Solutions to Performance Problems in VOIP over 802.11 Wireless LAN

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Outline

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- VOIP background
- Problems faced in 802.11
  - Low VOIP capacity in 802.11
  - Interference between VOIP and TCP
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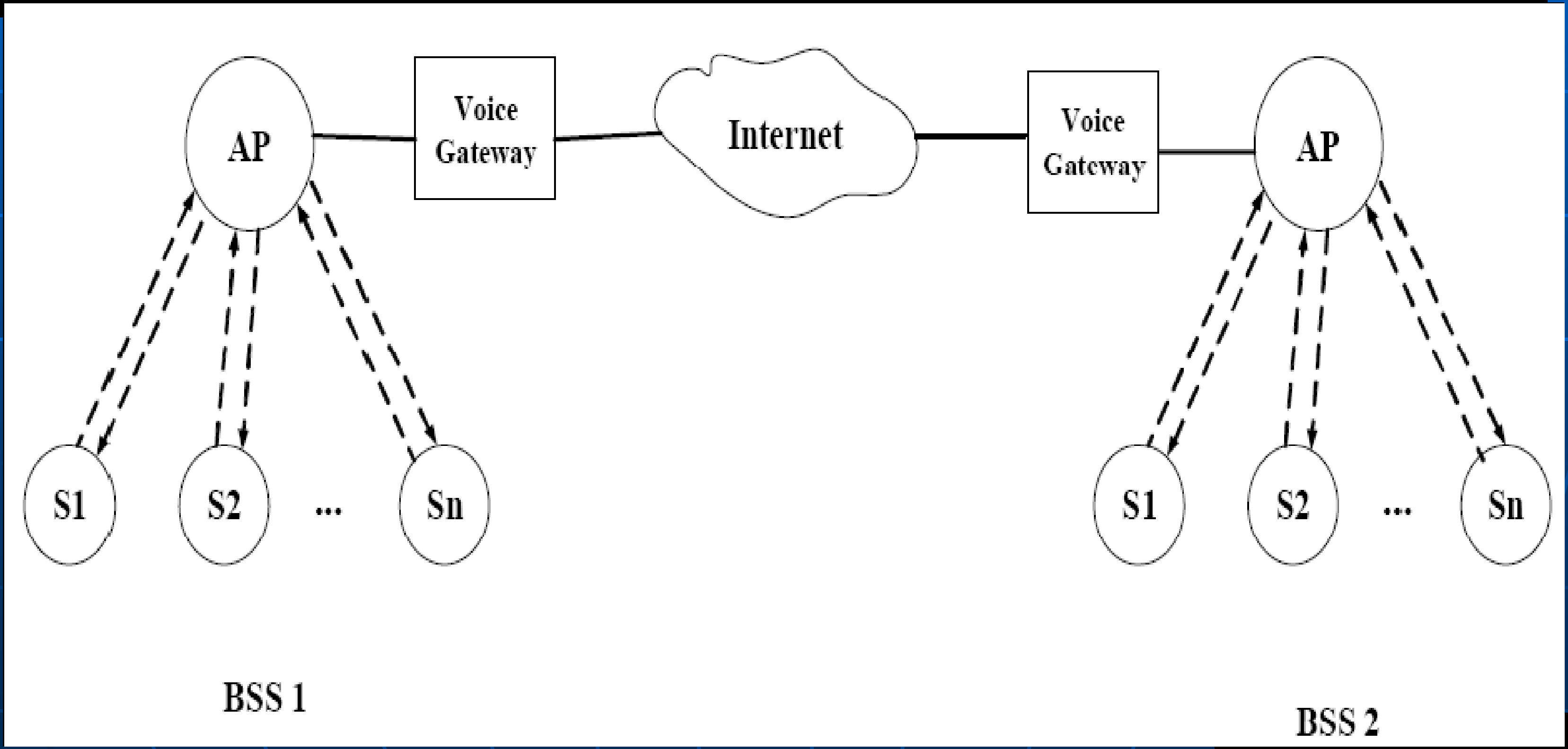
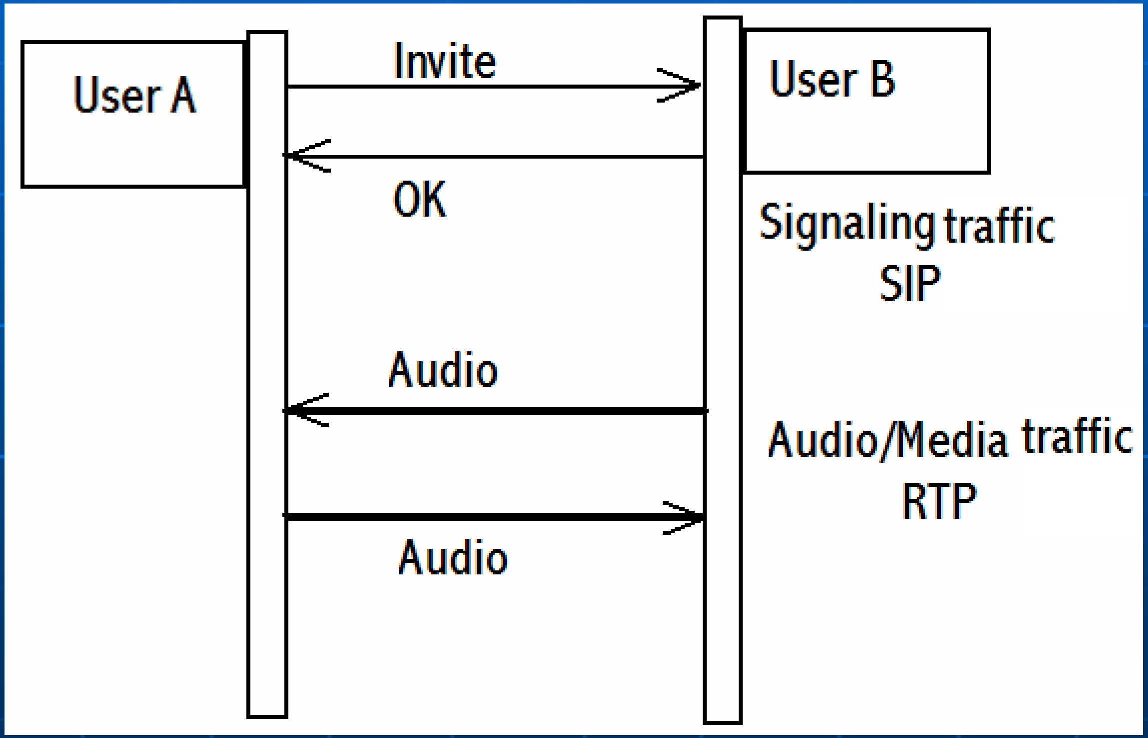
Introduction

VOIP in an 802.11 Network



VOIP Background

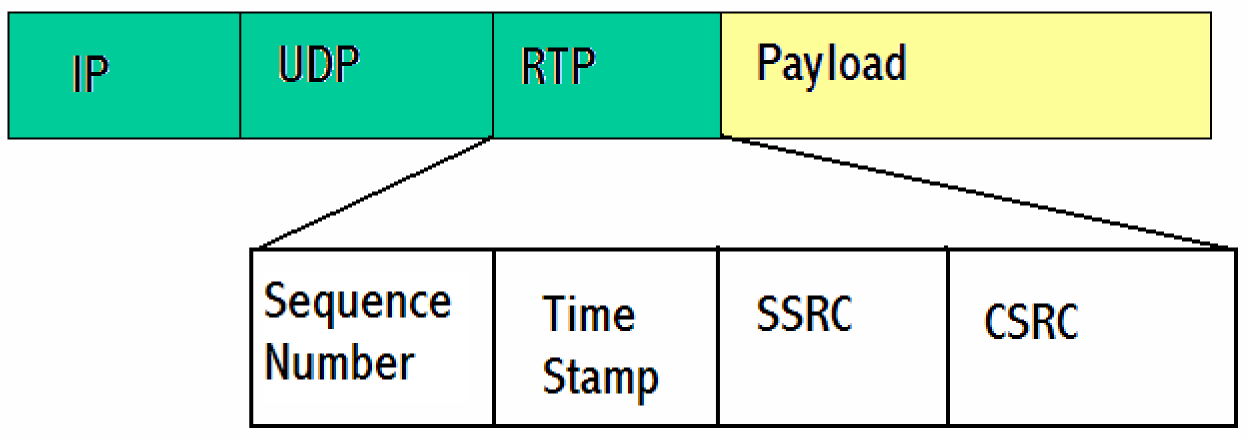
VOIP Call



VOIP Background

RTP

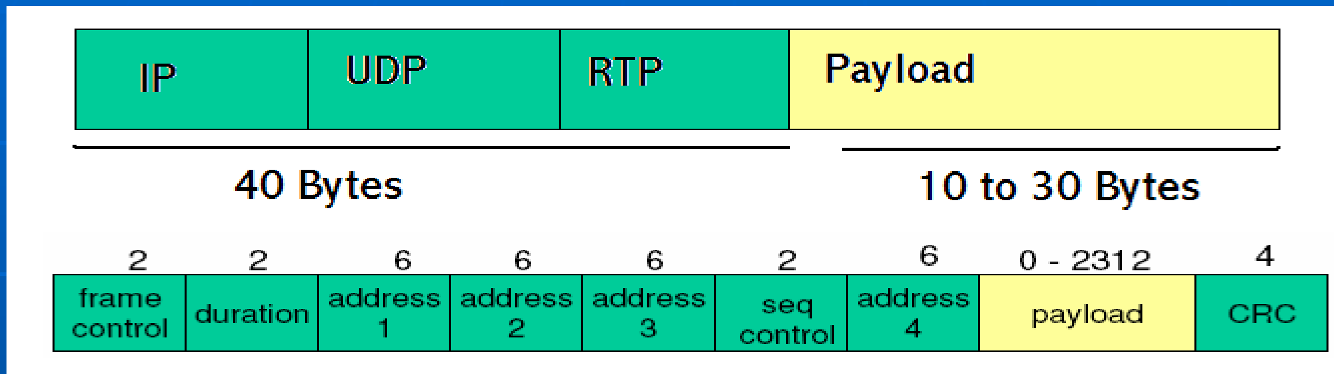
- Real-time transport protocol.
- Built on UDP.
- Sequence Numbering.
- Time Stamping.
- Sent at a continues rate every 20ms.

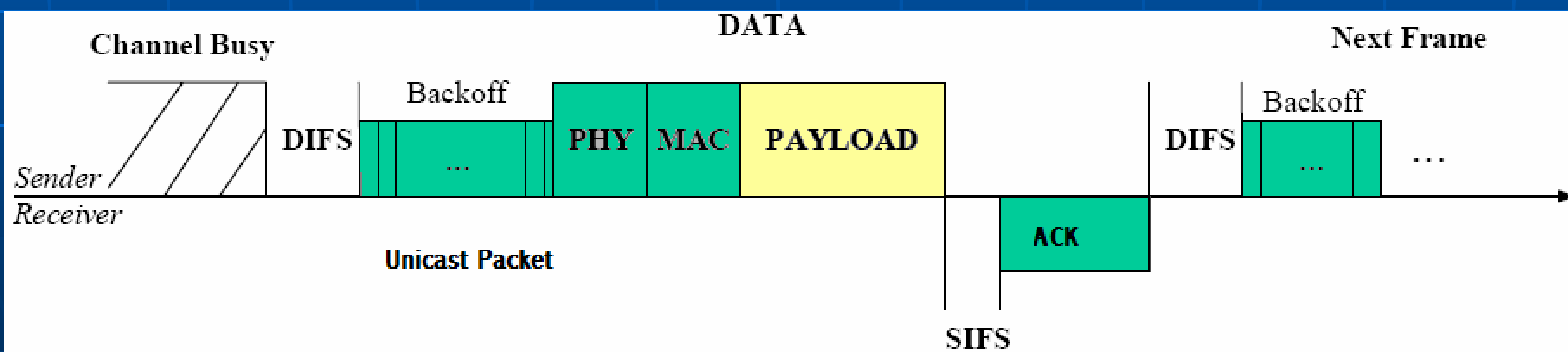


Problems

- Low system capacity in WLAN network for VOIP calls.
- Interference between VOIP traffic and data traffic.

Too Much Overhead





Too Much Overhead

DIFS	50 μsec
SIFS	10 μsec
Slot Time	20 μsec
CWmin	32
CWmax	1023
Data Rate	1, 2, 5.5, 11 Mbps
Basic Rate	2 Mbps
PHY header*	192 μsec
MAC header	34 bytes
ACK*	248 μsec

22 μs to transmit the payload

29 μs to transmit the header

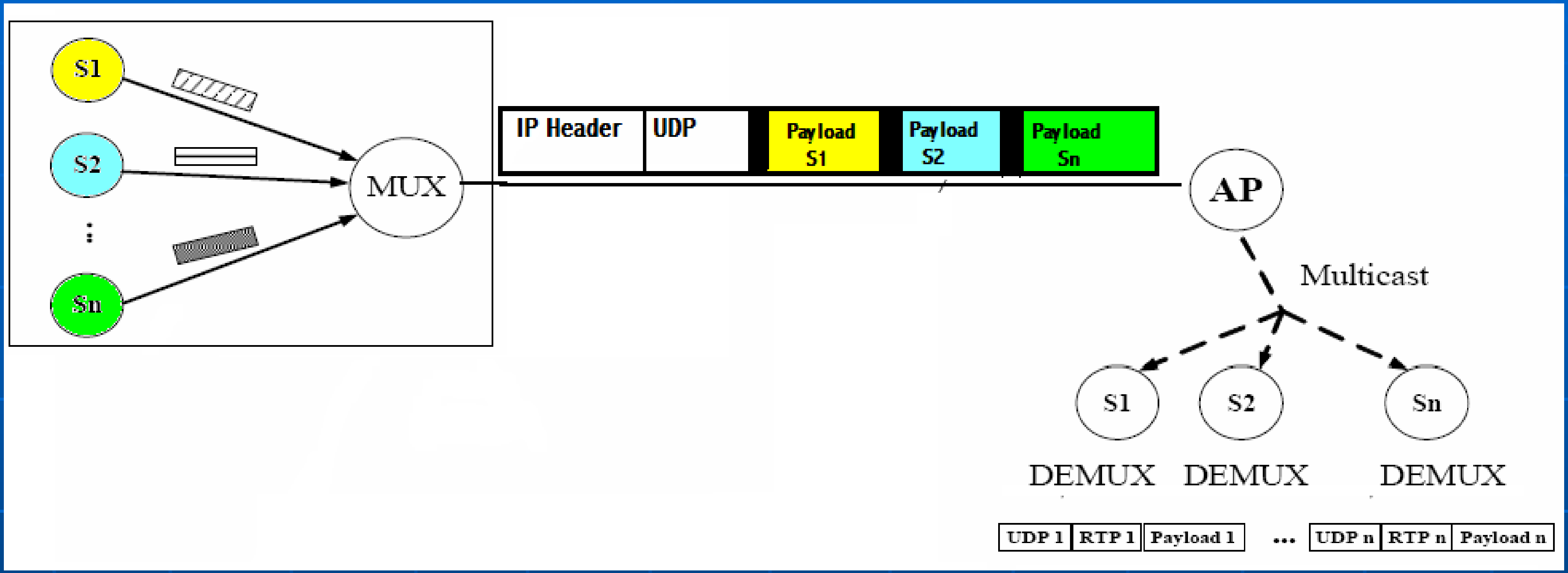
802.11 MAC\PHY layer additional ***800 μs***

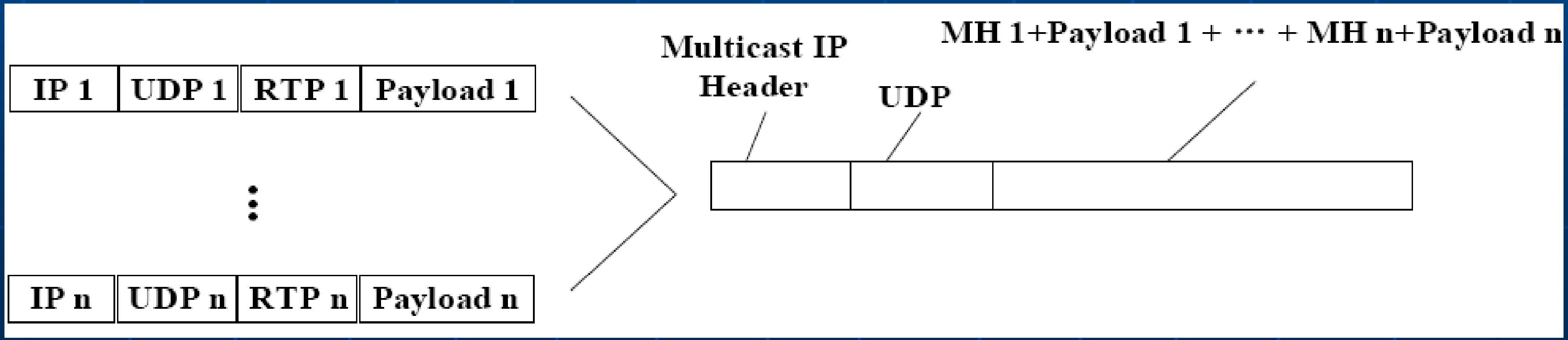
Solution is to use **Voice Multiplex-Multicast**

Voice Multiplex-Multicast (M-M)

- Multiplex packets from various streams into one stream to reduce overhead.
- Replace RTP,UDP and IP header with a mini-header.
- Each payload is preceded with this mini-header representing an ID.
- The AP broadcasts this packet and ID is translated into RTP,UDP IP header at the receiver.

Voice Multiplex-Multicast (M-M)





Issues

- Broadcast issue.
- Security.
- Power Saving Mode of AP.

Voice Capacity Analysis

- **Original VOIP**

$$1/T_{avg} = \text{number of streams} * \text{number of packets sent by one stream in one second.}$$

$$OH_{hdr} = H_{RTP} + H_{UDP} + H_{IP} + H_{MAC}$$

$$OH_{sender} = DIFS + averageCW + PHY$$

$$OH_{receiver} = SIFS + ACK$$

$$T_{down} = T_{up} = (Payload + OH_{hdr}) * 8 / dataRate + OH_{sender} + OH_{receiver}$$

$$T_{avg} = (T_{down} + T_{up})/2$$

$$n = 11.2$$

Voice Capacity Analysis

- **M-M VOIP**

Mini-header is used to compress the IP/UDP/RTP header into 2 bytes.

$$T_{down} = [(Payload + 2) * n + H_{UDP} + H_{IP} + H_{MAC}] * 8 / dataRate + OH_{sender}$$

$$T_{avg} = (T_{down} + n * T_{up}) / (n + 1)$$

$$1 / T_{avg} = (n + 1) * N_p$$

$$n = 21.2$$

Voice Capacity Analysis

Two type of sources

- Constant-bit-rate (**CBR**)
- Variable-bit-rate (**VBR**)

In VBR average traffic is **45%** of CBR.

Original VOIP with VBR source can have **26** sessions.

M-M VOIP with VBR source can have **50** sessions.

Simulation

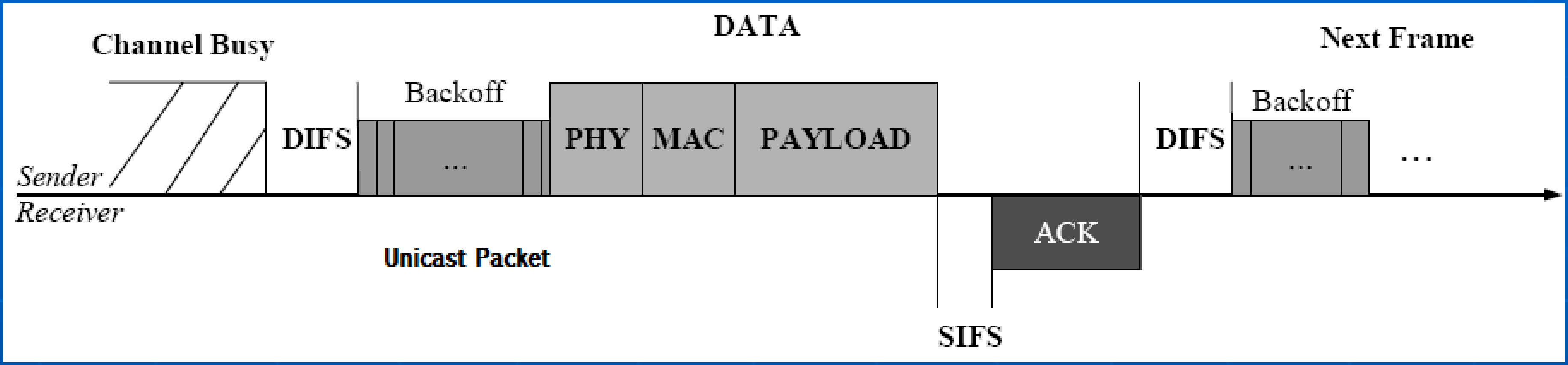
- ns-2 simulation
- Payload size GSM 6.10 codec
- Increase the number of streams until we reach 1% packet loss

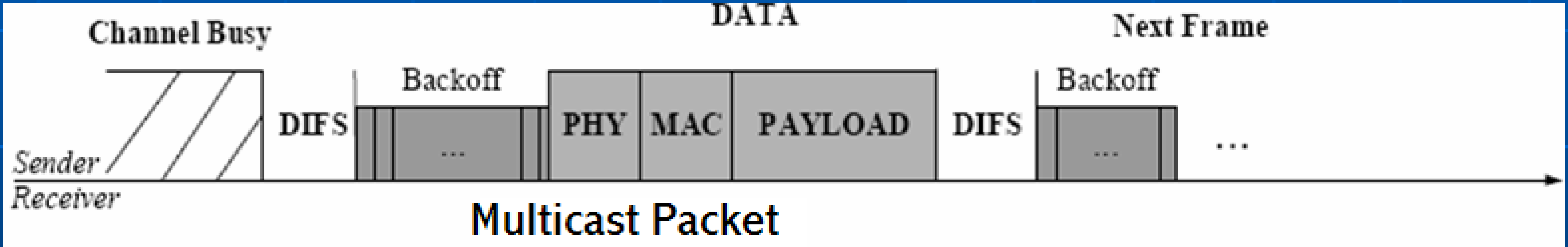
Different Schemes	CBR		VBR	
	Analysis	Simulation	Analysis	Simulation
Original VoIP	11.2	12	26.3	25
Multiplex-Multicast Scheme	21.2	22	49.8	36*

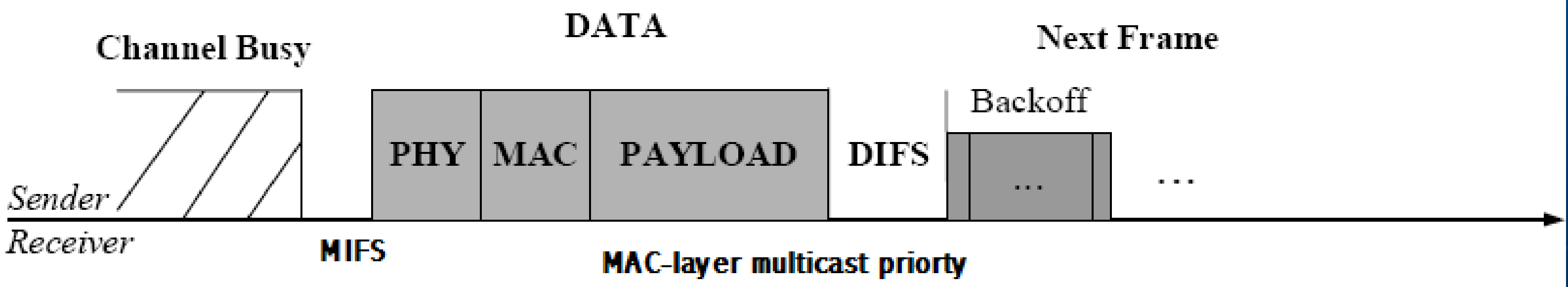
Simulation Observations

- Adding 13th session to CBR for original VOIP causes 6% Packet loss.
- VBR for M-M scheme number of sessions is lower then expected.
- VBR traffic is bursty causing packet collisions.
- Multicast frames will be dropped after first collision.

802.11 Packets



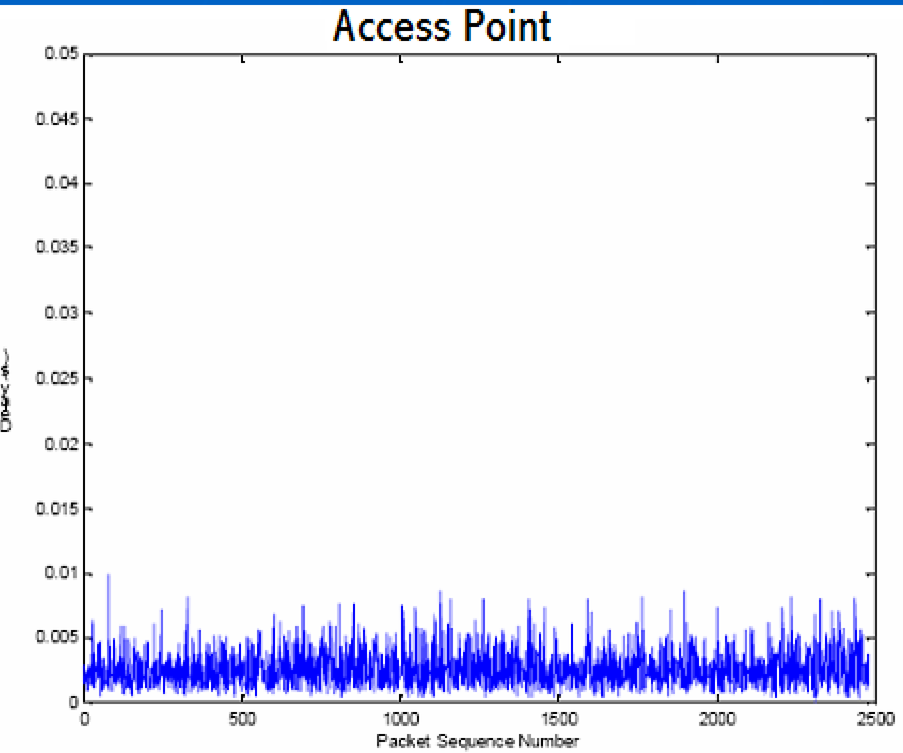


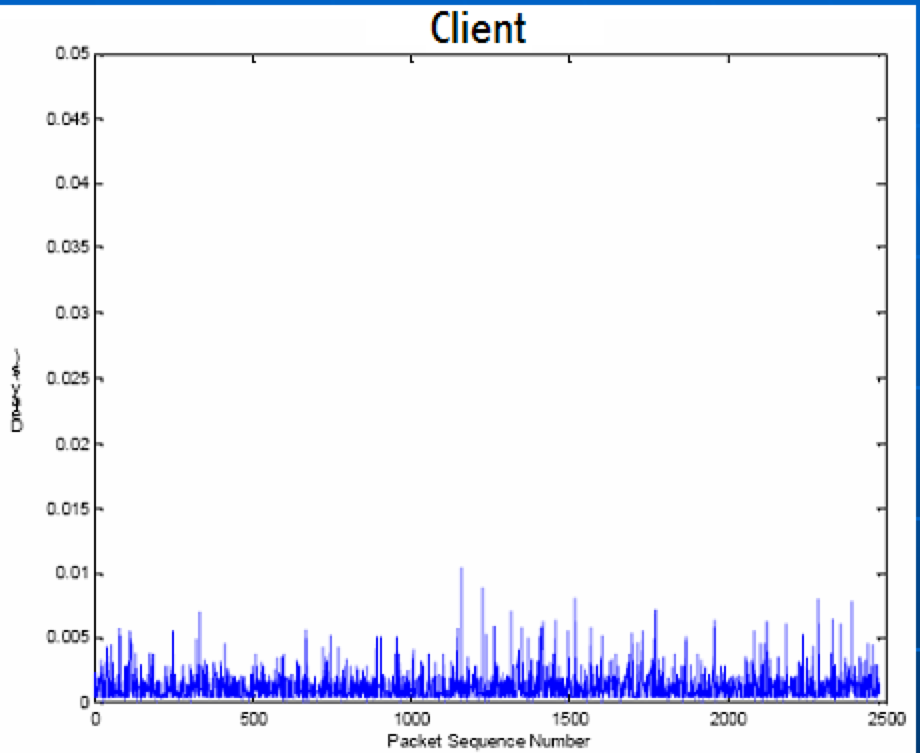


Delay Performance

- To provide good quality minimizing the delay is important.
- Delay jitter is the variation in delay.
- Delay of simple VOIP is AP + Receiver Station.
- Delay of M-M is AP + MUX + Receiver.
- We want to keep less then 1% packets with delay of 30ms.

Delay Performance (CBR)





Original VOIP with 12 sessions

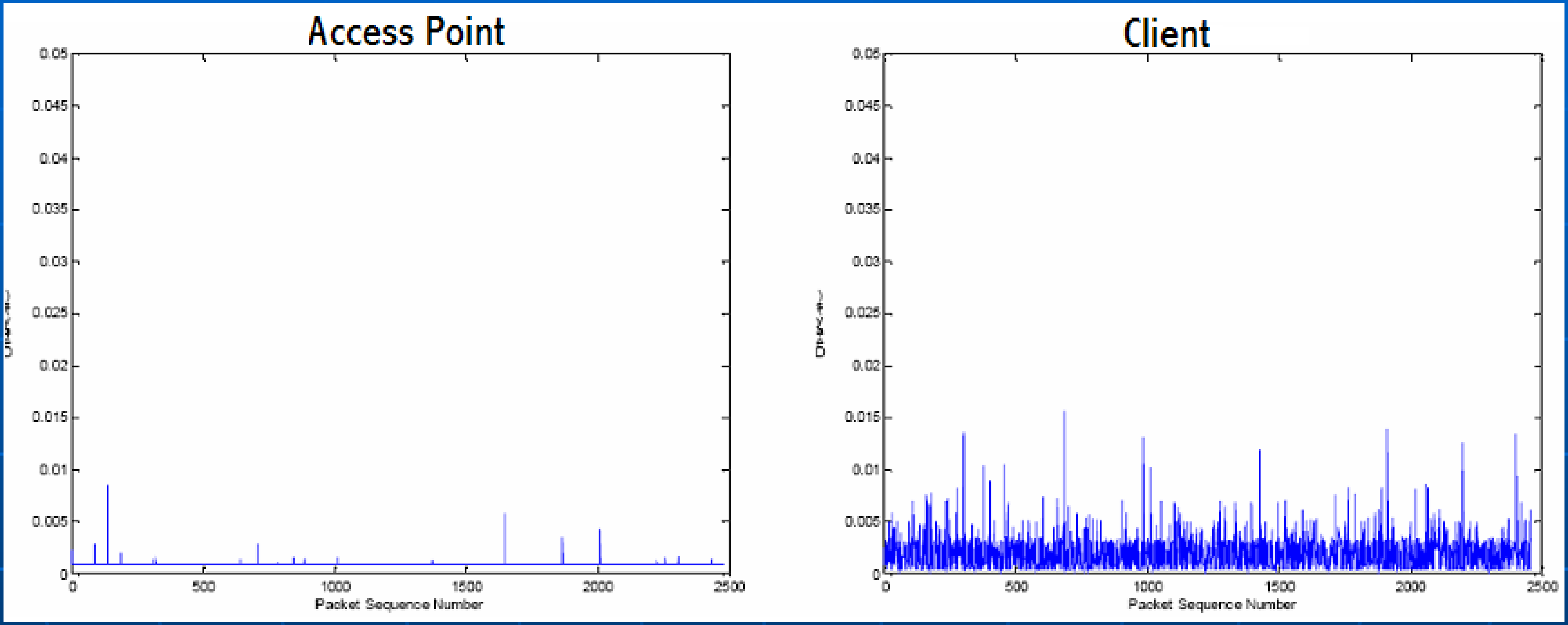
Average delay 2.5ms

Jitter delay 1.4ms

Average delay 1.2ms

Delay jitter 1ms

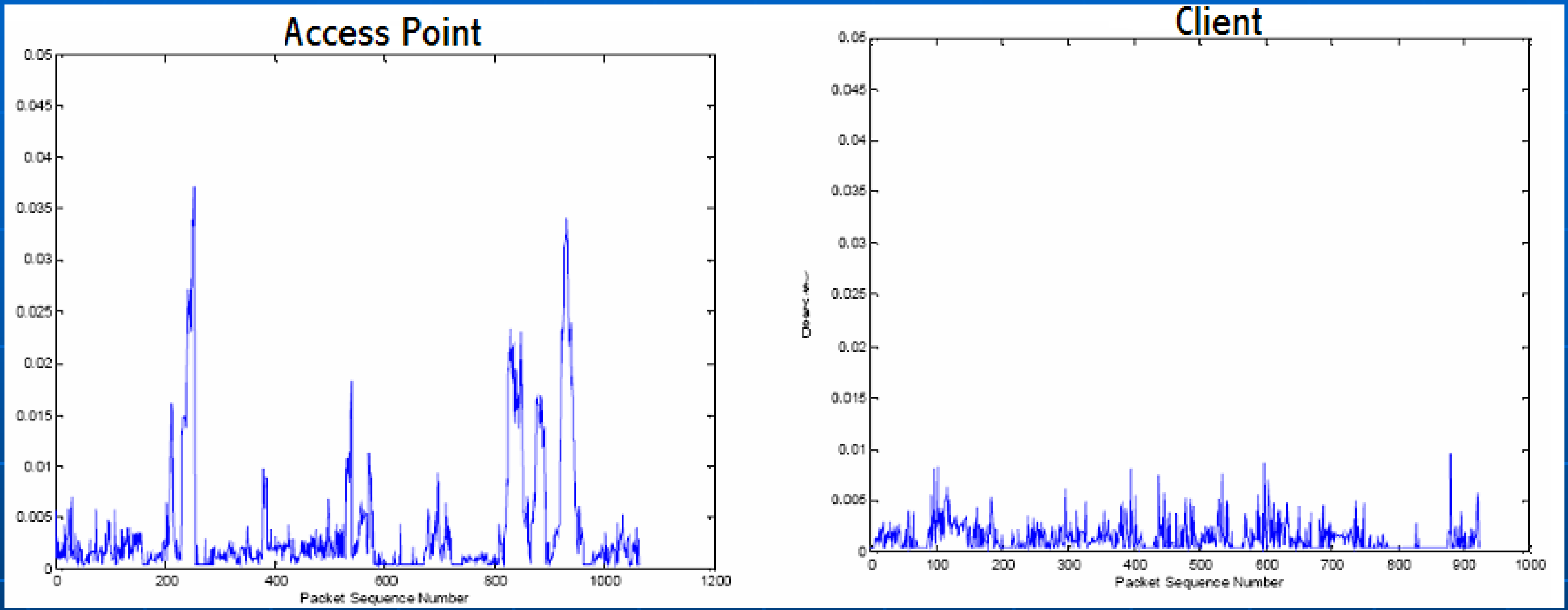
Delay Performance (CBR)



M-M VOIP with 22 sessions

Average delay 0.9ms	Average delay 2ms
Jitter delay 0.2ms	Delay jitter 1.5ms

Delay Analysis (VBR)



Original VOIP with 25 sessions

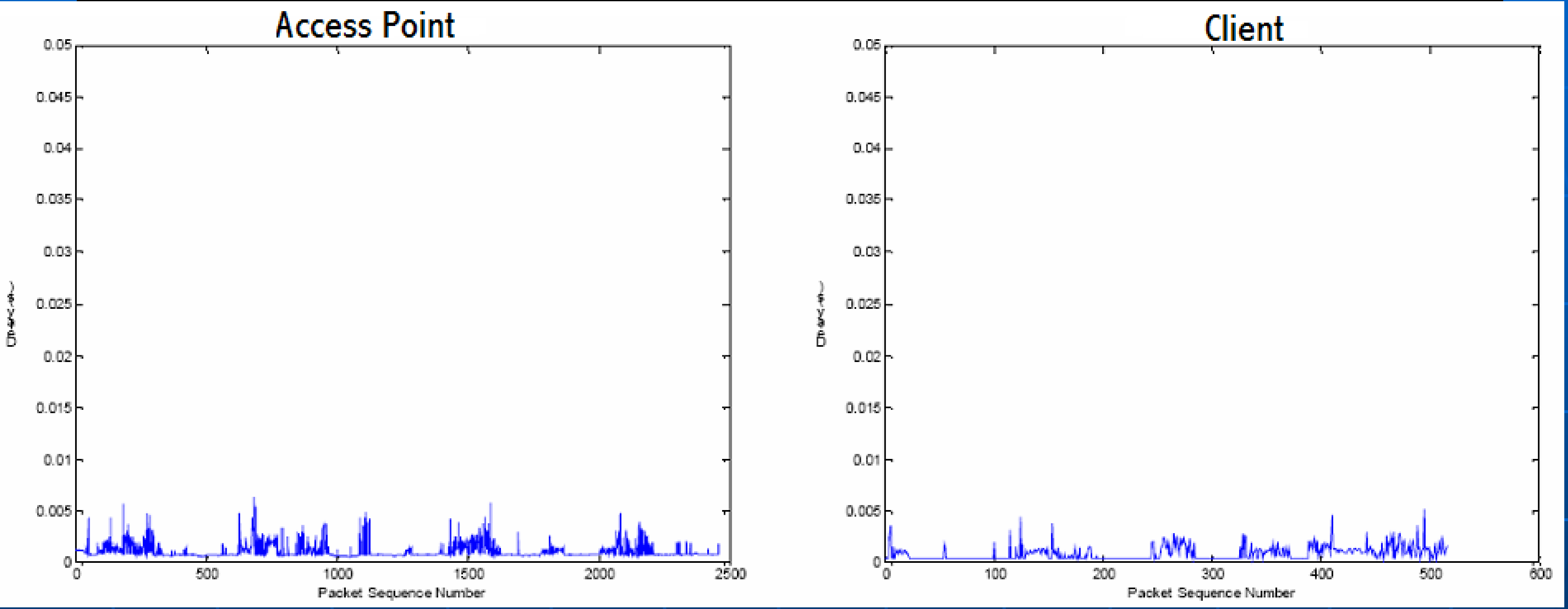
Average delay 3.6ms

Jitter delay 5.9ms

Average delay 1.4ms

Delay jitter 1.3ms

Delay Analysis (VBR)



M-M VOIP with 36 sessions

Average delay 1.1ms
Jitter delay 0.7ms

Average delay 0.9ms
Delay jitter 0.7ms

Delay Analysis

Access Delay Distribution for Ordinary VoIP

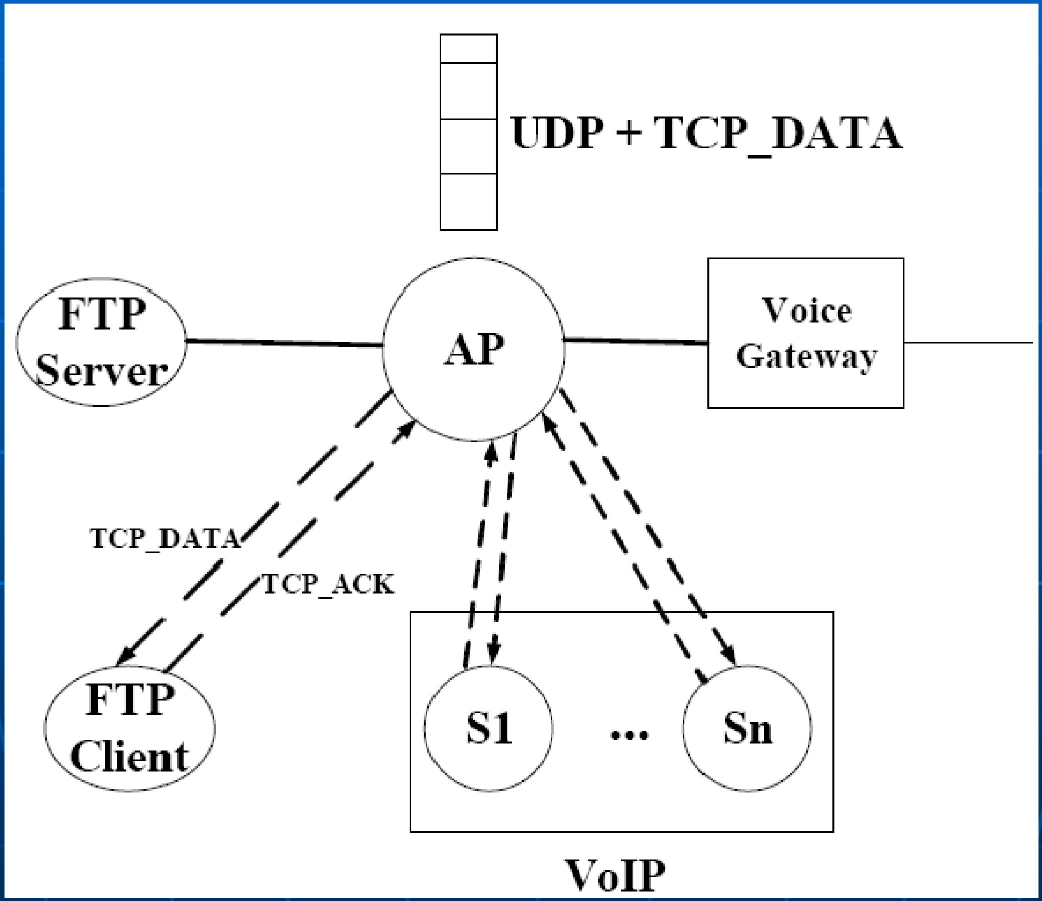
	Access delay for the AP (Local delay for downlink VoIP packets)		Access delay for the station (Local delay for uplink VoIP packets)	
	CBR(12)	VBR(25)	CBR(12)	VBR(25)
$\Pr[A \le 10ms]$	1	0.900	0.999	1
$\Pr[A \le 30ms]$	1	$\boxed{0.990}$	1	1
$\Pr[A \le 50ms]$	1	1	1	1

Delay Analysis

- Delay at the MUX is upper bound limited by 20ms.
- We are going to consider MUX delay to be distributed from 0 to 20 ms

Access delay for the AP plus MUX delay in the MUX (Local delay for the downlink VoIP packet)			Access delay for the station (Local delay for the uplink VoIP packet)		
	CBR(22)	VBR(36)		CBR(22)	VBR(36)
$\Pr[M + A \le 0.01s]$	0.455	0.447	$\Pr[A \le 0.01s]$	0.996	1
$\Pr[M + A \le 0.02s]$	0.955	0.947	$\Pr[A \le 0.02s]$	1	1
$\Pr[M + A \le 0.03s]$	1	1	$\Pr[A \le 0.03s]$	1	1

Coexistence with TCP



Coexistence with TCP

- TCP interferes with downlink FIFO queue.
- TCP ACK interferes with clients sending VOIP streams.

Coexistence with TCP

6 VOIP sessions with one TCP flow

Orignal VOIP

Access delay / jitter of the AP (ms)	Access delay / jitter of the station (ms)	VoIP downlink packet loss rate	VoIP uplink packet loss rate	TCP throughput (Mbps)
83.9 / 15.6	2.3 / 3.0	1.0 %	0	2.55

Orignal VOIP with Priority Queuing

Access delay / jitter of the AP (ms)	Access delay / jitter of the station (ms)	VoIP downlink packet loss rate	VoIP uplink packet loss rate	TCP throughput (Mbps)
3.0 / 1.5	2.6 / 2.2	0.01 %	0	2.55

Coexistence with TCP

6 M-M VOIP sessions with one TCP flow

	Access delay / jitter of the AP (ms)	Access delay / jitter of the station (ms)	VoIP downlink loss rate	VoIP uplink loss rate	TCP throughput (Mbps)
M-M	42.7 / 19.2	4.5 / 6.2	10.8 %	0	3.46
M-M + PQ	4.3 / 2.4	4.7 / 6.2	12.2 %	0	3.49
M-M + MMP	17.2 / 14.5	4.4 / 5.2	0	0	3.47
M-M + PQ+MMP	2.7 / 2.1	4.6 / 5.8	0	0	3.47

Coexistence with TCP

11 M-M VOIP sessions with one TCP flow

	Access delay / jitter of the AP (ms)	Access delay / jitter of the station (ms)	VoIP downlink loss rate	VoIP uplink loss rate	TCP throughput (Mbps)
M-M	32.5 / 25.8	6.6 / 10.2	15.6 %	0	2.55
M-M + PQ	4.5 / 3.2	6.7 / 13.5	12.0 %	0	2.54
M-M + MMP	20.3 / 21.7	8.9 / 20.8	0.2 %	0	2.54
M-M + PQ+MMP	2.9 / 2.7	5.8 / 7.2	0	0	2.54

Transmission Errors

- In a real world there are transmission errors.

Multicast Packet Loss Rate for Different Distances and Data Rates

Distance (m)	Multicast frames at 2 Mbps	Multicast frames at 11 Mbps
1	0	0.17%
5	0	0.15%
10	0	0.17%
20	0.02%	0.23%

802.11e

Implements QOS by having multiple queues for different types of packets.

- EDCA0: One queue for all the traffic, same parameter setting as in DCF
- EDCA1: CWmin[voice] = CWmin[data] = 31
- EDCA2: CWmin[voice] = 31, CWmin[data] = 63
- EDCA3: CWmin[voice] = 31, CWmin[data] = 127

802.11e

Performance of Different Parameter Settings for One VoIP + One TCP

	Access delay / jitter of the AP (ms)	Access delay / jitter of the station (ms)	TCP throughput (Mbps)
EDCA0	23.26 / 15.46	1.98 / 1.47	3.45
EDCA1	2.72 / 2.12	2.84 / 2.06	3.45
EDCA2	2.21 / 1.54	2.23 / 1.41	3.07
EDCA3	1.99 / 1.15	1.94 / 1.16	2.43

Performance of Different Parameter Settings for Six VoIP + One TCP

	Access delay / jitter of the AP (ms)	Access delay / jitter of the station (ms)	TCP throughput (Mbps)
EDCA0	56.15 / 26.62	4.12 / 2.65	2.19
EDCA1	14.58 / 6.43	4.89 / 4.17	2.44
EDCA2	10.82 / 3.02	4.29 / 2.94	2.16
EDCA3	9.23 / 2.03	3.86 / 2.56	1.71

Conclusion

- M-M improves the VOIP capacity.
- M-M requires no MAC change in the Client station.
- M-M doesn’t increase delay above 30ms.
- With Both Original VOIP and M-M VOIP quality is unacceptable with TCP.
- Priority queue can solve TCP interference problem.

Observations

- Didn't discuss the effect of mobility causing variable bandwidth.
- Need to look at scenario with majority of TCP and few VOIP users.
- Power utilization at the client stations.
- Only works between AP and AP.