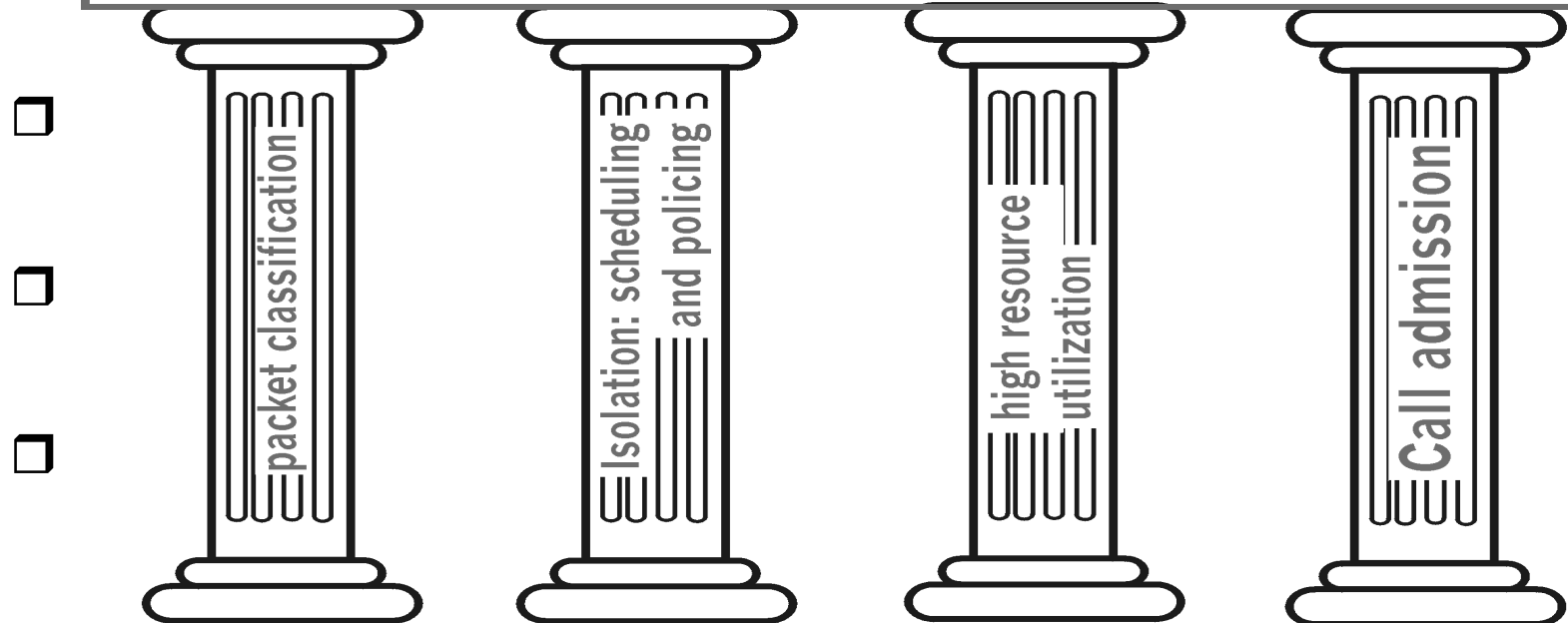
# Motivation

- ❑ Packet-switched solutions that take advantage of silences in a given voice call by multiplexing bits from other calls are more bandwidth-efficient than circuit-switched solutions
- ❑ In wireless networks, the use of packet-switched techniques for carrying multimedia traffic in 802.11 WLANs are indeed needed
- ❑ PCF mode provides a "switched connection-oriented" service, which is well suited for telephony traffic
- ❑ In order to poll the stations an AP must maintain a polling list, which is implementation dependent

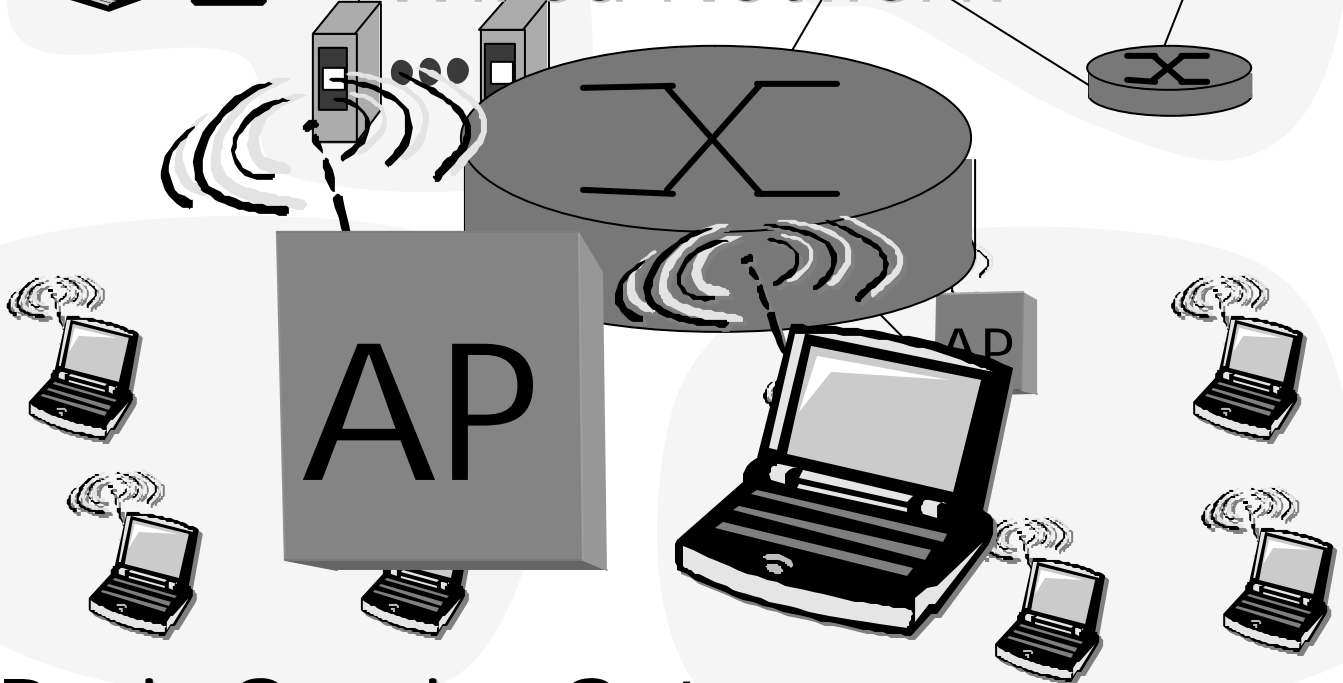
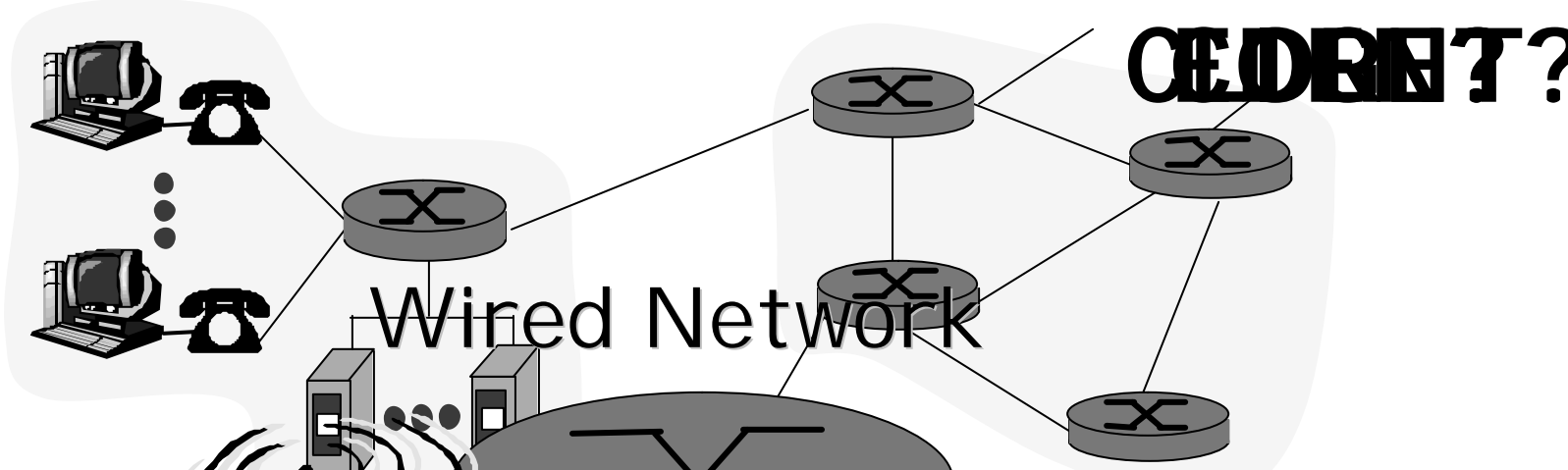
# Bandwidth management and QoS

- ❑ IETF groups are working on proposals including RSVP, Differentiated Services, and Integrated Services to provide better QoS control in IP networks

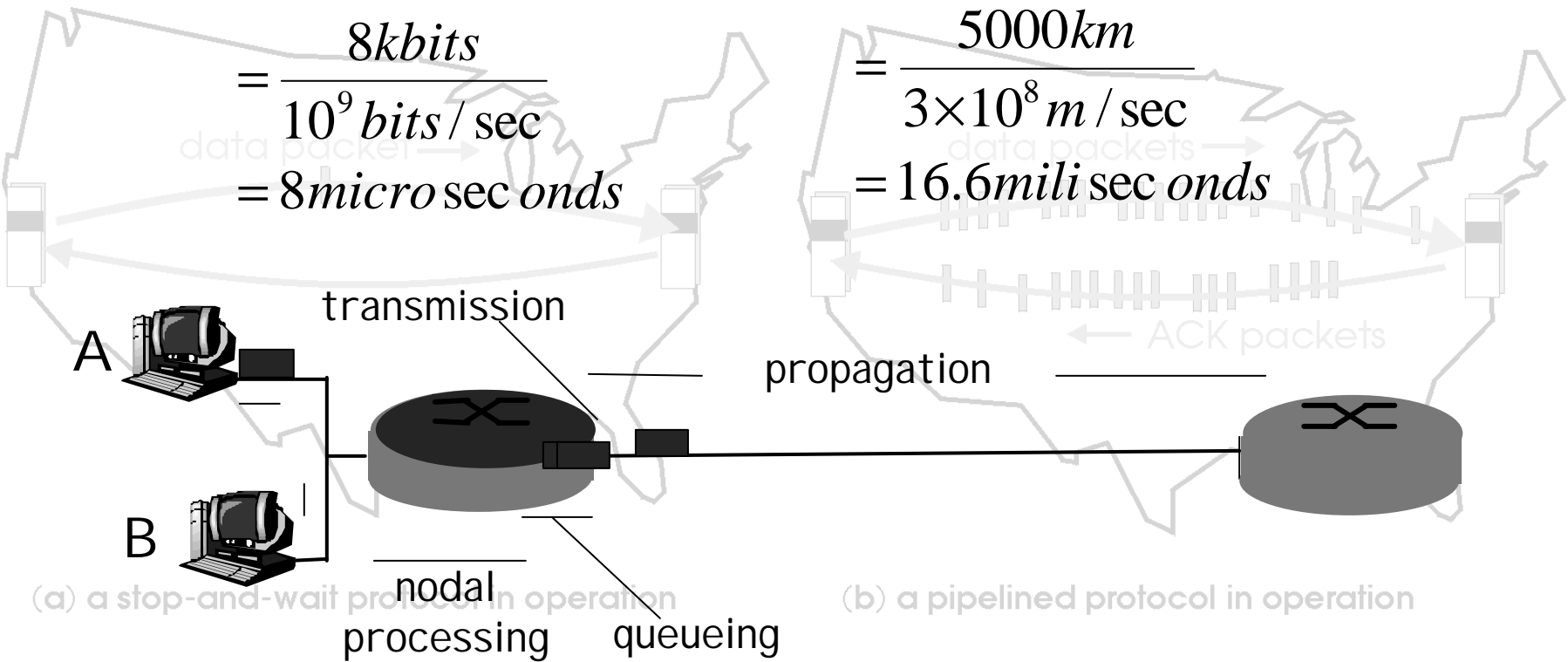
- ❑ QoS for networked applications



Should we support these functionalities in..



# Delay in packet-switched networks



$$(8 \times 10^{-6} + 16.6 \times 10^{-3}) \times 10^9 / 10^3 = 16000.608 \text{ packets}$$

# Issues in Mobile Terminal Design



# Basic Components

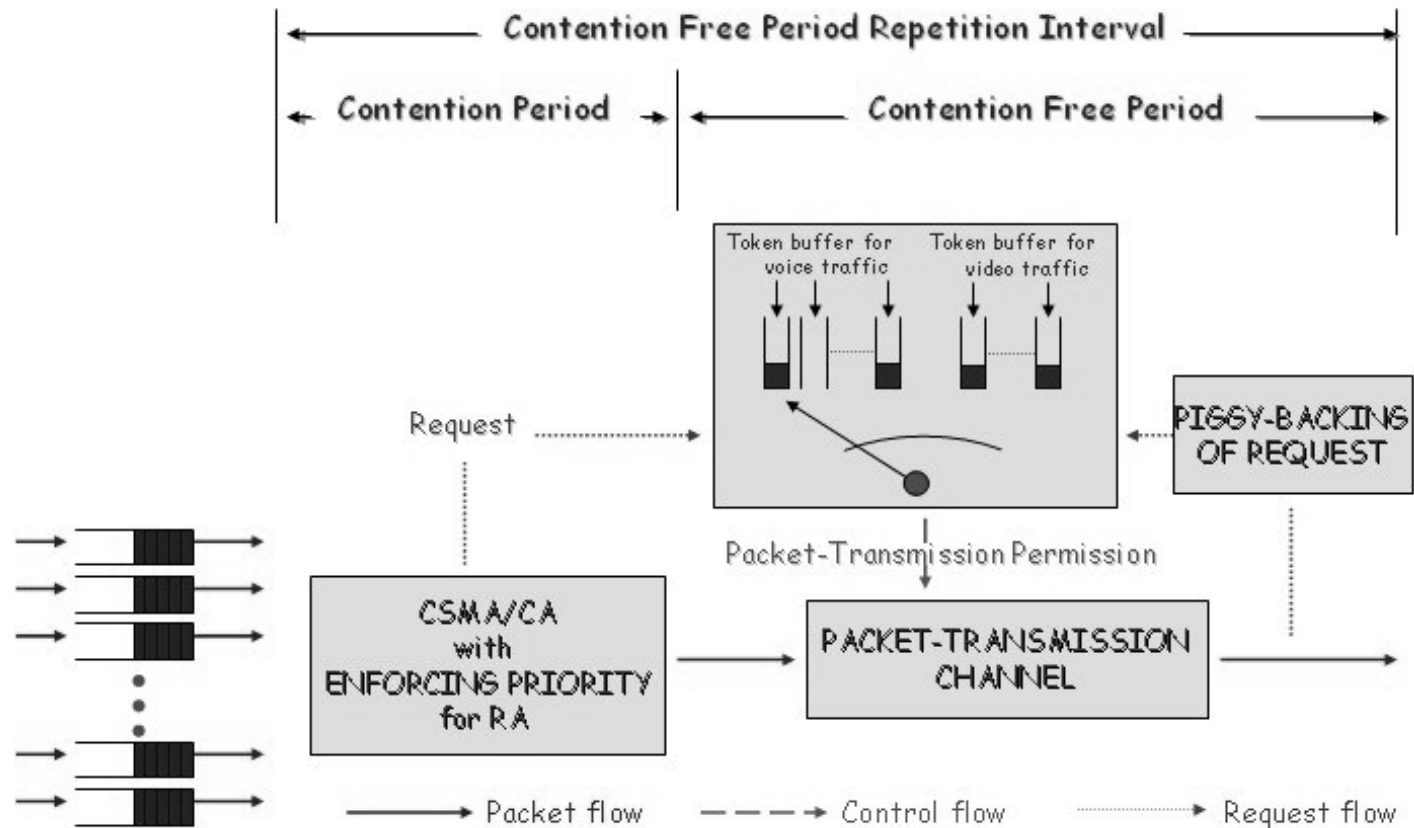
- Priority Enforcement Mechanism for Request Access collision

Packet classification

Backoff slot numbers Types of requests (k, m, n) Consecutive times (i)	1 <sup>st</sup>	2 <sup>nd</sup>	3 <sup>rd</sup>	4 <sup>th</sup>
Real-time handoff traffic (0, 1, 1)	0 - 3	0 - 7	0 - 15	0 - 31
Admitted inactivated video traffic (1, 1, 1)	4 - 7	8 - 15	16 - 31	32 - 63
Non-real-time handoff traffic New request traffic (2, 2, 1)	8 - 15	16 - 31	32 - 63	64 - 127

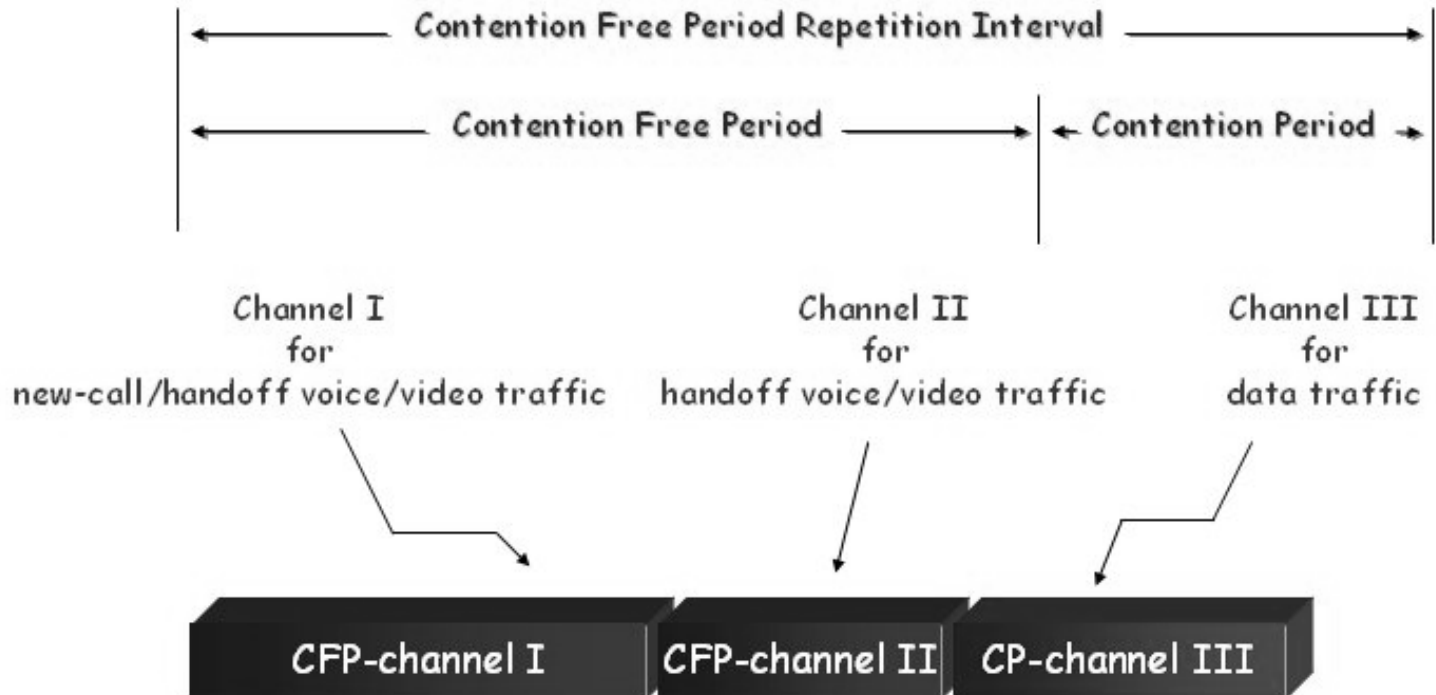
# Basic Components

- The Packet Transmit-permission Policy  
Isolation + Call admission



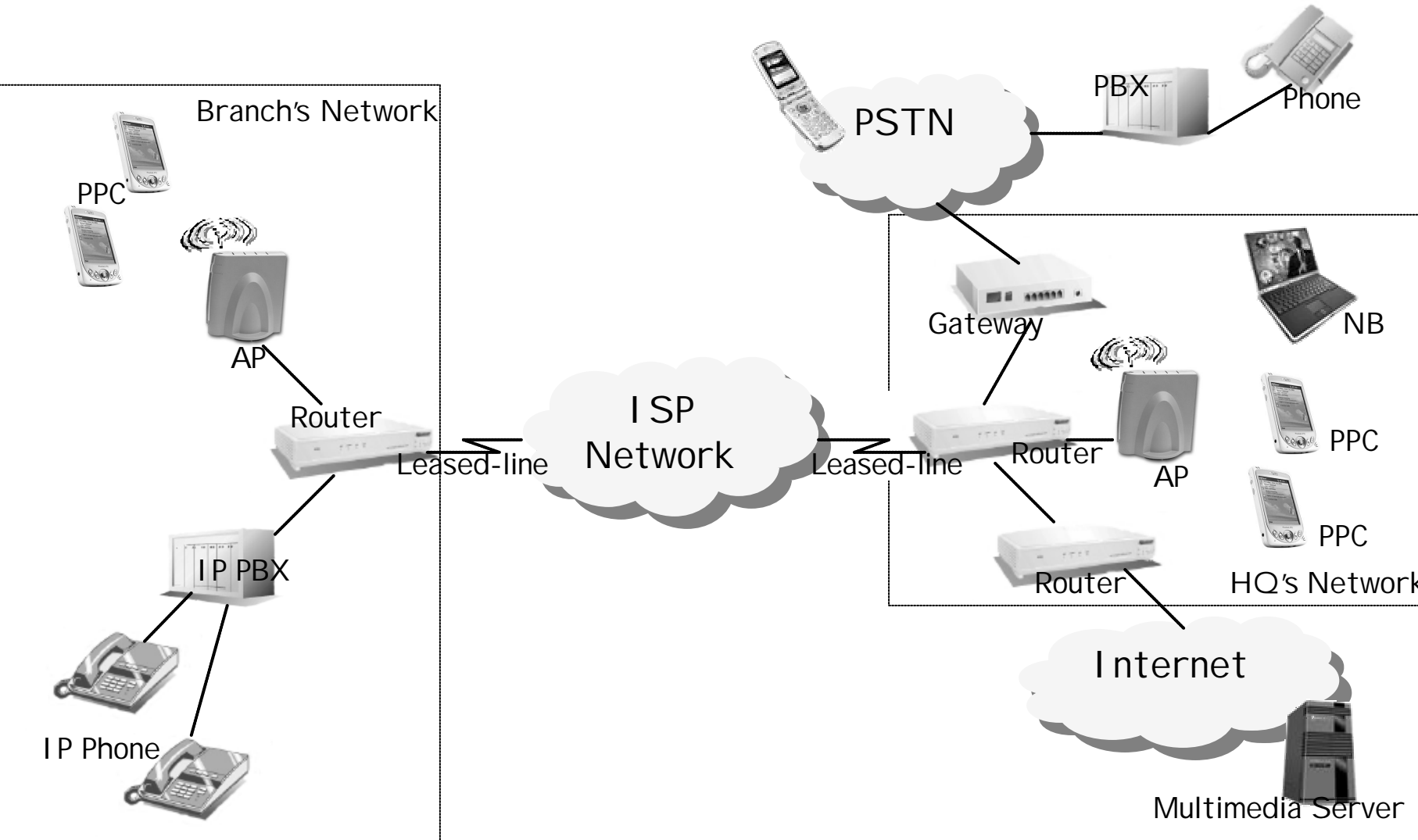
# Basic Components

- The Adaptive Bandwidth Management Strategy  
High resource utilization





# System Architecture



# Enforcing priority for RA

To support priority, we change the backoff time generation function

$$\lfloor \text{ranf}() \cdot 2^{2+i} \rfloor \rightarrow \lfloor \text{ranf}() \cdot 2^{m+i} \rfloor + k \cdot 2^{n+i}$$

Backoff slot numbers Types of requests (k, m, n)	Consecutive times (i)			
	1 <sup>st</sup>	2 <sup>nd</sup>	3 <sup>rd</sup>	4 <sup>th</sup>
<b>Real-time handoff traffic</b> (0, 1, 1)	0 – 3	0 -7	0 – 15	0 – 31
<b>Admitted inactivated video traffic</b> (1, 1, 1)	4 – 7	8 - 15	16 – 31	32 - 63
<b>Non-real-time handoff traffic</b> <b>New request traffic</b> (2, 2, 1)	8 – 15	16 - 31	32- 63	64 – 127

# Backoff Alg. in Congested Scenarios

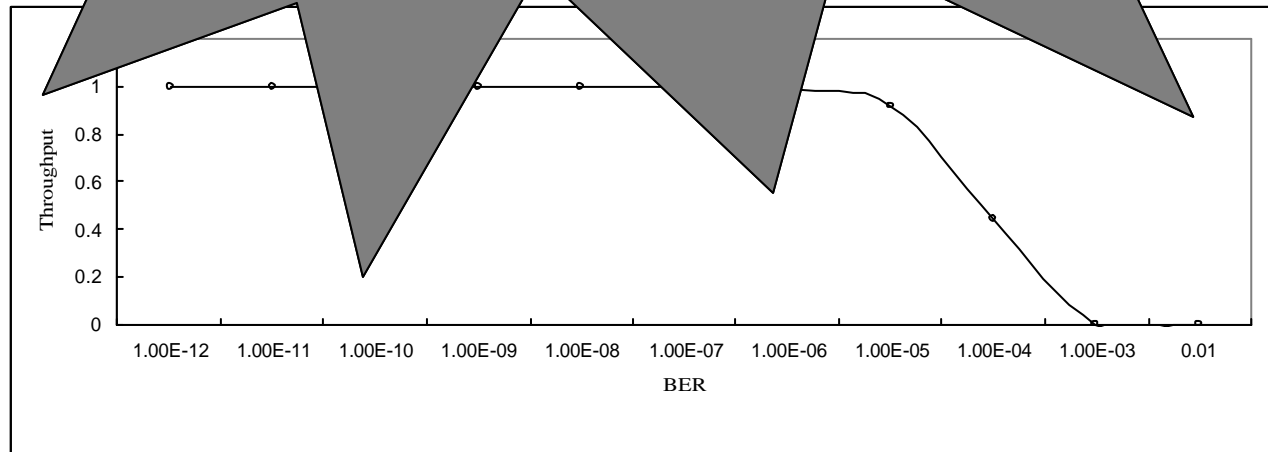
- ❑ The usage of backoff algorithm results in long access delays when the load is light because of the large contention window (initial parameter value of contention window is large) at a low level of congestion.
- ❑ This strategy minimizes the collision probability but it incurs a high collision probability and channel utilization is degraded in bursty arrival or congested scenarios.
- ❑ After a collision, the size of CW is set again to the minimum value without maintaining any knowledge of the current channel status.

# Backoff Alg. in Noisy Channels

- ❑ In DCF access method, immediate positive acknowledgement informs the sender that the frame was successful. The sender then transmits the next frame.
- ❑ In case of collision, the sender will not receive any acknowledgement. The sender will then wait for a random time before trying to retransmit the frame.
- ❑ When a frame is collided on wired network, the sender should slow down.
- ❑ When one is lost on a wireless network, the sender should try harder.

When a frame is collided on wired network, the sender should slow down

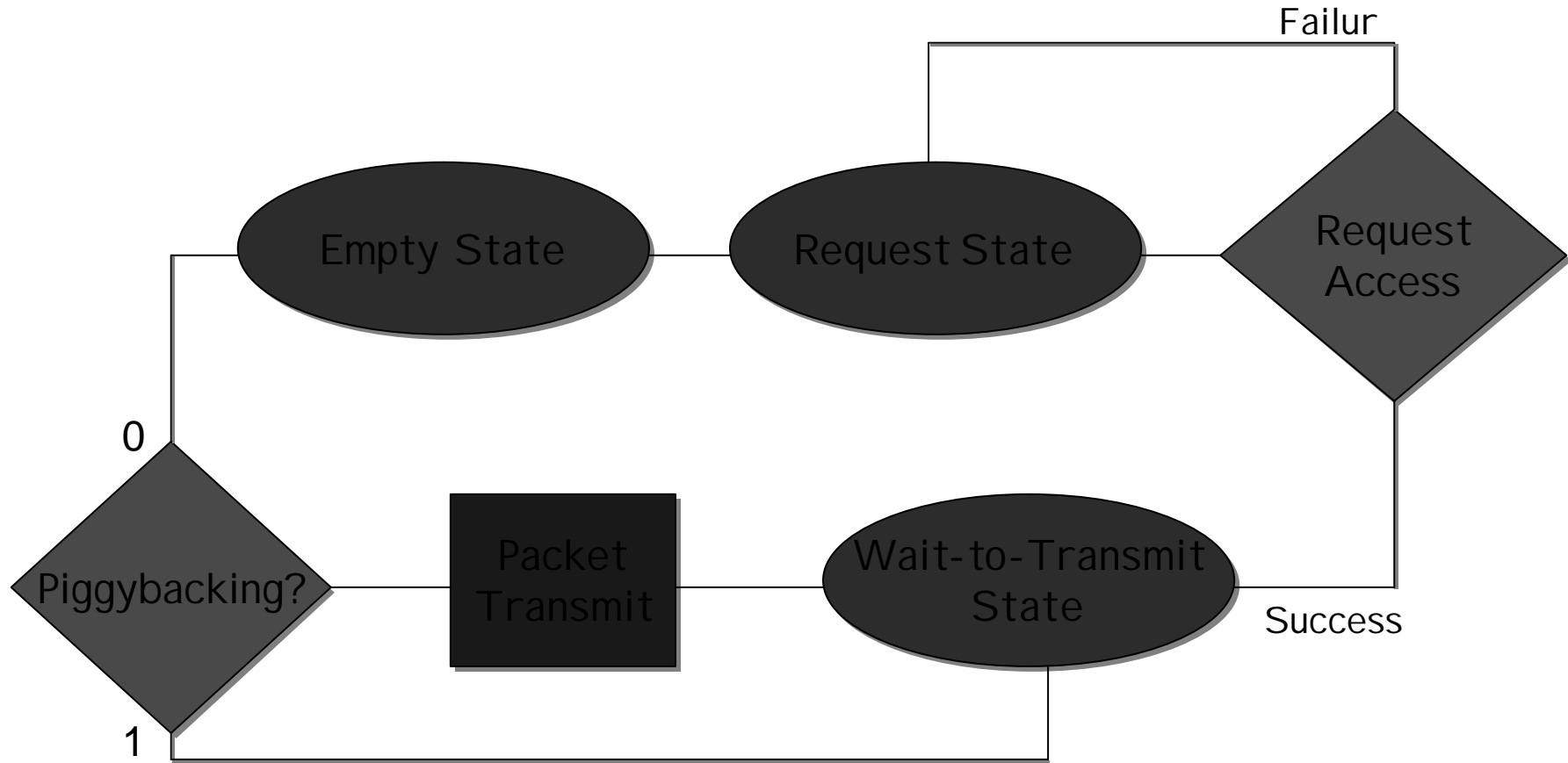
When one is lost on a wireless network the sender should try harder



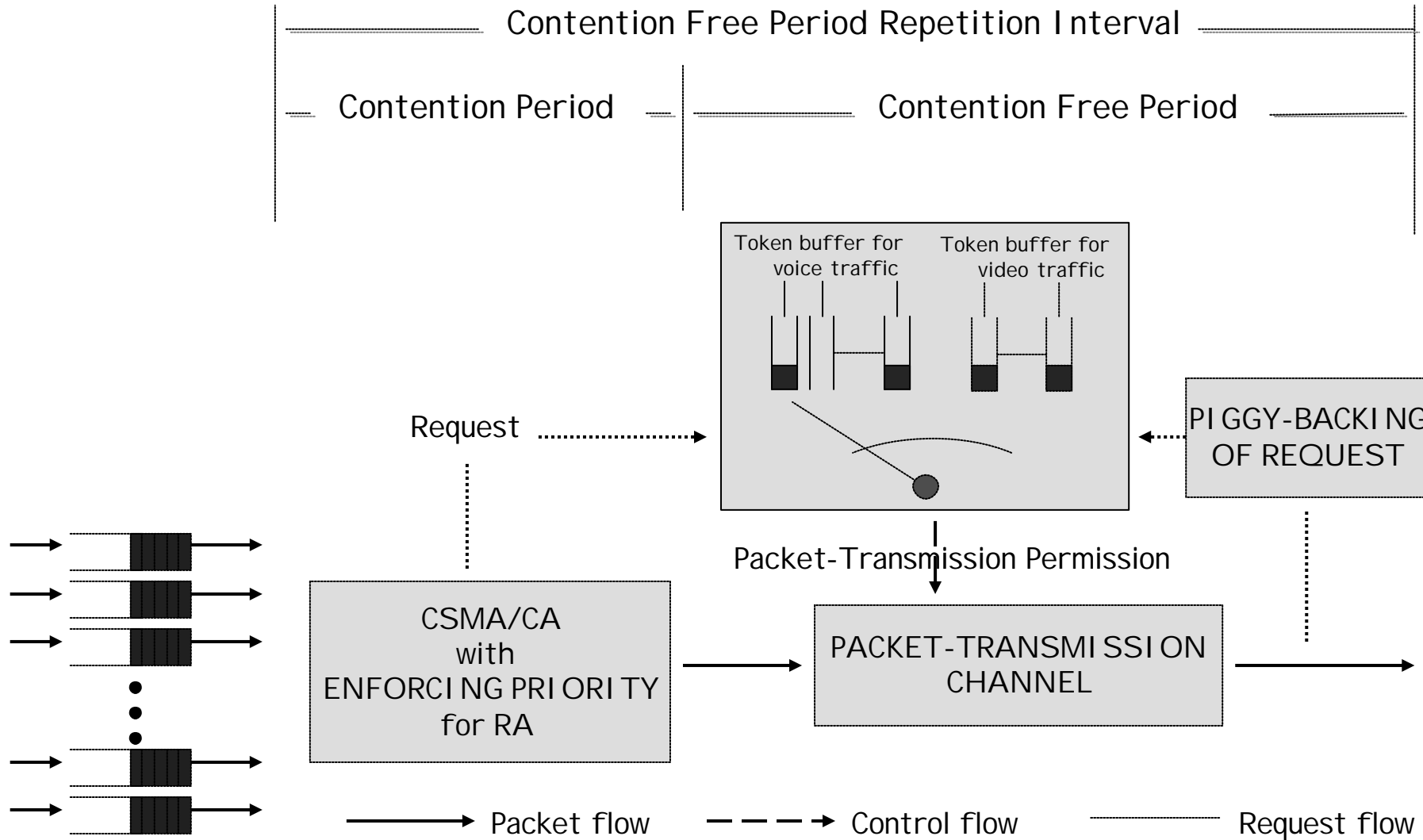
# Optimal contention window

- Step 1: 
$$p_{opt} \geq \frac{a}{M} = \frac{a}{1 + \frac{\log\left(\frac{1-a}{(1-BER)^{t_{slot}/1-q}}\right)}{\log\left(1 + \frac{2 \cdot (1-2 \cdot a)}{(2+W) \cdot a + W \cdot a \cdot (2 \cdot a)^m - (1+W)}\right)}}$$
- Step 2: 
$$Optimal\_CW = \frac{2}{p_{opt}} - 1$$
- Step 3: 
$$New\_CW = \chi \cdot Current\_CW + (1-\chi) \cdot Estimate\_Optimal\_CW$$
- Step 4: 
$$\left[ \text{rand}() \cdot 2^{\lceil \log(New\_CW) \rceil} \right] \cdot t_{slot}$$

# Channel Model for RT Station



# Packet scheduling policy in CFP



# Packet scheduling policy in CFP

- 1) The AP first scans the token buffers of voice sources according to the preset priority order. If a token is found, it removes one from this token buffer and polls this voice terminal. On receiving a poll the station transmits its packet after a SIFS interval. Then, the AP generates the next token for this voice source after  $1/r_c - (2 \cdot \text{SIFS} + \text{CFPoll} + t_p + \text{ACK})$  second if the piggyback was set while transmitting the packet.
- 2) If no tokens are found in the token buffers of voice sources, the AP continues to scan the token buffers for video sources according to the preset priority order. If a token is found, it polls this video source. And it will not remove the token if the piggyback was set while this video source transmit it packet. If the piggyback was not set and it is not the last packet (End-of-File) either, the AP removes the token, and then generates the next token for this video source after ***h*** seconds
- 3) If there is no token found in all token buffers, the AP will not know which, if any, of the stations have packets to transmit, then, it can end the CFP by transmitting a CF-End frame, and, for assuring the time constraint of admitted real-time traffic, the AP shall announce the beginning of the next CFP interval by observing the token buffer of highest priority among its polling list.



# Admission Control for voice traffic

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□ Let  $\mathbf{d}_i^* = t_p + \sum_{k=1}^{i-1} \left[ \frac{r_{ck}}{r_{ci}} \right] \bullet t_p$  ,  $i = 1, \dots, n_c$

$$t_p = 2 \cdot SIFS + CFPoll + Packet + ACK$$

If  $\mathbf{d}_i^* < 1/r_{ci}$  and  $\mathbf{d}_i^* \leq \mathbf{d}_i$  for all  $i = 1, 2, \dots, n_c$ , then all the packets generated by new-call voice sources meet their jitter constraints.

Furthermore, if  $\mathbf{d}_i^* + \mathbf{p}_i < 1/r_{ci}$  and  $\mathbf{d}_i^* + \mathbf{p}_i \leq \mathbf{d}_i$  for  $i^{th}$  sources which is handed off from other cells, then the packet generated by the  $i^{th}$  source after the handoff meets its jitter constraint.

# Admission Control for video traffic

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□ Let  $\bar{b}_0 = t_p \cdot (n_c + 1)$ ,  $\bar{r}_{v0} = t_p \cdot \sum_{i=1}^{n_c} r_{ci}$ ,  $\bar{b}_j = t_p \cdot (b_j + 1)$ ,  $\bar{r}_{vj} = t_p \cdot r_{vj}$ ,

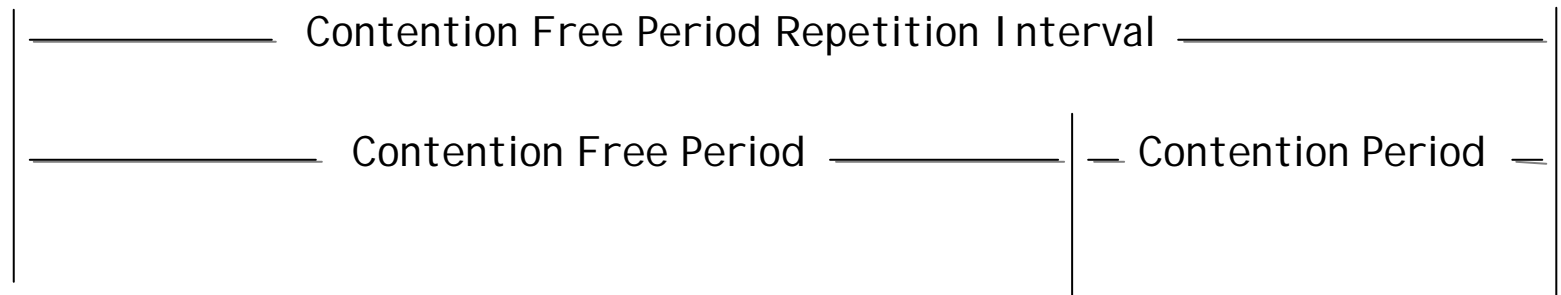
and  $d_j^* = h_j + \frac{\sum_{k=0}^j \bar{b}_k + t_p \cdot \sum_{k=1}^{j-1} (r_{vk} \cdot d_k^*)}{1 - \sum_{k=0}^{j-1} \bar{r}_{vk}}$ , where  $j = 1, \dots, n_v$ .

If  $\sum_{k=0}^{n_v} \bar{r}_{vk} \leq 1$  and  $d_j^* \leq d_j$  for all  $j$ , then the delay constraints are satisfied for all the new-call video sources. Furthermore, if  $d_j^* - h_j \leq d_j - p_j$  for  $j^{\text{th}}$  source which is handoff from other cells, then the packet generated by the  $j^{\text{th}}$  source after handoff meets its delay constraint.

# Minimized Ave. Waiting Time

- Suppose  $n_c$  voice sources are scheduled in the given priority order. The average waiting time is minimized for voice packets if  $r_{ci} \leq r_{cj}$  for all  $i < j$ .

# Adaptive Bandwidth Allocation Strategy



Channel I  
for

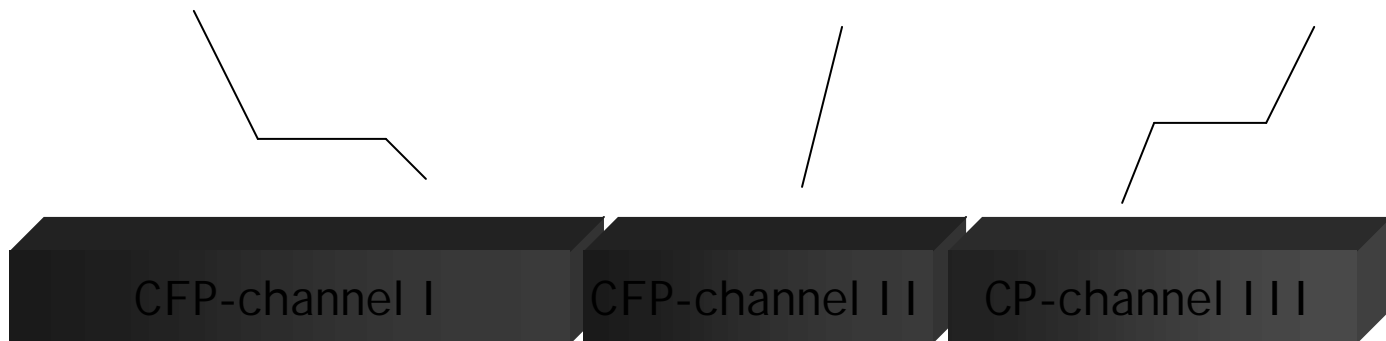
Channel II  
for

Channel III  
for

new-call/handoff voice/video traffic

handoff voice/video traffic

data traffic



# Adaptive Bandwidth Allocation Strategy

**IF** monitored dropping probability > threshold\_D **THEN**

**IF** bandwidth utilization <  $\alpha$  **THEN**

size of allocated bandwidth  $II = \min \{ \max \{ \text{size of allocated bandwidth } I, \text{ size of allocated bandwidth } II \} \text{ up\_}g, \text{ total bandwidth } \}$

**ELSE**

size of allocated bandwidth  $II = \min \{ \max \{ \text{size of allocated bandwidth } I, \text{ size of allocated bandwidth } II \} \text{ up\_}g, \text{ total bandwidth threshold\_up\_}II \}$

**ELSE**

# Adaptive Bandwidth Allocation Strategy (cont'd)

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**IF** monitored blocking probability > threshold\_B **THEN**

**IF** bandwidth utilization <  $\alpha$  **THEN**

size of allocated bandwidth I = min {size of allocated bandwidth I<sub>up\_g</sub>, total bandwidth threshold.1\_up\_I }

**ELSE**

size of allocated bandwidth I = min {size of allocated bandwidth I<sub>up\_g</sub>, total bandwidth threshold.2\_up\_I }

**ELSE**

**IF** bandwidth utilization <  $\beta$  **THEN**

size of allocated bandwidth II = max {size of allocated bandwidth II<sub>down\_g</sub>, total bandwidth threshold\_down\_II }

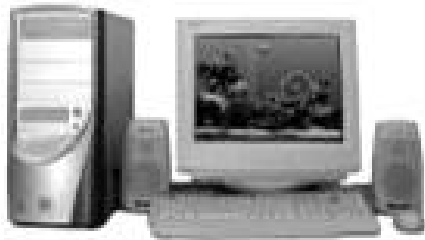
size of allocated bandwidth I = max {size of allocated bandwidth I<sub>down\_g</sub>, total bandwidth threshold\_down\_I }

# Conclusions

- ❑ A feasible and pragmatic non-preemptive priority based access control scheme was proposed
- ❑ Prioritization is key to optimizing overall performance
- ❑ Various QoS requirements are needed in the future
- ❑ Multilevel priorities, bandwidth allocation, connection admission control, and traffic policing all need to be considered together in the future networks
- ❑ There's no such thing as a free lunch  
Ongoing efforts to provide "perfect" solutions have illustrated that attempts to solve all possible problems result in technologies that are far too complex, have poor scaling properties, or simply do not integrate well into the diversity of the Internet.

Besides, we believe that it is almost impossible to increase the probability of success of transmitting a frame excepting frames fragmentation or FEC (Forward Error Control) in an extremely noisy wireless environment.

# Project (鉅格)



Multimedia PC

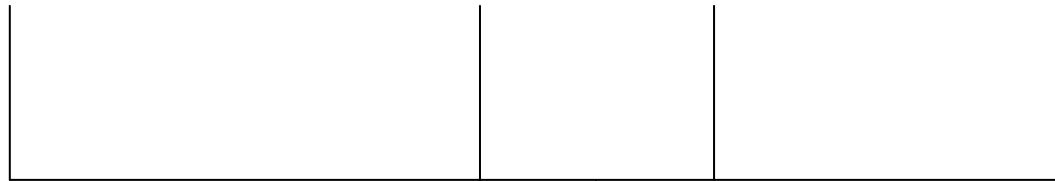


Walkman

CD player



MP3 player



PMP



# Project (鉅格)

## Future Digital Home System

Wireless  
Multimedia  
Gateway

