An evaluation tool for Wireless Digital Audio applications

Nicolas-Alexander Tatlas[1], Andreas Floros[2], and John Mourjopoulos[3]

[1] Audiogroup, Electrical Engineering and Computer Technology Department, University of Patras, 26500, Greece
ntatlas@upatras.gr

[2] Multimedia and Communications Group, Atmel Hellas SA, 26504, Patras, Greece
afloros@patras.atmel.com

[3] Audiogroup, Electrical Engineering and Computer Technology Department, University of Patras, 26500, Greece
mourjop@wcl.ee.upatras.gr

ABSTRACT

Despite the recent advances in the wireless networking technology, wireless multichannel digital audio delivery is not yet efficiently realized, because of the additional implementation issues raised for managing possible distortions introduced. In this work, a novel, open-architecture, software platform for evaluating wireless digital audio distribution is presented. This tool facilitates the assessment of real-time playback distortions induced by variable packet reception delays and packet losses, typically encountered in WLAN transmissions. Moreover, this platform can be also employed for producing audio streams corresponding to the wirelessly delivered digital audio, in order to investigate the audibility of such distortions.

0. INTRODUCTION

Broadband wireless technologies deployed recently can meet the requirements of bandwidth-demanding applications such as wireless reproduction of high definition audio. The development of fully interoperable, high-rate, wireless networking products is leading towards the integration of the WLAN computer-based technology with the Consumer Electronics (CE) technology, allowing users to conveniently share an increasing amount of digital media stored across different devices and locations. Given the multiple sources for multichannel digital audio (such as DVD and SACD) introduced in a typical modern home environment, the need for digital wireless multichannel audio streaming has become more demanding than ever. This need is even more pressing when the significant advantages of multichannel wireless speakers/receivers may be envisaged, functioning in a transparent to the user mode, in a WLAN home environment [1].

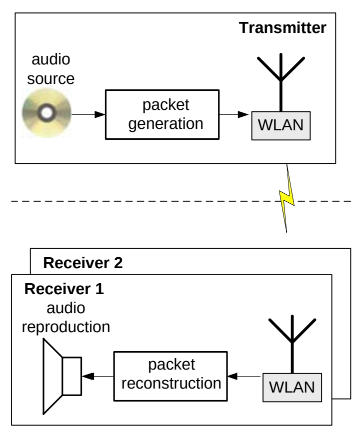

Fig. 1: Wireless digital audio system set-up

However, despite the advances on the data rates supported by the WLAN technology (i.e. theoretical 54Mbps throughput for the IEEE 802.11g protocol), wireless digital audio streaming cannot be yet efficiently implemented over WLANs in real-time, because of the additional application-specific issues raised for compensating possible reproduction distortions.

In order to evaluate such a system, a novel computer-based tool was developed and is presented here, consisting of two stages: (a) the pre-processing stage, where an appropriate 2-channel PCM digital audio test signal is created and then converted to a format which can be used as input to a Wireless Network Simulator (WNS) emulating the wireless data transmissions [2], following the rules of the desired wireless protocol and (b) a post-processing stage where the output from the WNS is processed and a new wave file is created, containing a "wirelessly reproduced" version of the original digital audio data. Distortion information for each receiver can be then extracted and analyzed from this audio file. It should be mentioned that, besides the digital audio test signal, any PCM-coded audio file can be used as input to the proposed platform in order to create its "reproduced" audio replica for the wireless setup. The above platform can be easily extended to accommodate compressed or other high-quality stereo and multichannel audio streams.

1. DESCRIPTION OF THE EVALUATION PLATFORM

1.1. Test Signal Generator

In order to test the wireless audio streaming system and to detect all possible distortions induced by the wireless data transmissions, a discreet-time periodic test signal is selected as the WNS input.

The period N_p must be chosen appropriately in order to ensure that every type of playback distortion (delay and data loss) can be clearly detected. The choice of N_p depends on the network's packet payload length L_p (in samples), as well as the total duration of the audio waveform T_{dur} (in samples). In order to avoid loss of phase information, it is necessary to ensure that if a number of consecutive packets are lost, this will not force the periodic test signal received to resume from its previous state.

In this work, a software application was developed for generating the appropriate test signal, given the networks packet payload length L_p and the total desired simulation duration T_{dur}.

1.2. Wireless network simulator

The Wireless Network Simulator (WNS) is a user friendly, open architecture software tool (see Fig. 2) developed for simulating and evaluating the quality of wireless data transmissions, using a variety of wireless protocols. The tool was developed using Visual C++ and consists of a core application realizing the desired Medium Access Control (MAC) layer functionality, and a set of plug-ins (external libraries) implementing additional networking features (such as channel modeling, vendor-specific protocol algorithms, etc).

In this work, the well-established IEEE 802.11b [3] standard was employed as the basic transmission protocol with the MAC layer Quality of Service (QoS) enhancements defined in the current version of the IEEE802.11e amendment [4], which is currently under standardization process. Due to the strict time-critical nature of the digital audio data transmissions, the centralized polling channel access scheme of the 802.11e was selected as the transmission coordination mechanism, as it provides better service guarantees and achieves greater overall QoS performance [5].

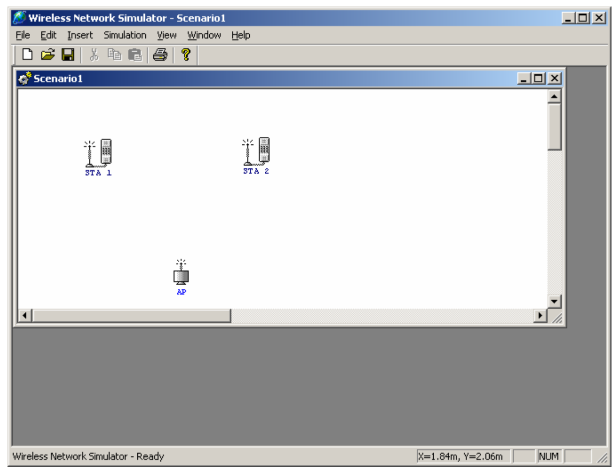

Fig. 2: The WNS application

The WNS application incorporates a set of traffic generators for modeling standard applications (such as G.711 voice), but it also allows input traffic from trace files. This feature was found to be very useful for representing each digital audio channel as trace file derived by the Wave-to-Trace application and described in the following Section. Additionally, numerous well-established performance statistics are calculated, such as the mean and instant measured throughput and the total packet delay, while the packet transmissions can be analytically traced over time. This advanced feature was used as input to the post-processing stage and the Trace-to-Wave converter application described in Section 1.4.

1.3. Wave-to-Trace Converter

As it was mentioned in the previous paragraph, using the WNS, digital audio traffic is modeled using the trace file format, which provides an accurate description of the traffic flow packet generation as a function of time. In particular, a trace file contains the generated source frame sizes and the corresponding packet generation time. This information can be used for mapping the transmitted data packets to specific segments of stereo wave files containing the original digital audio samples. During this work, this process was performed in the pre-processing stage using the developed Wave-to-Trace application (see Fig. 3), providing a trace file for each discrete audio channel.

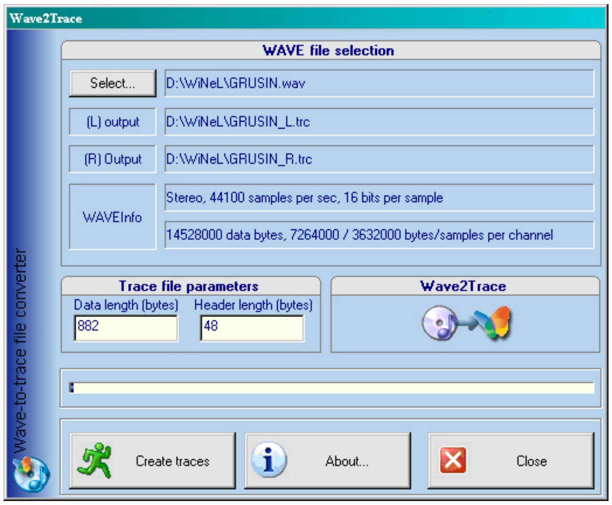

Fig. 3: The Wave-to-Trace application

The trace file conversion parameters consist of the desired pure audio data transmission packet length L_p, as well as the corresponding packet header length. For this work, the User Datagram Protocol (UDP) was employed as the transport layer protocol; hence the packet header was equal to 48bytes in all test cases, while typically, different packet length values (L_p) were employed.

1.4. Trace-to-Wave Converter

The WNS produces a trace file containing the timing information for each successfully transmitted packet per audio stream as well as information for lost / dropped packets. During the simulation post-processing stage the Trace-to-Wave converter creates a wave file representing the wireless playback equivalent of the original digital audio data, by mapping the output trace files to the original audio samples.

The application implements a finite-size reception queue (buffer), which is fed with the successfully received data packets. On the other hand, the buffered audio data are read in a sample-by-sample basis, at a rate equal to the original audio sampling rate. The reception queue length, the initial latency (or pre-buffering time interval) for launching the playback for each receiver as well as the protocol header length, are variable parameters defined by the user. Both the reception queue length and the header length are defined in bytes, while the initial playback latency is measured in beacon periods, equal to 100ms each.

The percentage of the free memory space at the reception queue over time can be also derived from this application, showing the instances where the queue is empty or data overflows occur, causing in both cases playback audible distortions.

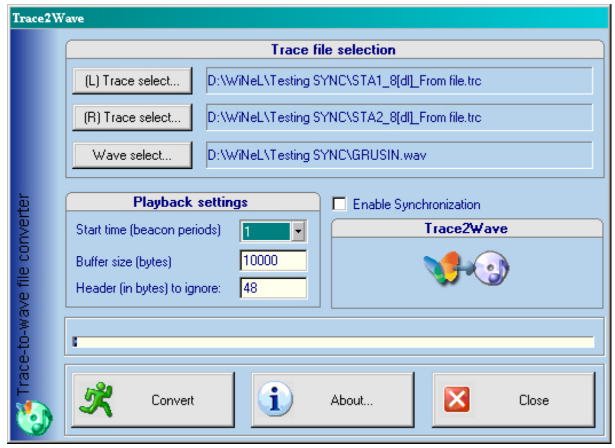

Fig. 4: The Trace-to-Wave application

2. RESULTS

In this Section, a wireless, stereo, digital audio transmission case study is presented and the results obtained using the proposed evaluation platform are discussed. In this case, the physical transmission rate was *PHY*=11Mbps, and the transmitted packet size was equal to L_p=882 bytes. The simple scheduler reference design defined in [4] was employed as the channel access coordinator, while the pre-buffering time interval at both receiver sides was set to 100ms.

The "reproduced" version of the wave file may be distorted, either by the presence of silence gaps caused by delayed reception or by waveform discontinuities, caused by packet losses. These types of distortions are detected though the extraction of the relative delay information for each stream with reference to the initial test signal.

Fig. 5 shows the Transmission Queue (TxQ) data length derived from the Network Simulator and the Reception Queue (RxQ) obtained from the Trace-to-Wave application over time per stream, as well as the differential delay for the two streams, calculated as a function of time.

Provided that the maximum allowed TxQ and RxQ data lengths for each audio traffic stream has been set to 4608 bytes and 5000 samples respectively, the maximum data that can be inserted into the corresponding queue equal to the nearest, lower integer multiple of the packet size employed, respectively 4410 bytes and 4998 samples.

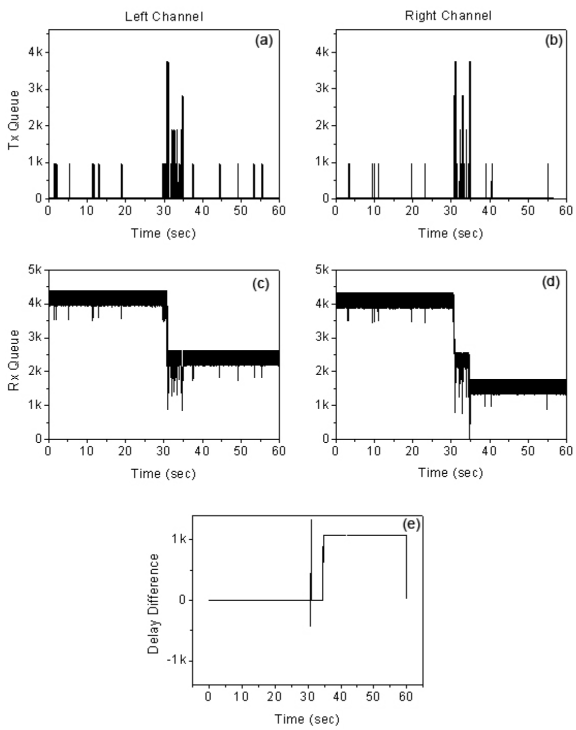

Fig. 5: (a), (b) TxQ State; (c), (d) RxQ State; (e) Differential Delay (for *PHY*=11Mbps, L_p=882 bytes and Simple Scheduler)

From Fig. 5 it is clear that significant overflows occur in the TxQ for both audio channels, between the 30sec and 35sec time interval, leading to audio data losses. While not easily distinguishable, on top of such overflow, the RxQ also becomes empty at a number of time instances, causing silence gaps in the reproduction, in spite of the 100ms pre-buffering time employed. As can be seen in Fig. 5 (e), while the delay difference fluctuates between 30sec and 35sec, it becomes constant after that period. Since no remote synchronization (or phase-shift compensation) algorithm is available within the protocol, the audio channel out-of-phase reproduction is perpetuated throughout the overall simulation time.

The effect of the data overflows in the TxQ and the wireless variable packet delay transmission is clearly shown in Fig. 6, where the original transmitted and the wirelessly reproduced waveforms are shown for a single audio channel. Apart of the silence gaps, a significant shift of the original waveform to the right side of the plot diagram is observed, which introduces relative channel phase delay. The audibility of both types of distortions introduced (silence gaps and relative channel delay) was verified through a sequence of listening tests in a typical stereo reproduction environment. A typical set of the corresponding audio material can be obtained from http://www.wcl.ee.upatras.gr/AudioGroup/.

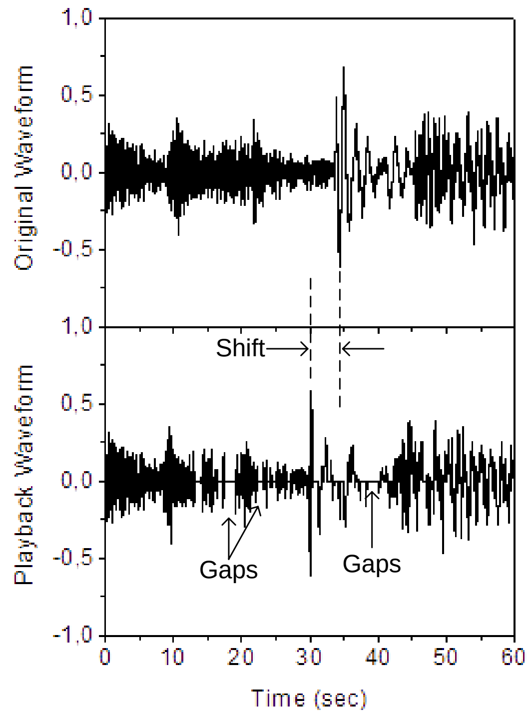

Fig. 6: (a) Original digital audio source waveform (b) Reproduced (playback) waveform

3. CONCLUSIONS

In this work, a novel wireless evaluation tool was developed in order to study and assess real-time wireless digital playback. The tool has been successfully employed in conjunction with an open-architecture wireless networking simulation platform (WNS) for the analysis of a stereo wireless digital audio playback environment, based on the IEEE 802.11b protocol, and the centralized, polling wireless access defined in the current IEEE 802.11e amendment.

While the wireless transmission protocol employed provides QoS service guarantees, it is evident that significant reproduction distortion may be introduced, due to the non-ideal nature of the wireless channel, which may cause lack of data in the reception buffers or packet overflows in the transmission queue. Moreover, playback distortions induced only in one audio stream, can lead to out-of-phase reproduction between channels, as shown in the example test case. An advanced remote synchronization mechanism can possibly overcome the audio channel phase shifting during the reproduction, by ideally minimizing the relative channel delay value.

4. REFERENCES

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