SIMULATION ANALYSIS OF MULTIMEDIA STREAMS TRANSMISSION IN IEEE 802.11 NETWORKS

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Introduction

This summary presents a simulation analysis of the MAC protocol described in the IEEE 802.11 standard in order to verify the possibility of multimedia streams provision. This standard specifies a wireless local computer network. The key issue is to design a WLAN in such a way to secure the effective transmission for the existing applications and those which appear in near future like e.g. HDTV, video-conferencing or video on demand. It seems that multimedia applications, which include various types of information (video, pictures, sounds and text), will be the most attractive in the near future. They possess different characteristics and network requirements.

An intensive simulation was carried out to test the network efficiency. The network was configured to 1 and 2 Mbit/s medium capacities. Various network configurations working at different modes were simulated. The simulations were performed in some stages. The DCF (*Distributed Coordination Function*) efficiency tests were made in the first stage. Connections between DCF and PCF (*Point Coordination Function*) were checked in the second one. Implementation possibility of multimedia streams was examined in the third one. Telephony (2×32 kbit/s full duplex service after ADPCM compression), video-telephony (2×64 kbit/s full duplex service), high quality audio CD transmission (256 kbit/s), video-conference (p×32 kbit/s sound + asynchronous bandwidth for transmission of images, graphics and text) and MPEG-1 (Lambs, Mr Bean, Race - 3 files encoded at University of Wuerzburg) as isochronous services were considered. All others services were of asynchronous type.

A large volume of data was gathered which has allowed us to determine the quality of service and number of acceptable streams (in terms of the throughput and delay). The summary presents these results and key conclusions, which give some hints how to integrate asynchronous and isochronous services in an IEEE 802.11 network. The simulations results determined the average delay for asynchronous traffic. Basing on the obtained results, a number of conclusions were drawn on the network behaviour at very variable, sometimes extreme, situations.

Analysis

The medium access protocol for IEEE 802.11 wireless networks incorporates two access methods. The first method is mandatory and based on the CSMA/CA protocol. It is called the Distributed Coordination Function (DCF). The second one, the Point Coordination Function (PCF), is optional and used for provision of time bounded services. The knowledge of tradeoffs between DCF and PCF functions allow us to determine the network capability of isochronous service transfer necessary for full implementation of multimedia streams. The attention was laid on the following issues:

- direct reflection of isochronous services in parameters used in simulations;
- definition of network capacity in terms of the number of implemented services and their types;
- examination of isochronous services behavior in an IEEE 802.11 network and their coexistence with asynchronous ones.

All isochronous services require delivery of consecutive frames in proper time gaps. This is caused by the necessity of transmission fluency and synchronisation. During simulation two values of delays occurring in the realisation of isochronous services were assumed: 25 and 150 ms. Taking these remarks in mind, the proper network parameters were been searching. The following goals attained our attention:

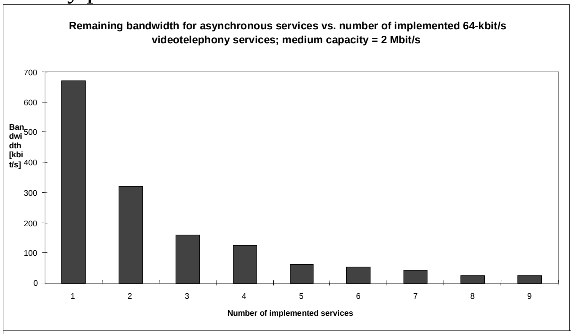
- ensuring sufficient bandwidth and delay below a given threshold for isochronous services;
- minimalization of CFP period (or maximalization of remaining bandwidth for asynchronous services);
- maximalization of data field length used for isochronous services (in order to reduce the number of cycles necessary to obtain the access to the medium).

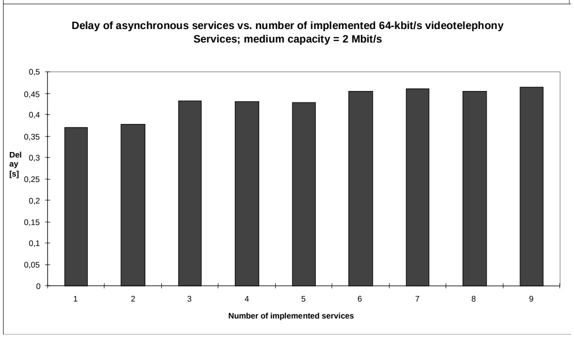
The number of services possible to implement with the assumed quality was rather not large (especially for 1 Mbit/s medium capacity). Table 1 presents the maximum numbers of isochronous services realised simultaneously in a network of fixed capacity and at the maximum delay.

Table 1. The number of streams possible to implement as functions of assumed capacity and delay

Type of service	1 Mbit/s	2 Mbit/s
Audio-CD - 25ms delay	**2** simplex	**4** simplex
Videotelephony - 25ms delay	**3** duplex	**7** duplex
Videoconference – 25ms delay	**9** simplex	**20** simplex
ADPCM telephony - 25ms delay	**5** duplex	**10** duplex
Audio-CD - 150ms delay	**2** simplex	**4** simplex
Videotelephony – 150ms delay	**3** duplex	**9** duplex
Videoconference – 150ms delay	**14** simplex	**36** simplex
ADPCM telephony - 150ms delay	**7** duplex	**18** duplex

Performed simulations allowed us to determine the amount of bandwidth for asynchronous service. Thus this enabled us to add some asynchronous services to the existing isochronous ones. Simulation runs also allowed us to determine the average delays (measured from the arrival of a call to the moment of proper ACK reception) arising in asynchronous transmission. In this summary plots obtained for videotelephony, at fixed delay of 150 ms and 2Mbit/s medium capacity are only presented.





Transport of MPEG information brings a problem of wireless link capacity/buffer size decision (due to bursty nature of VBR MPEG). We have implemented a model of MPEG encoder in COMNET III simulation environment. The detailed structure of MPEG I, P, B frames is modeled by using a concept of triggered session sources. Each source has detailed information about the probability density function, which is built-in using stored histogram of frame sizes. These histograms were obtained by thorough scanning of MPEG files. Obtained results allow us to draw the conclusions about link capacities and buffer sizes. We assumed following values:

- **Lambs** - 230 kbit/s link; 65000 bytes buffer size;
- **Mr**. **Bean** - 480 kbit/s link; 70000 bytes buffer size;
- **Race -** 790 kbit/s link; 60000 bytes buffer size.

The simulation results are presented in Table 2.

Table 2. The number of MPEG streams possible to implement, remaining asynchronous bandwidth, mean delays and size of Contention Free Period (variable to maximize asynchronous bandwidth)

	Number of MPEG streams	**Mean isochr. delay [s]**	**Remaining asynchr. bandwidth [kbit/s]**	**Mean asynchr. delay [s]**	**CFP [s]**
1 Mbit/s	Lambs - **3**	0.019	121.9	0.53	0.43
	Mr. Bean - **1**	0.008	363.2	0.55	0.28
	Race – **1**	0.006	75.3	0.51	0.46
2 Mbit/s	Lambs - **7**	0.021	73.4	0.55	0.48
	Mr. Bean - **3**	0.010	217.3	0.48	0.43
	Race - **2**	0.006	99.1	0.51	0.47

The results obtained from simulations allow us to draw the following remarks:

- The network is scalable when considering the capacity for asynchronous services. However, doubling the nominal network capacity does not mean doubling the remaining capacity. Scalability may also be observed in the maximal number of realised isochronous services for both capacities;
- Delay incurred by an asynchronous services increases when the number of services increases, however, for all cases it exhibits saturation at level of 0.4-0.6 s;
- For both assumed isochronous delays (25 and 150 ms), the network of a given type and the number of implemented services, at a given nominal medium capacity, leaves the bandwidth for asynchronous traffic of same magnitude;
- Further increase of isochronous delay (above 150 ms) does not implicate further increase of the number simultaneous isochronous services provided, since then we reach the nominal capacity of the network;
- We should use the adequate decoder buffer size to smooth the bursty nature of MPEG streams, but it affects the total transmission delay (in our case up to 2s, so we get near Video on Demand);
- When implementing different isochronous services then some bandwidth remains for use to provide asynchronous services. These services may include data, graphics, text transmission, WWW, FTP, Telnet and so on. This allow us to integrate asynchronous and synchronous services;
- If the remaining capacity is not sufficient to immediate realisation of desired asynchronous service (neglecting access time) then the time of asynchronous service realisation is lengthened.

Conclusions

The objective of the presented work was to analyse an IEEE 802.11 network in terms of provision of multimedia streams. Basing ourselves on simulation, we defined which types of services can be implemented at a given level of quality. The presented analysis led us to some important conclusions briefly presented below.

- The maximum throughputs are achieved for asynchronous services. Implementation of isochronous services causes a substantial reduction of asynchronous throughput.
- There is a possibility of integration voice, sound with both moving and still images. This allows for provision of telephony, videotelephony, videoconference and MPEG streams.
- The proper choice of parameters for isochronous services is the key issue since it allows us to achieve maximum throughputs. These parameters include the length of superframe, CFP period, length of data fields in frames.
- Parameters like the length of transmitted frames, number of accessing stations, number of hidden terminals, threshold for starting the RTS/CTS mechanism, type of backoff plays the role for asynchronous services.
- The network is scalable when providing asynchronous services, Scalability refers also to isochronous services but proportions are non-linear.