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Dynamic Codec with Priority for Voice over IP in WLAN

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VoIP in WLAN … is the next Killer app

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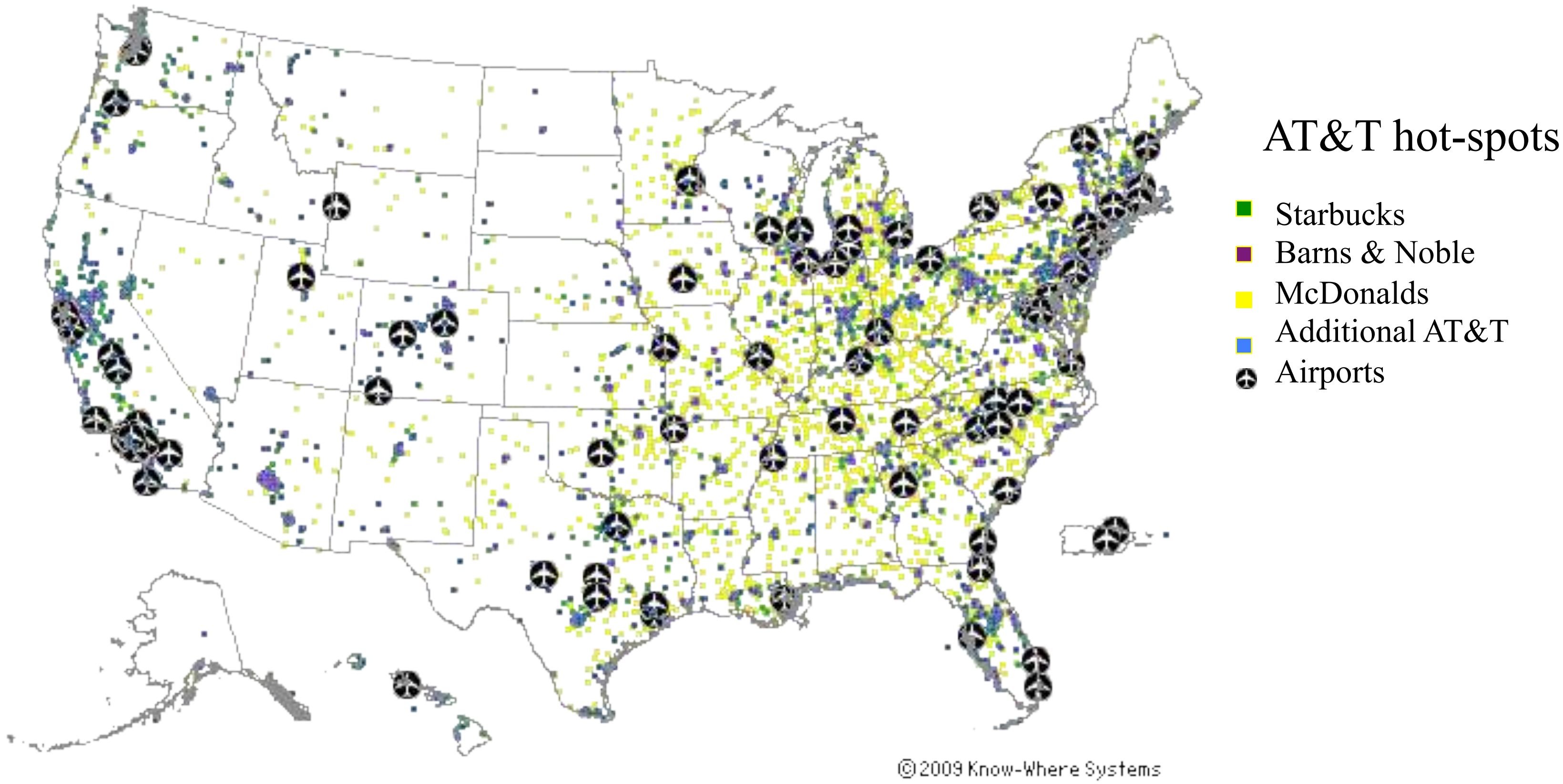

VoIP in WLAN … is the next Killer app, but …

- Limited voice over IP capacity (number of calls) in IEEE 802.11 infrastructure WLAN
  - An 802.11b WLAN with 11Mbit/s should be able to support 11/0.015 ≈ 733 voice calls (G.729, 8Kbit/s each way)
    - Actual capacity: 5 to 7 calls using the same codec
  - Capacity limited by channel access mechanism rather than bandwidth
    - With an increasing number of voice calls, the probability of the AP winning the channel contention is decreasing
      - 1/(N+1) [AP] vs. N/(N+1) [wireless nodes]
    - Access point (AP) becomes bottleneck in WLAN
      - Packet loss and long delays occur when network becomes saturated

> And all this in ideal conditions, e.g. no cross traffic, an ideal channel, etc.

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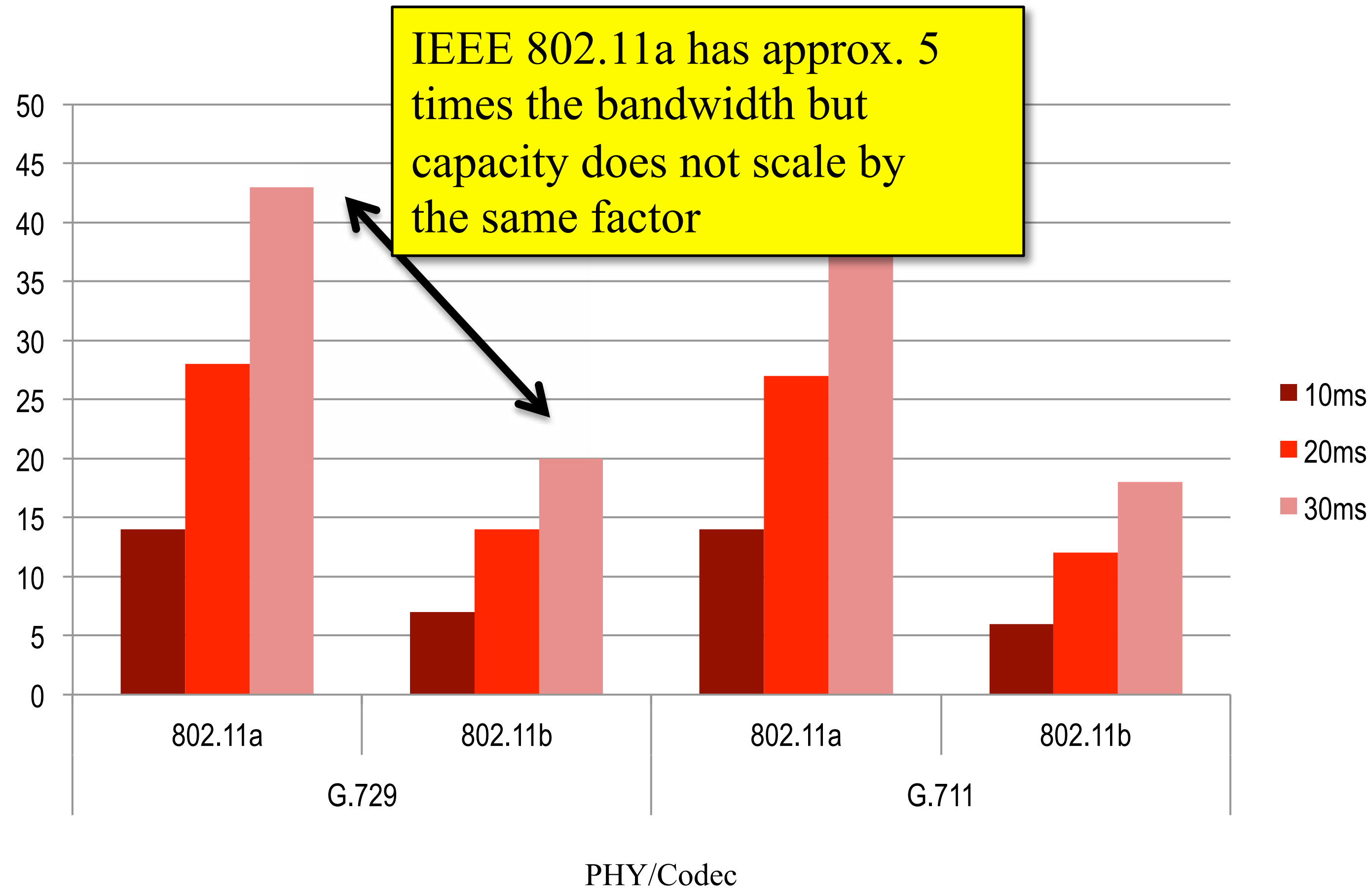

How do we fix it ??

How do we fix it?

- Different proposals ranging from increased bandwidth, new medium access control, new protocols or MAC parameter optimization
  - Problems
    - Performance gain can be achieved, but at what level of call quality (individual call/all calls)?
    - Assume *static* voice codec
      - No adjustment to changing network characteristics
  - Solution
    - The solution is twofold: a) *dynamic* voice codecs, b) access prioritization

Dynamic codec with priority

- *Dynamic* voice codecs
  - Codec/VoIP application monitor network characteristics
    - Packet loss, delay, jitter, …
  - Based on feedback, the codec/VoIP application adjusts codec settings, e.g. sampling rate, packet rate, DTX, etc.
  - Example: SILK used in Skype V.4
- Channel access prioritization
  - Increase priority for voice over IP data at the AP
  - Use of IEEE 802.11e protocol (EDCA)
    - Different values of CW_{min}/CW_{max} parameter (increased channel access frequency = guaranteed throughput)

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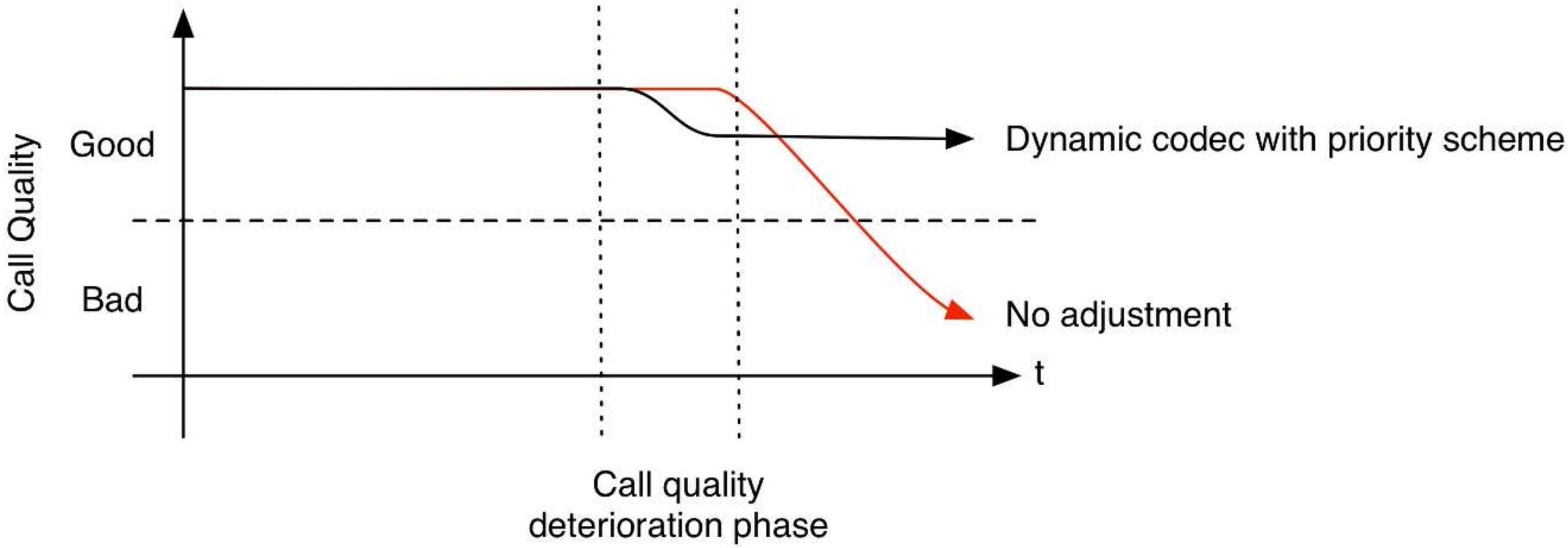
Dynamic codec with priority

- During periods of high contention, encourage user to switch to a lower quality codec
  - Changes sampling rate and payload size,
    e.g. G.711, 10 ms (R = 93) to G.729, 20 ms (R = 84)
- Provide incentive to switch by placing lower quality calls into higher priority access queue at AP
  - Encourages a less aggressive behavior ➔ reduced contention
- Benefits to the user
  - Continue with call at reduce quality, rather than not being able to maintain the call
  - Guaranteed throughput of lower quality call

Dynamic codec with priority



Additional benefit: Voice capacity increase if more users switch

Implementation/Approach

- Analytical model and simulation
  - Two traffic classes at AP
    - Differentiated by contention window size only

Codec	Quality level	*R*-value
G.711, 10 ms	High	*R* = 93 (Excellent)
G.729, 20 ms	Medium	*R* = 84 (Good)
G.723, 30 ms	Low	*R* = 79 (Fair)

    - Simple recursion to obtain conditional collision probability at AP
  - Evolved around fixed-point formulation
  - Capacity reached if packet loss exceeds 2%
  - Quality assessment using ITU-T E-model
    - R = [0, 100] = [low quality, high quality]

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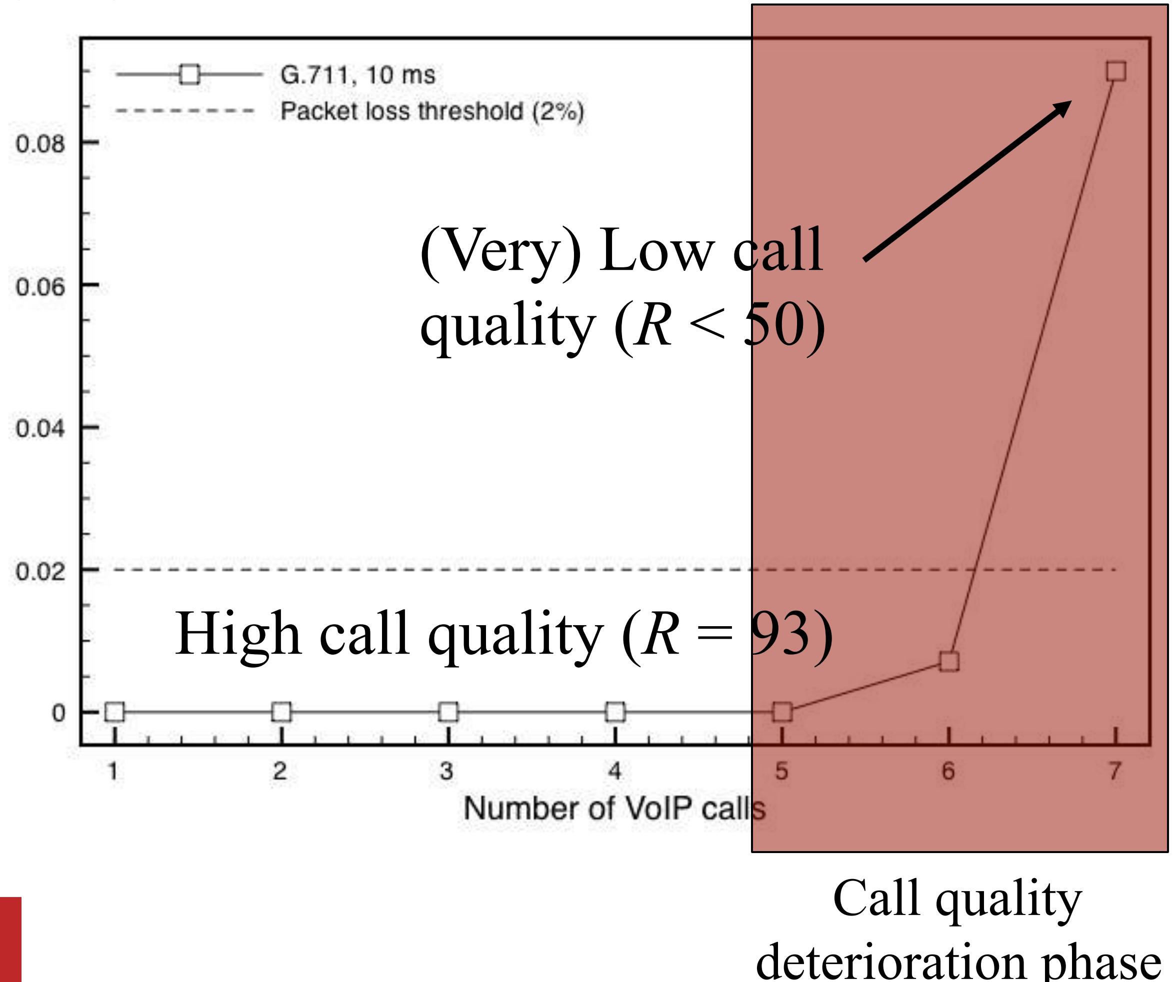
Results & Discussion

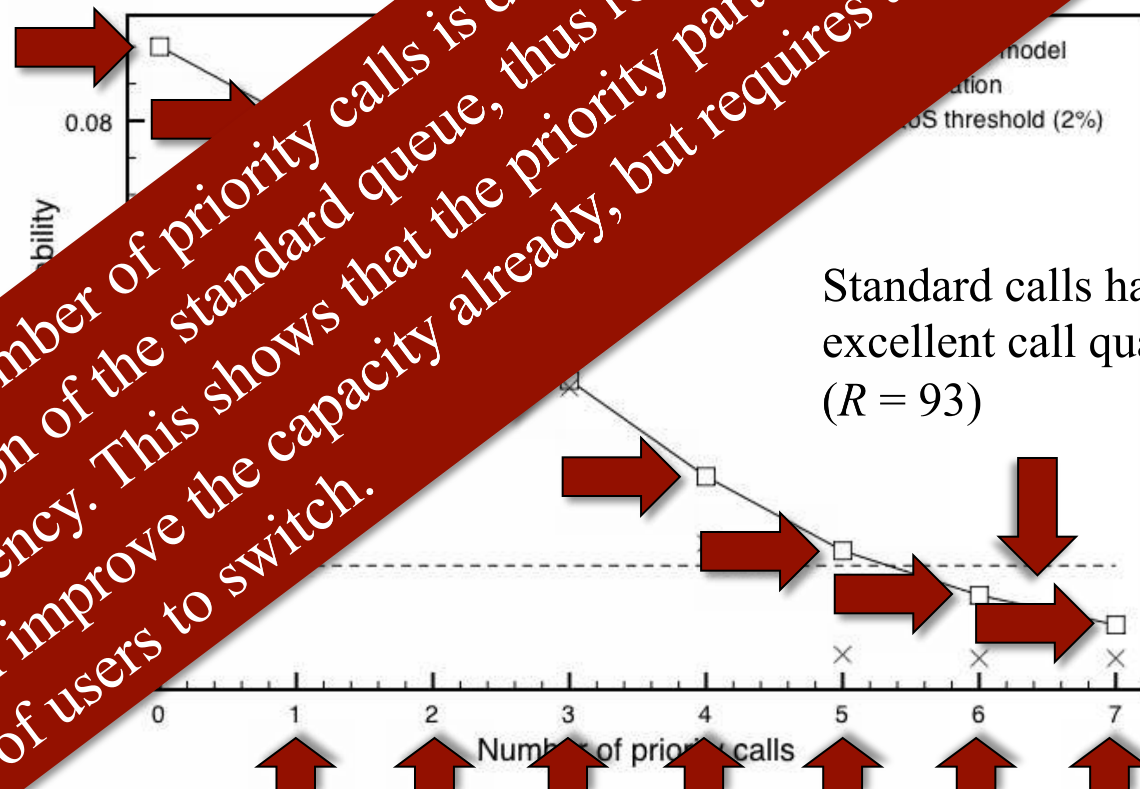
- Baseline



Results & Discussion

- Scenario 1 (all calls stay G.711)

Initially 7 G.711 voice calls



An increasing number of priority calls is decreasing the queue utilization of the standard queue, thus reducing loss. This shows that the priority part of our access frequency. cheme can improve the capacity already, but requires a number of users to switch.

Standard calls have excellent call quality ($R = 93$)

Standard calls h poor call q ($R < 50$ prio good c ($R = 84$)

Number of priority calls is increasing, while number of standard calls is decreasing

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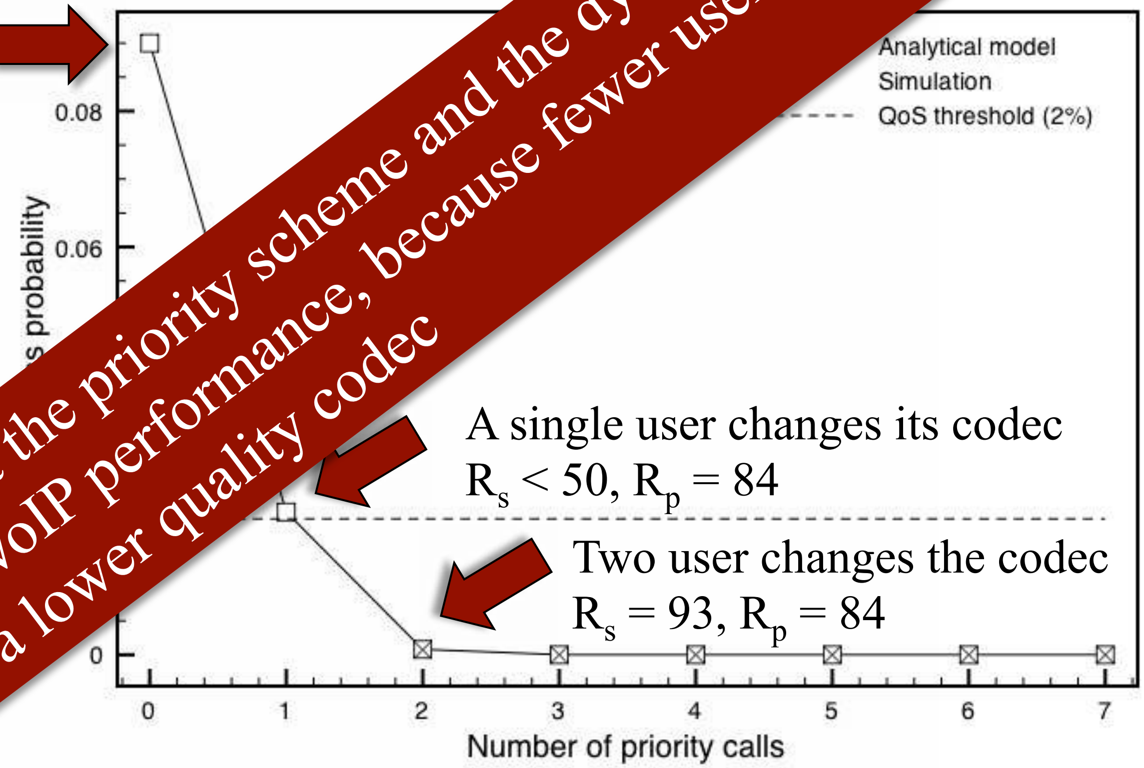
Results & Discussion

- Scenario 2: High quality ➔ medium

Initially 7 G.711 voice calls



A single user changes its codec
$R_s < 50$, $R_p = 84$

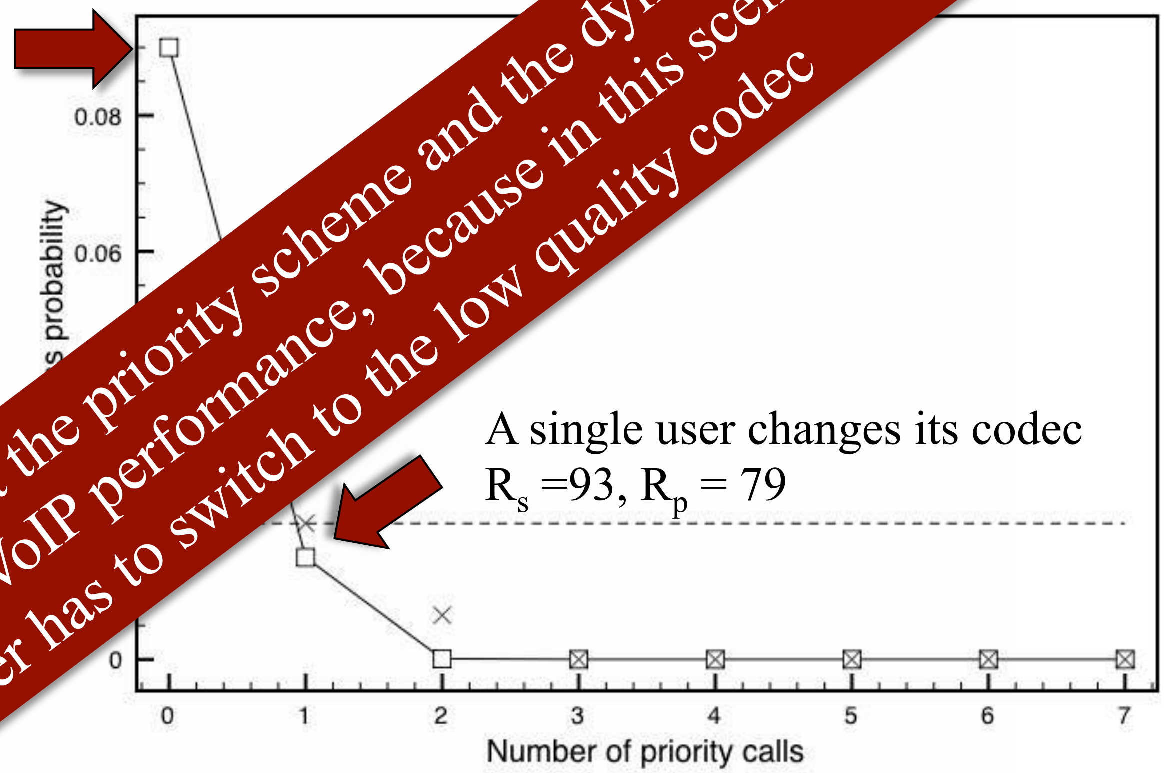
Two user changes the codec
$R_s = 93$, $R_p = 84$

This shows that the priority scheme and the dynamic codec improve the VoIP performance, because fewer users have to adapt to a lower quality codec

Results & Discussion

- Scenario 3: High quality ➔ low quali

Initially 7 G.711 voice calls

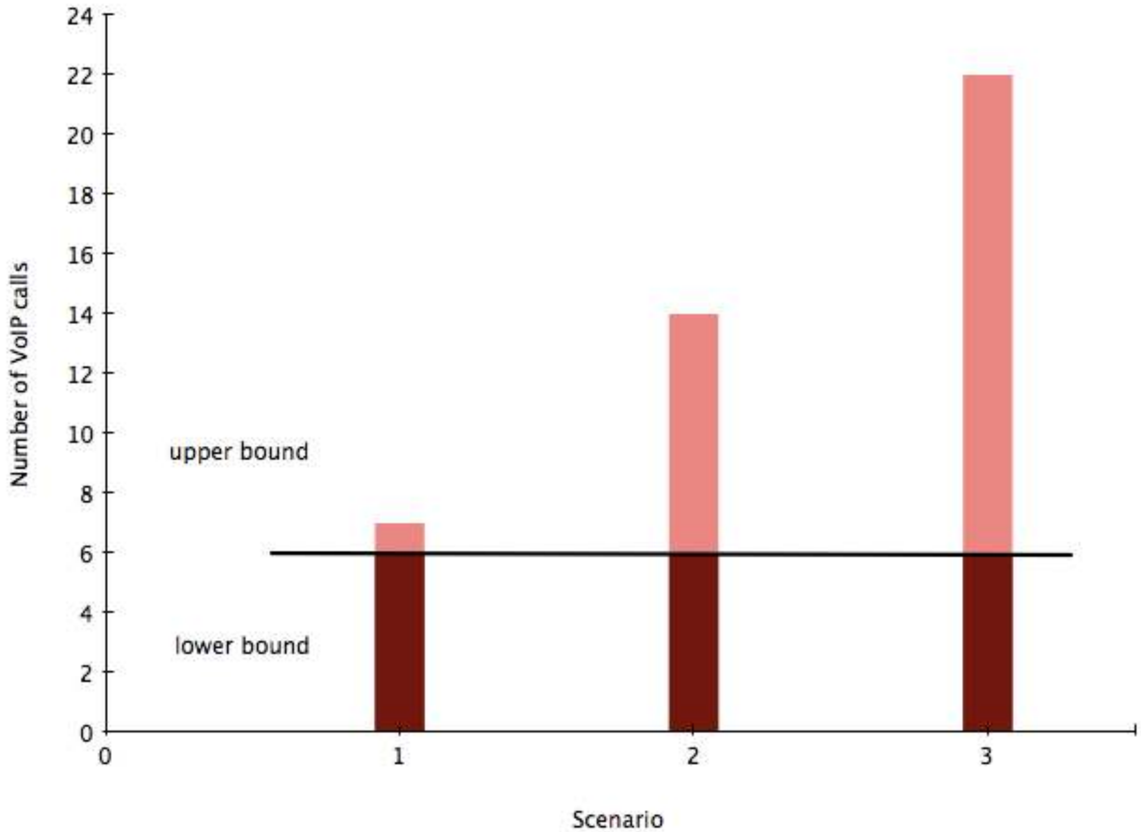


This shows that the priority scheme and the dynamic codec improve the VoIP performance, because in this scenario only a single user has to switch to the low quality codec

Results & Discussion

- Initially the WLAN can only support 6 voice calls using the G.711 voice codec (high quality)
  - Using the proposed scheme we showed that with an increasing number of priority calls an additional call can be supported
  - Once the required number of calls have switched to a new codec, e.g. all calls experience no loss, additional calls can be added to the WLAN

Results & Discussion



Total capacity gain of 200+% can be achieved

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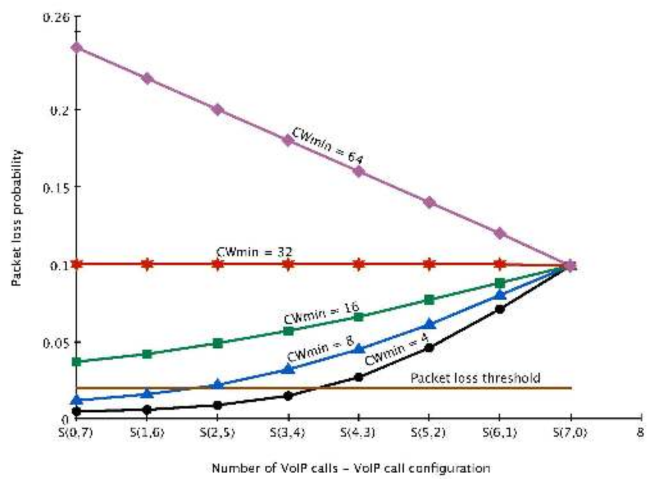
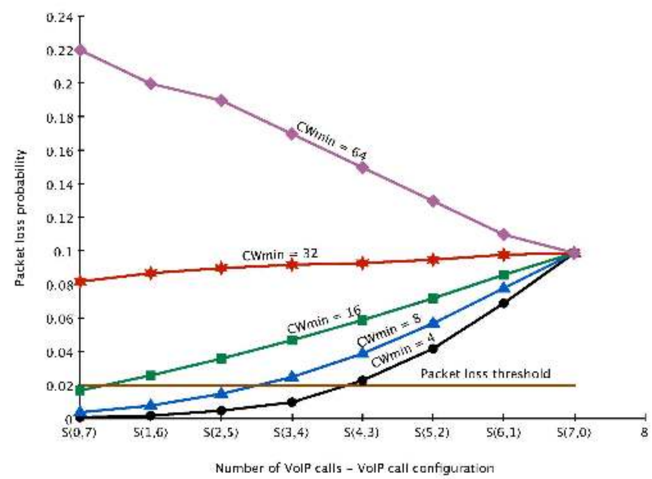
Results & Discussion

- Can the performance be increased with different settings of CW_{min}, CW_{max}?
  - Yes and No
    - No: Larger CW_{min} will no increase the capacity
    - No: Changes to CW_{min} has no impact when sampling rate is changed
    - Yes: Minor increase with smaller CW_{min} in priority queue





Summary/Conclusion/Contribution

- Proposed a novel scheme to reduce the overall contention in a highly congested WLAN
  - Scheme based on dynamic voice codecs and traffic priority
- Improved voice capacity while maintaining an acceptable voice call quality
  - Capacity gain off between ~ 16% and ~ 200% (depending on (lower quality) codec)
- The analytical model versatile enough to be used for other traffic types
  - Captures internal collision
  - Dual-queue
- Traffic differentiation can be implemented using the DIFFUSE tools developed at CAIA

Questions

- Ask now
- Ask later
- Ask via Email (kstoeckigt@swin.edu.au)
- Ask via Skype (k.stoeckigt)
- Ask via Twitter (@kstoeckigt)