Voice Traffic Simulation in an IEEE802.11 Wireless LAN Environment

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- Introduction to IEEE802.11b WLAN
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Why WLAN

- Today's business professionals expect
  - Mobility
  - High speed access
- Easy and Fast to implement (Fewer cables to deploy)

VoIP over WLAN

- voice over package can save cost for corporations—one network to maintain
- VoIP was first developed in 1995 (PC-to-PC)
- VoIP has continued to mature.

Standard of WLAN

- **HIPERLAN by ETSI**
  - HIPERLAN/1—20Mbps—5GHz band
  - HIPERLAN/2—54Mbps—5GHz band
- **IEEE802.11 series by IEEE**
  - IEEE802.11a—54Mbps—5GHz band
  - IEEE802.11b—11Mbps—2.4GHz band
  - IEEE802.11g—54Mbps—5GHz band

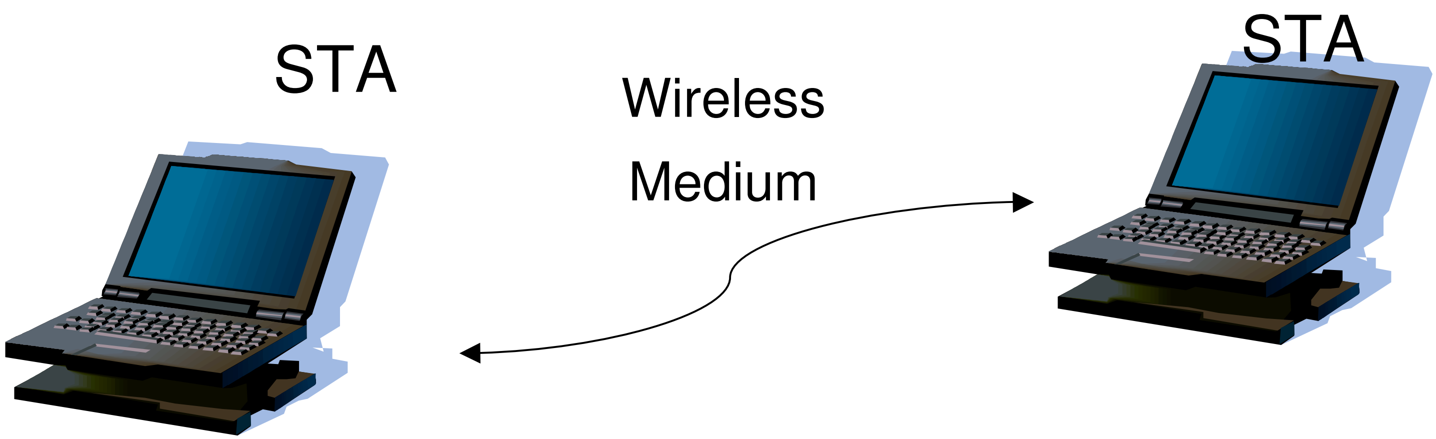
IEEE802.11b WLAN

- Standardized in 1999
- Operate in the 2.4GHz ISM (Industrial, Scientific and Medical) band
- Basic access method DCF
- Optional access method PCF
- Max rate 11Mbps

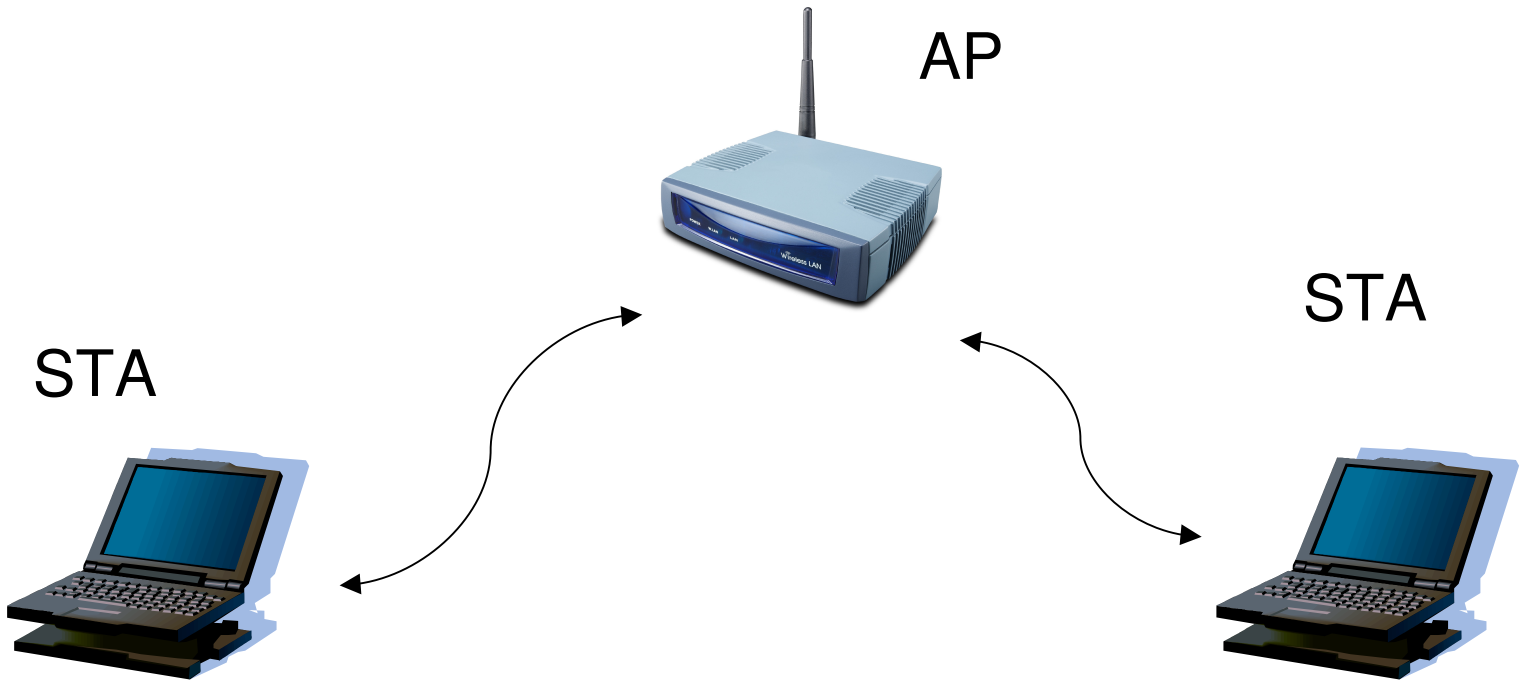
Components in IEEE802.11 WLAN

- Station (STA)
- Access Point (AP)
- Distribution System (DS)
- Basic Service Set (BSS)
- Independent BSS
- Infrastructure BSS
- Extended Service Set (ESS)

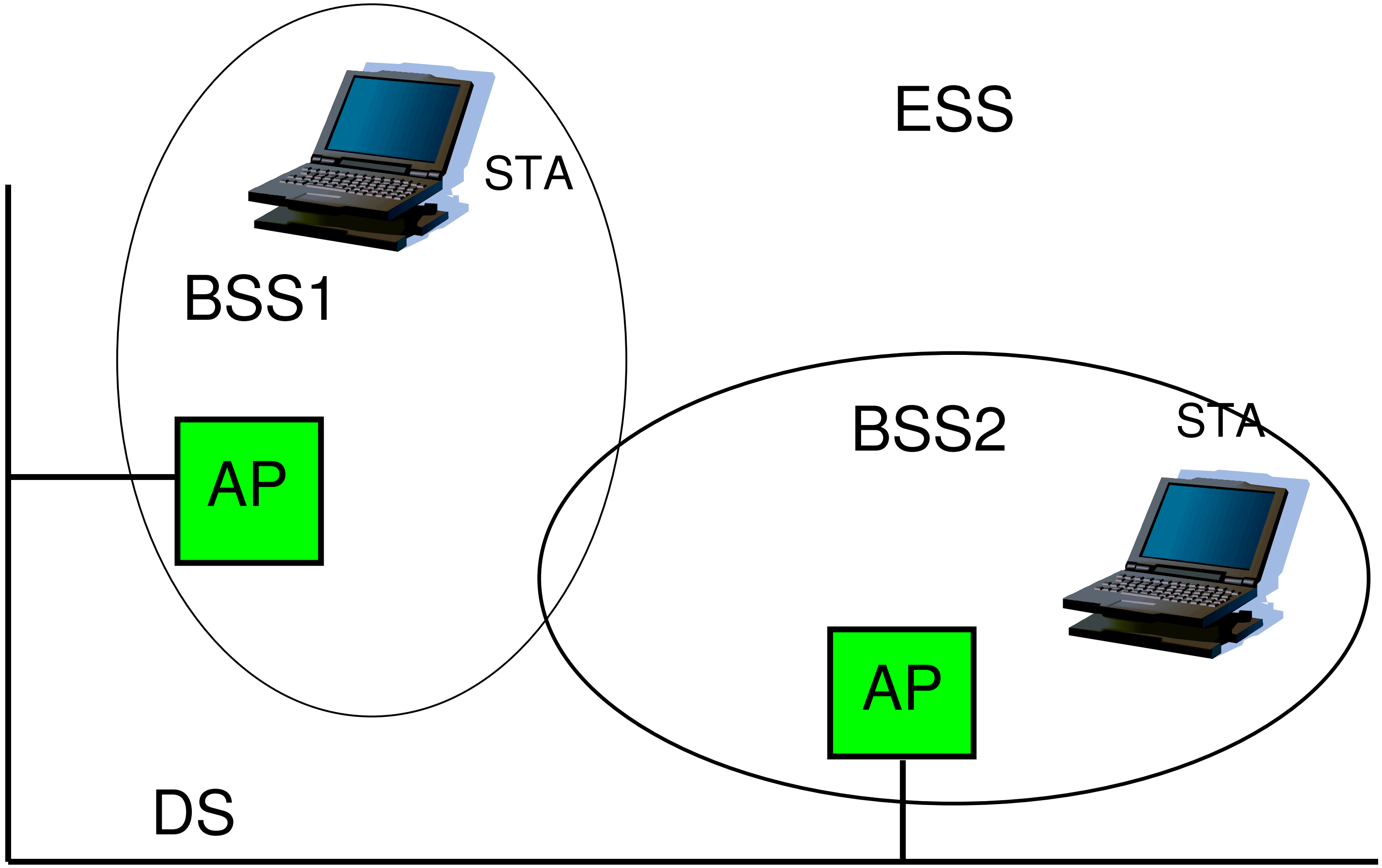
Independent BSS



Infrastructure BSS



BSS and ESS



Services in IEEE802.11

- Station service
  - Authentication
  - Deauthentication
  - Privacy
  - MSDU delivery
- Distribution system service
  - Association
  - Disassociation
  - Distribution
  - Integration
  - Reassociation

Frame format

- Control frame
- Data frame
- Management frame

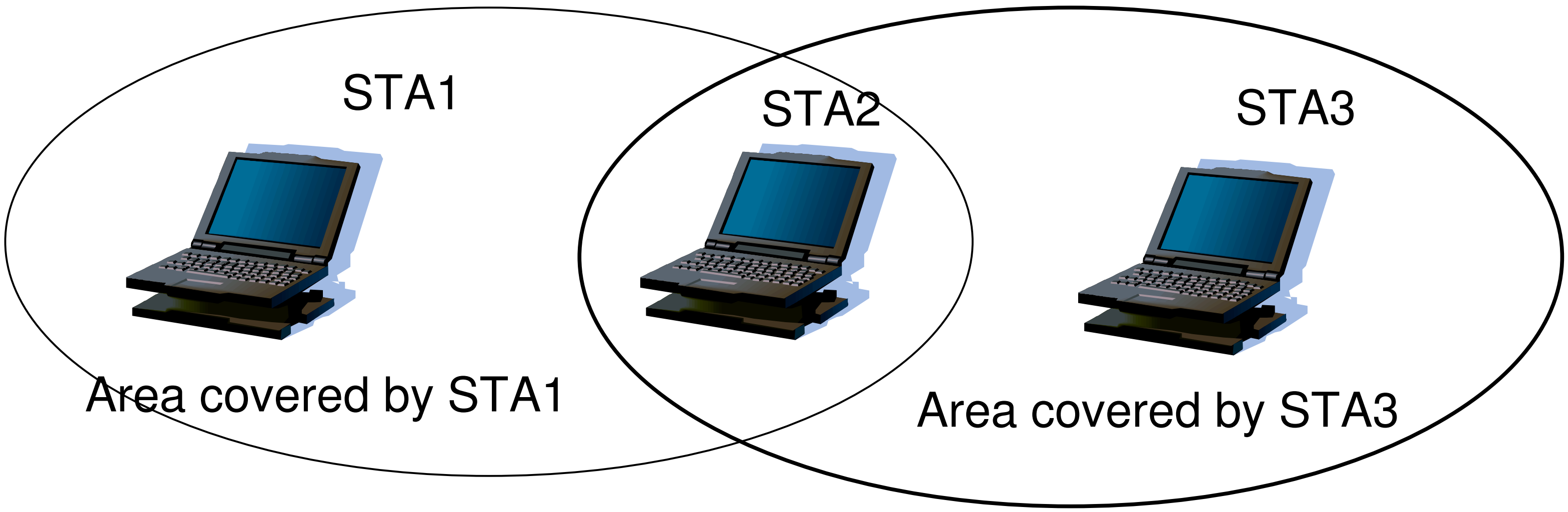
MAC Header

Frame Control	Duration /ID	Address 1	Address 2	Address 3	Sequence Control	Address 4	Frame Body	FCS
2 bytes	2 bytes	6 bytes	6 bytes	6 bytes	2 bytes	6 bytes	0-2312 bytes	4 bytes

Access Methods

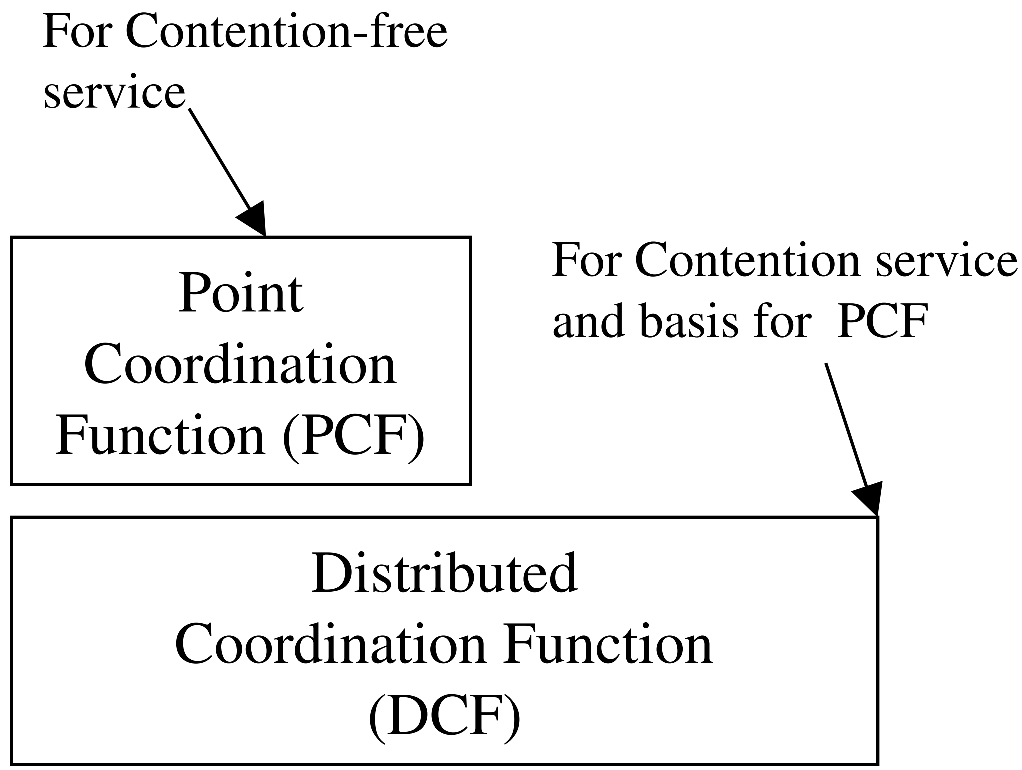
- In the Ethernet, carrier sense multiple access/collision detection (CSMA/CS) is used for contention-based medium access
- In WLAN, carrier sense multiple access with collision avoidance (CSMA/CA) is used instead, because detection is difficult. For example, hidden node problem

hidden node problem example: STA 1 and 3 can't "hear" each other



Access Methods in IEEE802.11

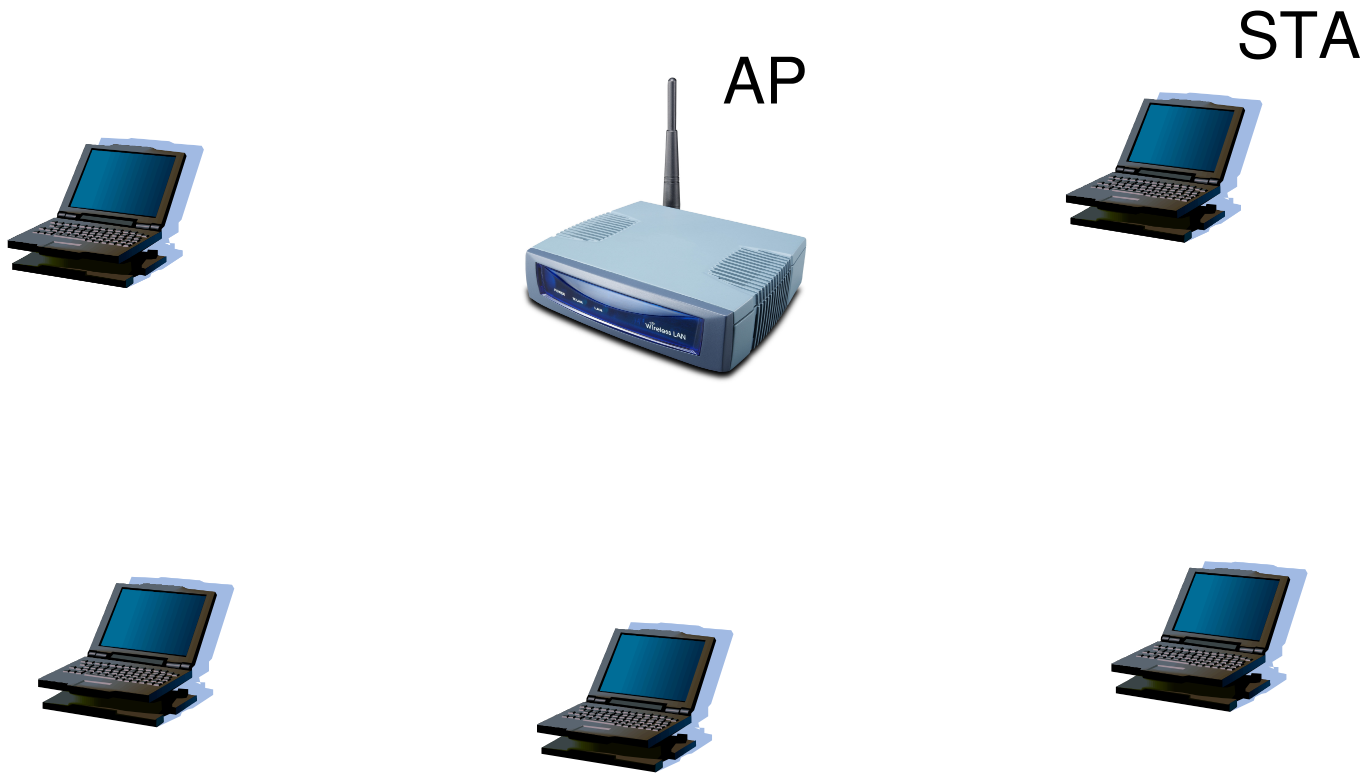
- DCF (CSMA/CA)
- PCF



Simulation requirements

- One AP and a specified number of voice STAs in the network
- voice connections are active in all STAs
- PCM over RTP over UDP over IP over LLC over WLAN
- DCF is the access method

Network topology in simulation



Protocol stack

PCM G.711
RTP
UDP
IP
LLC 802.2
MAC 802.11
PHY 802.11b

Functionality included in the simulation

- DCF
- Random erroneous frames
- Buffering packages
- Retransmission in case of collision or erroneous frame
- Discard package after maximum retransmission
- Only data and ACK frames considered

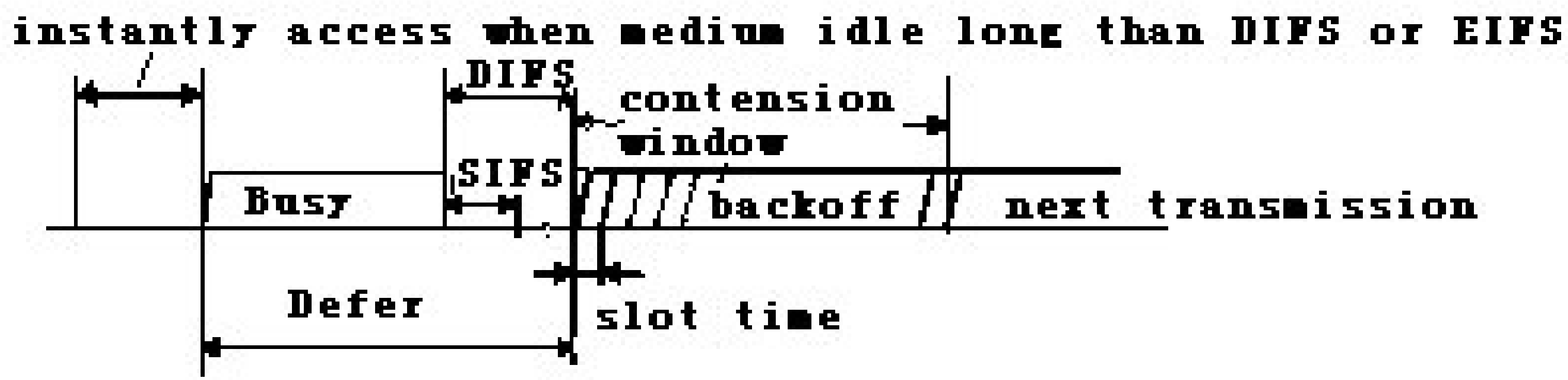
Functionality not included in the simulation

- RTS/CTS (countermeasure against the hidden node problem)
- Fragmentation
- Error in the ACK

Introduction to the DCF

- Frame can be transmitted immediately after:
  - Medium idle >= DIFS follows a correct frame
  - Medium idle >= EIFS follows an error frame
- when medium busy, frame sent after actions:
  - Wait until the end of current transmission
  - Medium idle for DIFS
  - Invoke backoff procedure (random backoff)

Timing relationship in the DCF



Distribute reservation time

- The Exchange of RTS and CTS prior to data frame
- The Duration/ID field in the directed frame

Backoff procedure

- Random backoff timer
  - Timer=random(CW) * SlotTime

  CW (new) = CW (old) *2+1 when failed frame

Frame length

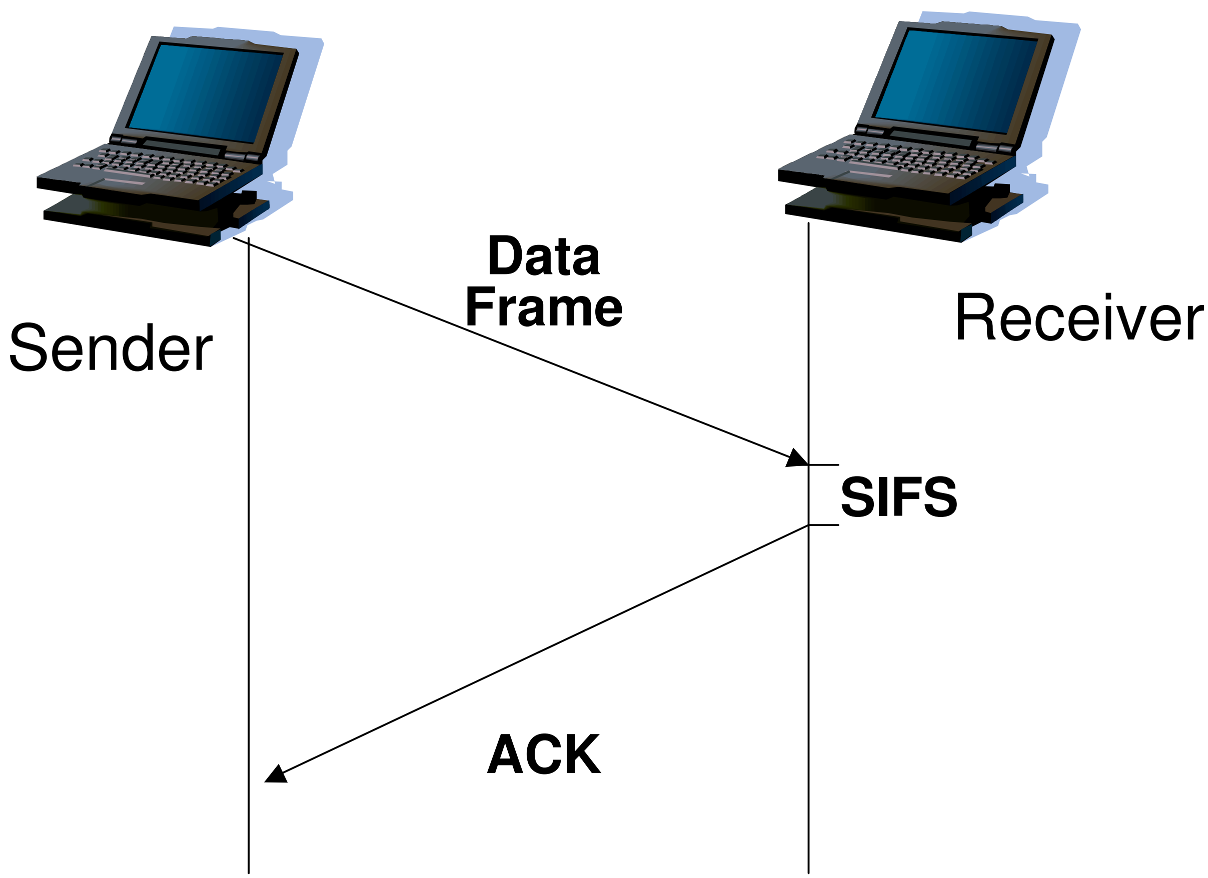
- Length of PSDU=PCM frame + RTP header + UDP header + IP header + LLC header + MAC header and PCS =160+12+8+20+4+34=**238** bytes
- ACK frame length =**14** bytes
- PLCP preamble and header =**24** bytes

Time spacing in simulation

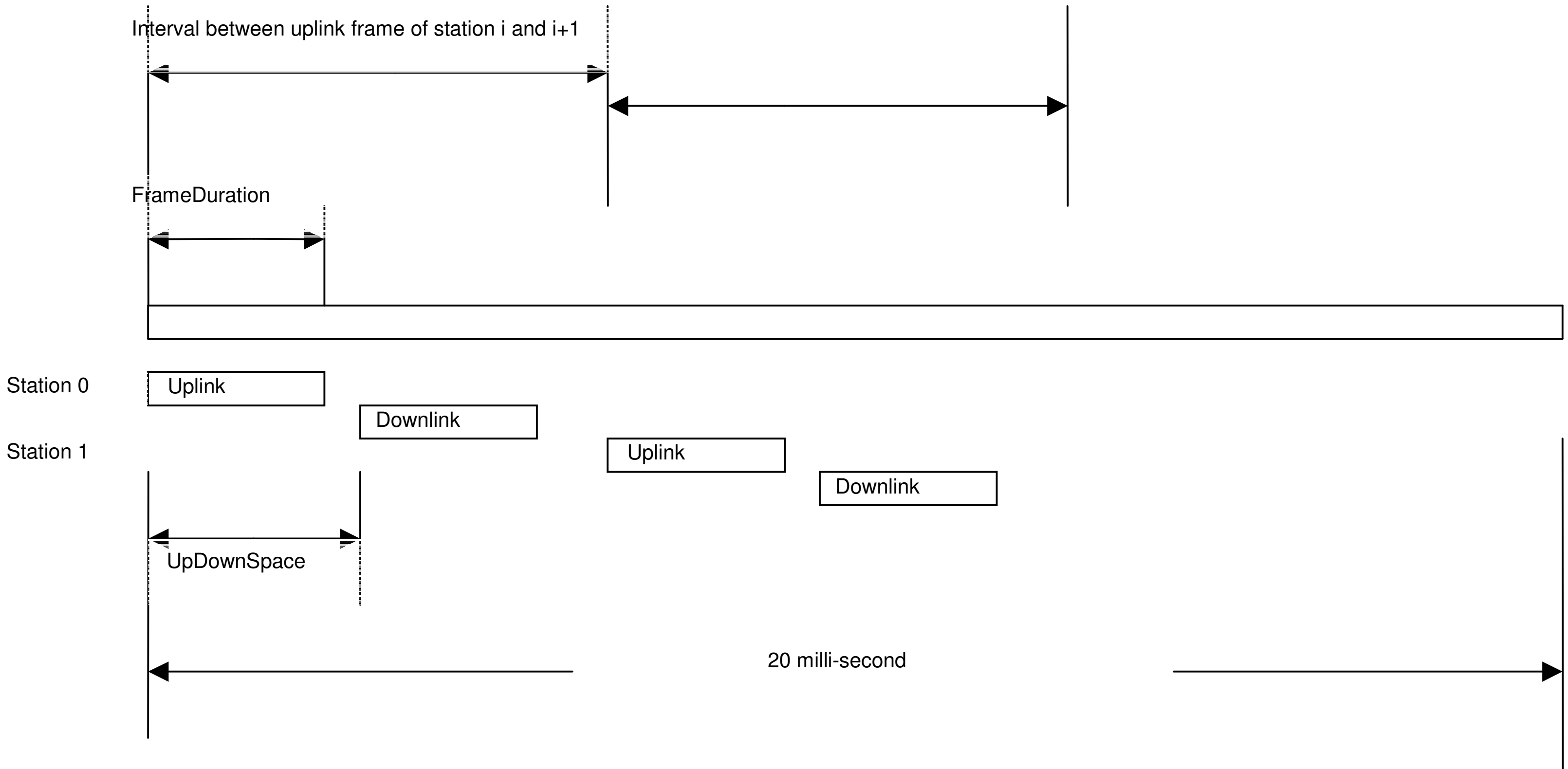
- SlotTime =20 μs
- SIFS= 10 μs
- DIFS = 50 μs
- Data frame duration = 365.09 μs
- ACK frame duration =202.18 μs

Medium occupation ≈ 630 μs

Sequence of sending frame and receiving the ACK



Frame sequences



Results

- Main parameters
  - Number of stations
  - Max retransmission time
  - Downlink buffer size
- Criteria of performance evaluation
  - FER <=0.08
  - Delay < 20 ms

Voice and data traffic

- Voice traffic is delay-sensitive, but less loss-sensitive
- Data traffic is not delay-sensitive, but more loss-sensitive

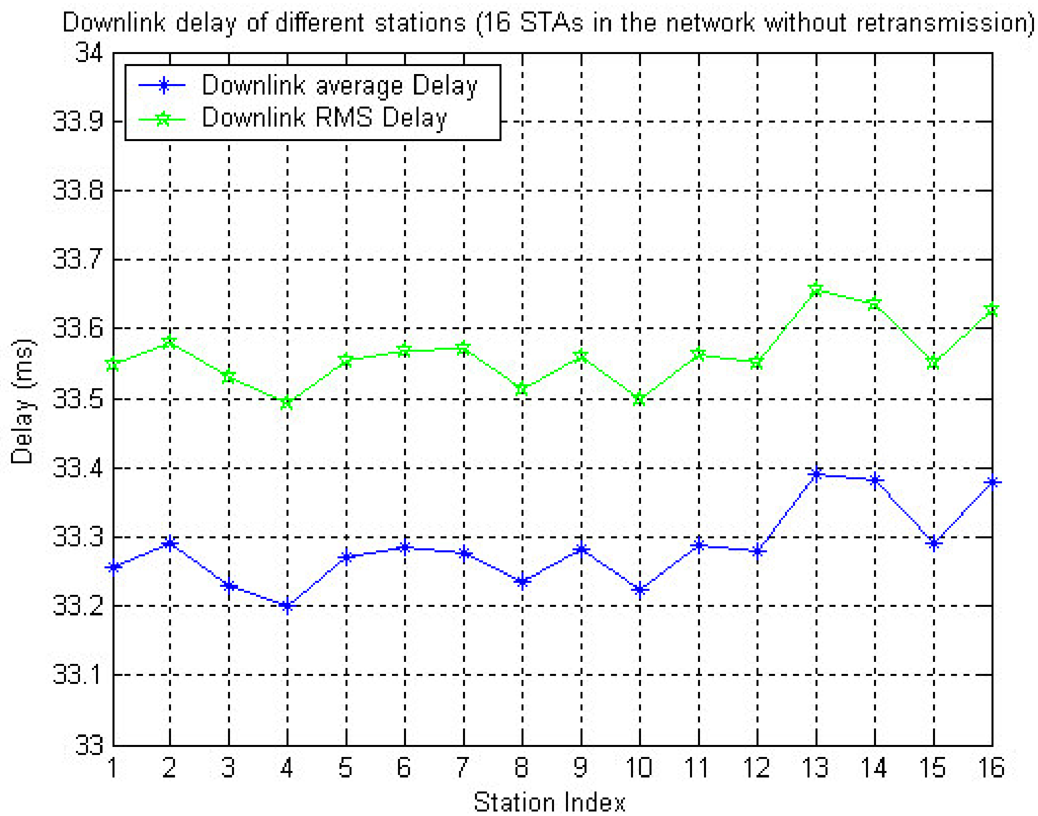
Voice traffic characteristic (1)—end-to-end delay

- End-to-end delay for voice (ITU G.114)
  - 25 ms: without echo cancellers
  - 150ms: with echo cancellers for excellent quality voice
  - 400ms: with echo cancellers for acceptable quality voice
- In practice
  - End-to-end delay lower than 250 ms
  - Buffer and queuing delay less than 20 ms

Voice traffic characteristic (2)—Delay Jitter

- the variation over time from point to point
- Critical in voice over IP—large delay jitter causes bad quality
- **Jitter buffer** is a solution
  - Without jitter buffer, standard deviation of 0.5ms causes voice quality unacceptable
  - with jitter buffer depth of 10ms, std of up to 13 ms is still acceptable

16 stations, no generated error frame, no retransmission (Delay)



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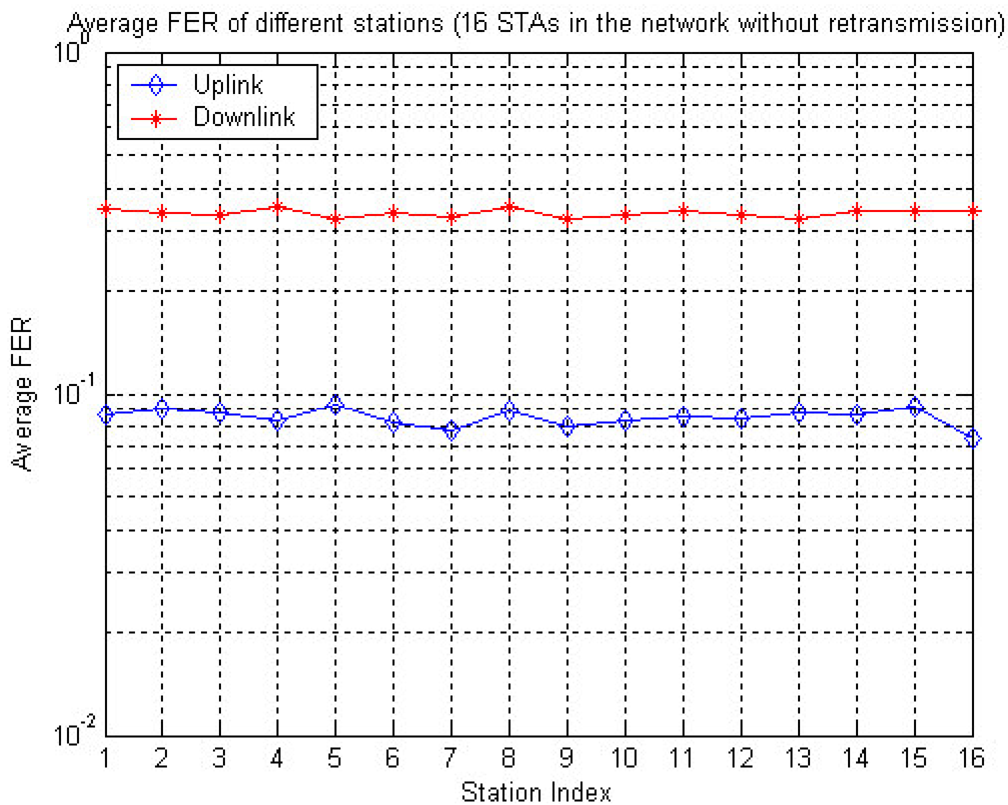
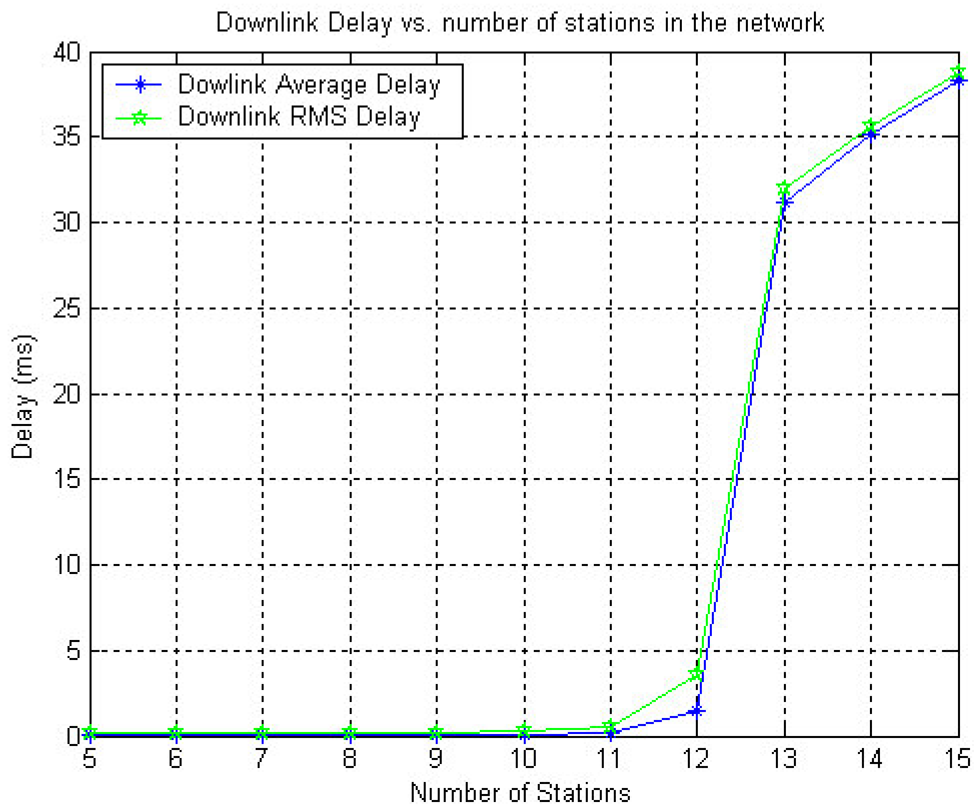
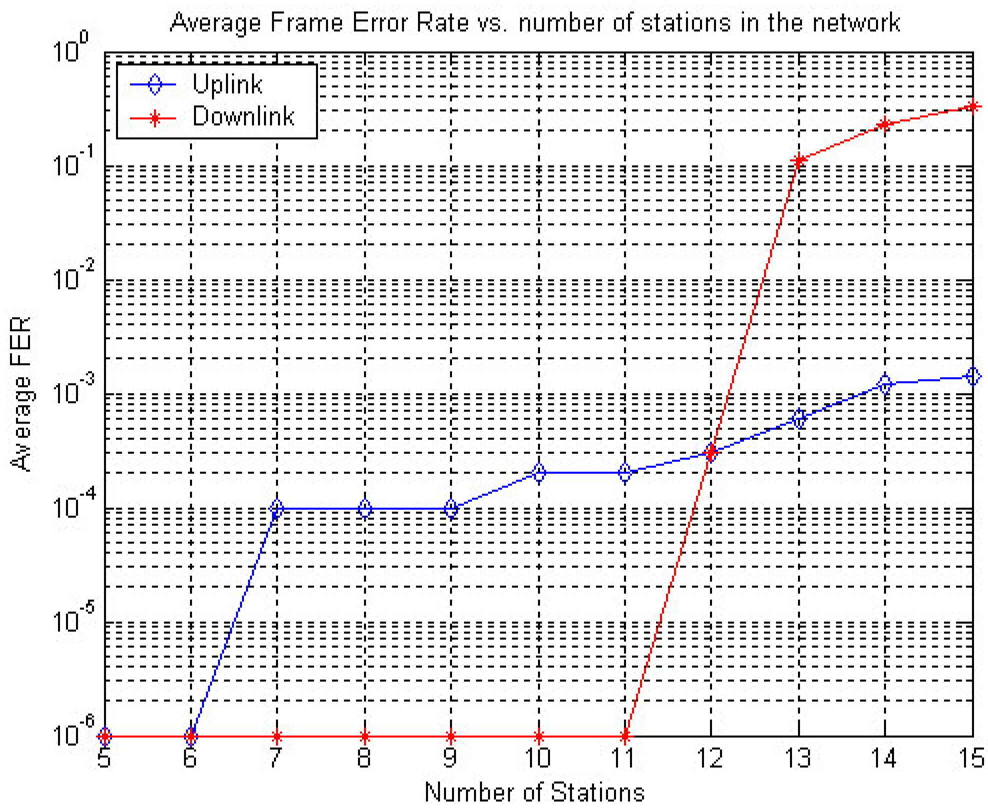


Retry limit=3, DLbuffer=20, Generated FER=0.03 (DL delay)

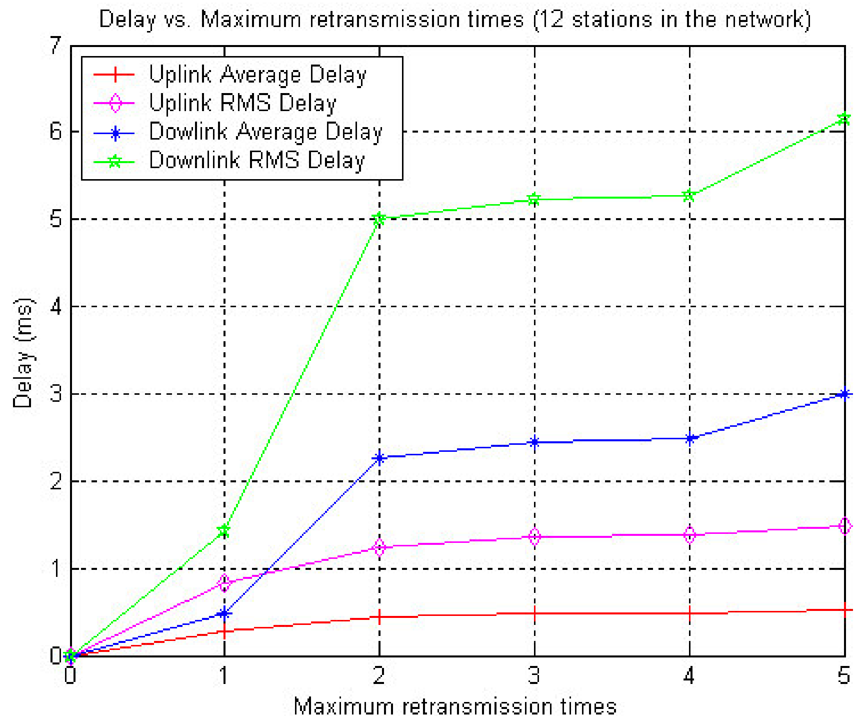


Retry limit=3, DLbuffer=20, Generated FER=0.03 (FER)



Effect of max retry times (Delay)

- One retransmission is good enough



Effect of max retry times (FER)

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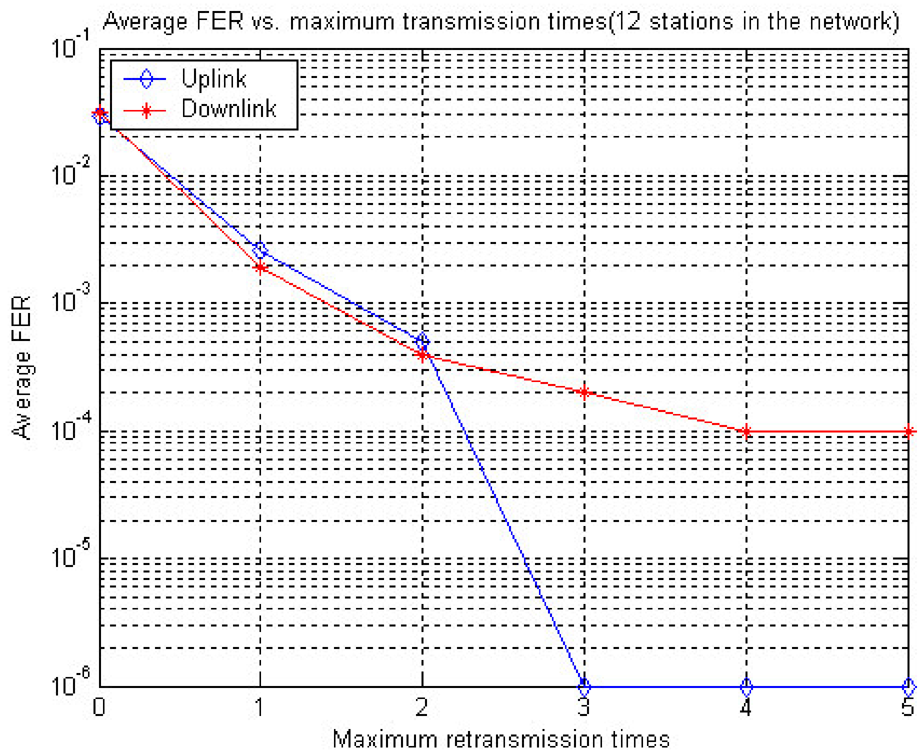
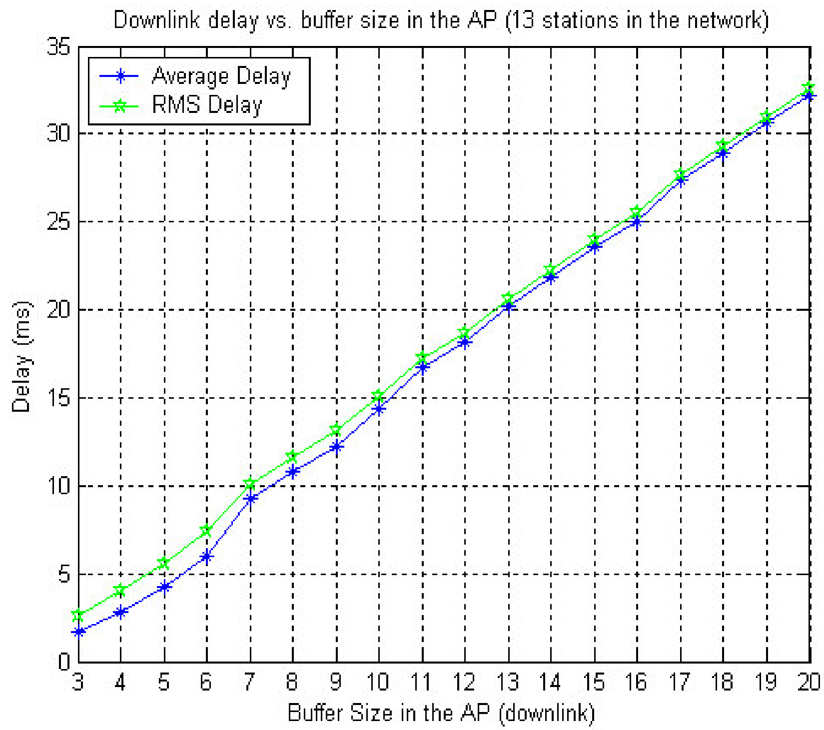


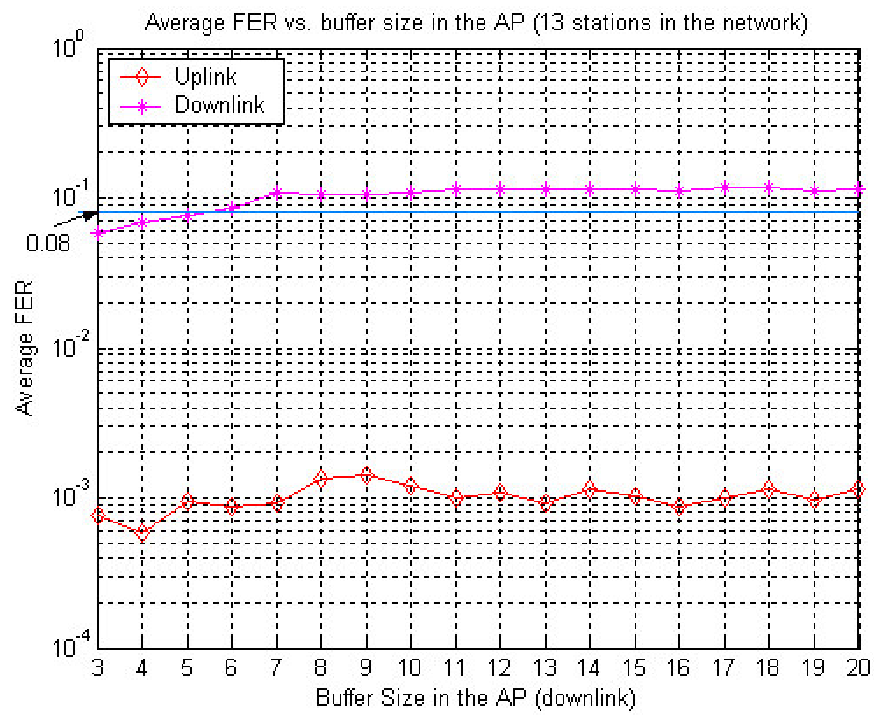
Effect of DL buffer size (1)

- Too many buffers degrade performance when many stations coexist

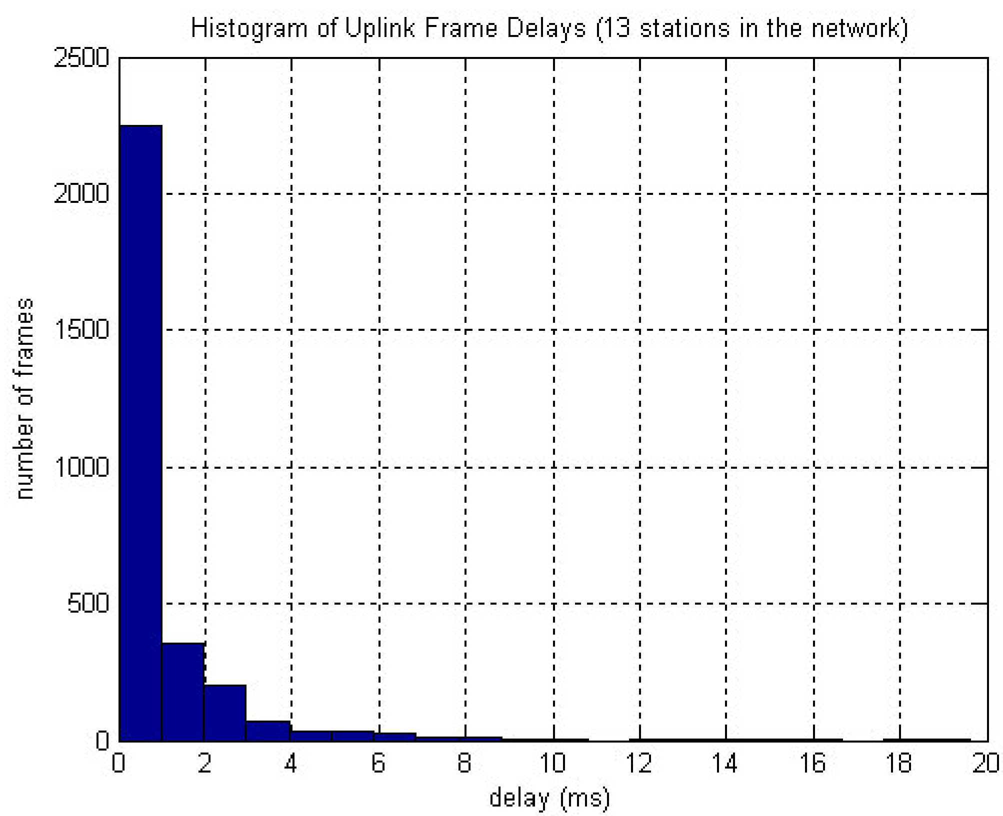


Effect of DL buffer size (2)

- Too many buffers degrade performance when many stations coexist



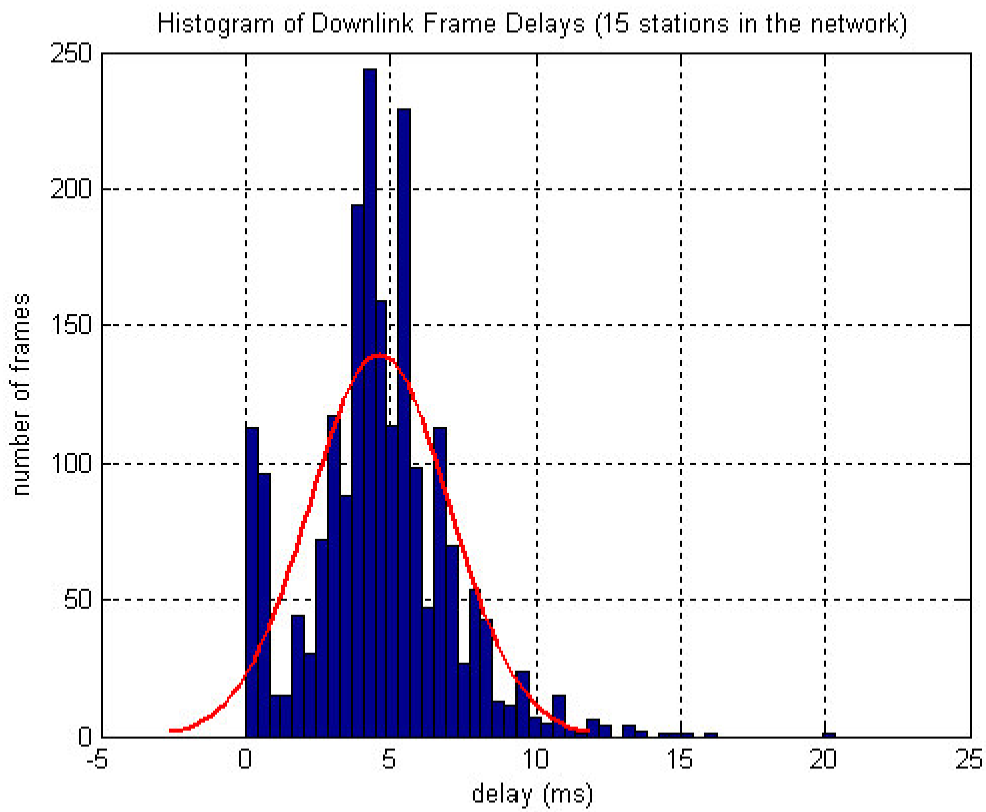
Histogram


Histogram of Uplink Frame Delays (13 stations in the network)
number of frames
2500
2000
1500
1000
500
0
0
2
4
6
8
10
12
14
16
18
20
delay (ms)

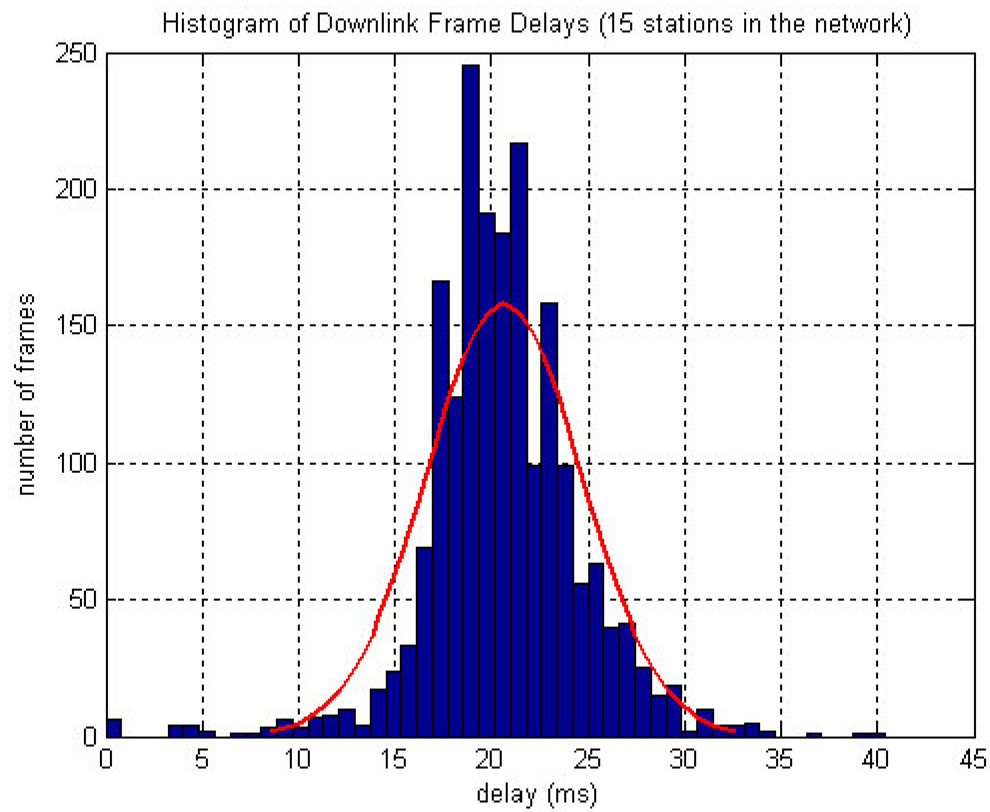
Delays vs. dl buffer size (15STAs)

Buffer size	mean delay (ms)	rms delay (ms)	delay std
3	4.6293469	5.2227234	2.418424
4	6.5597876	7.061593	2.615057
5	8.6120648	9.1063055	2.959944
6	10.5578625	10.9692847	2.976732
7	12.7835181	13.1556555	3.107608
8	14.6591895	15.1096692	3.662801
9	16.7427185	17.1557153	3.742558
10	18.5278369	18.921145	3.838674
11	20.6315747	21.0201025	4.023772
12	22.5343066	22.8693042	3.900894
13	24.5609399	24.9396167	4.330496
14	26.7404883	27.0810254	4.28211
15	28.72979	29.0997266	4.626213
16	30.9728638	31.2636914	4.255358
17	32.3369556	32.7299878	5.058135
18	34.8990161	35.2368237	4.868602
19	36.555188	36.889519	4.956494
20	38.5487207	38.9030737	5.239991

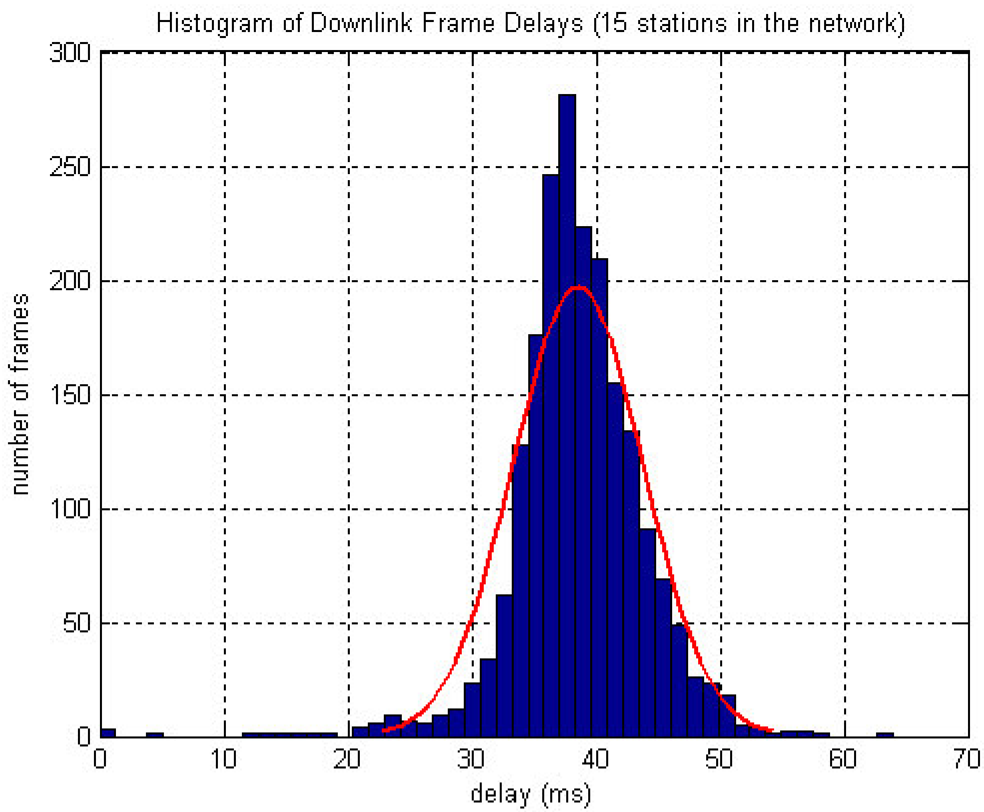
Buffer size=3



Buffer size=11



Buffer size=20



Conclusions

- Theoretical capacity is 15 stations, practical capacity is less than this value because of error frames
- One retransmission for error frame is a good solution to improve performance
- Large buffer size degrades performance under a special configuration

Future work

- Other voice traffic over WLAN
- Random frame generation
- Data and voice coexist

Thank you!

Any Question?