

Recent Developments in Aircraft Flyover Noise Simulation at NASA Langley Research Center

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ABSTRACT

The NASA Langley Research Center is involved in the development of a new generation of synthesis and simulation tools for creation of virtual environments used in the study of aircraft community noise. The original emphasis was on simulation of flyover noise associated with subsonic fixed wing aircraft. Recently, the focus has shifted to rotary wing aircraft. Many aspects of the simulation are applicable to both vehicle classes. Other aspects, particularly those associated with synthesis, are more vehicle specific. This paper discusses the capabilities of the current suite of tools, their application to fixed and rotary wing aircraft, and some directions for the future.

1.0 INTRODUCTION

The NASA Langley Research Center (LaRC) has a long record in psychoacoustic studies of aircraft flyover noise. Such studies require human subjects to listen to flyover sounds and to rate them according to various scoring methodologies. Two methods of presenting the flyovers are available: field tests and laboratory tests. Field tests using real aircraft flying over the subjects are problematic because exactly what each listener is hearing is unknown, the same flyover can never be reproduced exactly, and flight tests are extremely expensive. On the contrary, simulated environments in the laboratory enable the exact replication of stimuli, are able to present the same sound at different levels, and perhaps most important in the current work, allow an assessment of future aircraft and operations.

The use of recorded flyover noise in a real-time virtual three-dimensional audio-visual environment is quite realistic because the stimuli consist of the actual noise received by an observer. Source noise directivity, propagation effects including atmospheric attenuation, Doppler shift, and spreading loss, and ground plane reflection (if recorded at head height) are all embedded in a monaural recorded signal. Such a virtual

environment was created and first presented in 2003 [1]. The use of head-tracking allows the subject to interact with the virtual environment by dynamically switching head related transfer functions on a dedicated audio server, creating a binaural sound field based on the instantaneous relationship between the virtual moving source and the stationary observer. As compelling as such a simulation may be, it is not without its drawbacks, including the difficulty in making flyover recordings devoid of background noise, unknown propagation path due to wind, temperature, and atmospheric turbulence, and fixed trajectories and observer locations. Perhaps the most restrictive liability is that recordings preclude the ability to assess new aircraft designs and/or operations that are in the conceptual or laboratory development stage.

To overcome the limitations of recordings, synthesized flyover noise may be used. A previously developed flyover noise synthesizer [2] was found to be unsuitable for current and future requirements. A new synthesizer developed and in use at NASA LaRC employs a three-stage process consisting of pre-processing, synthesis, and rendering. In the pre-processing stage, the source noise is characterized using experimental data and/or predictions. This characterization yields either source noise directivity as a function of frequency and emission angle or directionally dependent pressure time histories, depending on the characterization method employed. In either case, the pre-processing stage is not meant to be performed in real-time, but is conducted *a priori* to construct a database over a range of operating conditions for use in the synthesis stage. The Aircraft Noise Prediction Program (ANOPP) [3] has served as the primary tool for predicting component noise directivity for fixed wing subsonic aircraft. Other noise prediction tools may be employed for rotary wing aircraft, e.g. WOPWOP [4] or PSU-WOPWOP [5].

The second stage involves synthesis of the aircraft noise at the flying source. The time-domain pressure signal is constructed at a single point corresponding to the instantaneous emission angle, which is determined by the propagation path between the source and observer. At present, a constant speed of sound is assumed, necessitating a straight-line path. Extensions to a curved propagation path are discussed in the paper. Two forms of synthesis, broadband and pure tone, are currently employed. A third form, narrowband, is related to broadband synthesis and is partially implemented. A fourth form using wavetable synthesis will be required for time-domain predictions. Each form is discussed in the paper. Whichever form of synthesis is employed, the emission angle changes as the aircraft moves and the synthesis must be capable of smoothly transitioning between angles. A smooth transition is also required to accommodate changes in operational state. Since predictions generally do not account for temporal variations observed under even steady-state operating conditions, these effects may be reintroduced in the synthesized signal by modulating the predictions according to available empirical data [6, 7]. The synthesis stage is embodied in a custom computer application referred to as the Aircraft Source Noise Generator (ASoNG).

In the third stage, the synthesized sound at the flying source is rendered to the listener on the ground in a real-time fashion [7-9]. Included in the rendering stage are spreading loss, absolute time delay, atmospheric absorption, ground plane reflection, and listener effects required for playback over headphones or speakers. The choice of straight-line versus curved propagation path affects each, in a manner to be discussed further in the paper.

Finally, a software and hardware system for creation of a real-time virtual environment, including the aforementioned third stage rendering, is discussed. The system has a subjective testing element which presents questions and records subjective responses. It utilizes a combination of custom software applications and commercial off-the-shelf software and hardware, and is referred to as the Community Noise Test Environment (CNoTE). The CNoTE visualization system is scalable for use on diverse graphics hardware including head-mounted displays, desktop and large screen presentation, and CAVE™ [10] environments. The CNoTE auralization system utilizes the AuSIM3D [11] audio localizing server software and hardware,

and is capable of binaural rendering over headphones, or vector-panning over speakers. A description of the CNoTE system and how it may be used under different aircraft flyover simulation scenarios is discussed in the paper.

2.0 SYNTHESIS

ASoNG uses a hierarchical structure to define the sounds to be generated. Aircraft are comprised of multiple noise sources, e.g. port side turbofan engine or helicopter main rotor. Each source is assumed to be compact and is treated as a point source. All sources associated with an aircraft share a common aircraft trajectory, but each is offset and rotated relative to the aircraft coordinate system. Each source may be further divided into its multiple components. For example, the port side turbofan engine may be comprised of forward and aft radiating fan noise components, a jet noise component, a core noise component, etc. All components of a source are considered to be collocated and therefore share a common source trajectory. Synthesis is performed at the component level, with each component having its own set of specified operating conditions and noise emission characteristics. The emission characteristics are a function of operating condition and are determined via analysis in the pre-processing stage. Emission characteristics may be expressed as frequency and emission angle dependent directivity patterns or as time and emission angle dependent waveforms. Since the propagation of sound to an observer on the ground is of interest, emission characteristics are provided in the form of hemispheres below the aircraft. ASoNG currently recognizes components defined as one of three generic types: broadband, narrowband, and pure tone. As previously indicated, broadband and pure tone syntheses are fully implemented, while narrowband is partially implemented. A fourth form, wavetable synthesis, necessary for handling time-domain predictions, has yet to be implemented. Wavetable synthesis will likely be required for some rotary wing aircraft components. The component type specified dictates how the sound for that component is to be generated. The sound generated from all components in each source is mixed to a single output stream for the rendering stage. By rendering each source simultaneously but separately, the phase difference between sources is realized due to slightly different trajectories used for each source. Interactions between sources and the airframe, e.g. acoustic shielding provided by the fuselage or wing, are not modelled. In the future, shielding effects may be incorporated by modification of the source emission characteristics, but at present this is an area of active research [12].

Instead of synthesizing the sound at all possible directivity angles, synthesis is performed at a short distance from the moving source at the emission angle at the time of emission, for a known observer position. Propagation or receiver effects are included during the rendering stage. For straight-line propagation, the emission angle is determined from the straight-line path between the source and observer at the time of emission, as shown in Figure 1. For curved propagation paths, either a ray-tracing [13, 14] or parabolic equation code will be needed to determine not only the emission angle, but also the receiver angle and the path itself for use in the rendering stage.

The output of the synthesis process is analogous to the signal that a single microphone would record at the emission angle. As the source moves, the emission angle changes in a continuous fashion and, by analogy, the microphone location smoothly traverses the hemisphere below the aircraft. In this manner, the source directivity is embedded in the synthesized signal and does not need to be explicitly represented at the time of rendering. This approach to synthesis makes simulation of arbitrary trajectories straightforward.

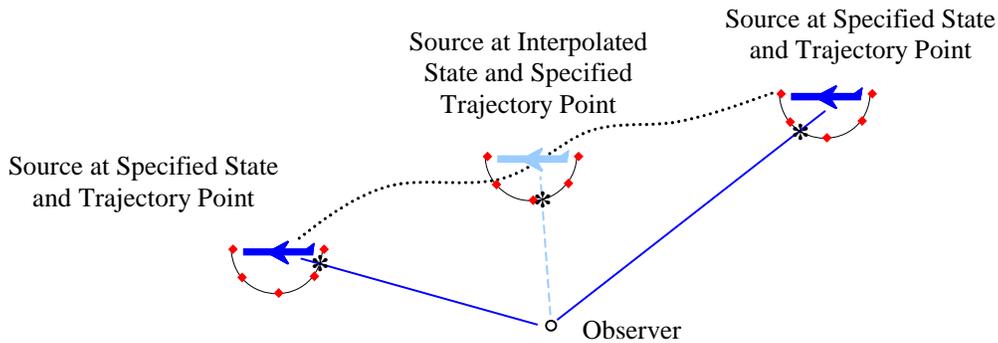


Figure 1: Synthesis is performed at the directivity angle * at the time of emission.

2.1 Synthesizer Stages

The ASoNG program is intended to run in one of two double buffering schemes, one of which is for synthesis performed in real time and the other for non-real-time synthesis. Both schemes use a preparation chain performed at time buffer $i-1$, and a combined synthesis, mixing and output chain at time buffer i . Each buffer is of size 2^n samples and is typically 512 samples. Processing is performed at an audio sampling rate of 44.1 kHz, giving a typical buffer hop size of 11.6 ms ($512/44100$ s). The two schemes differ in the manner in which the clock is advanced. In the non-real-time scheme shown in Figure 2, the clock is advanced when both chains are completed. Therefore, the speed of operation is dictated by the slower of the two chains. Consequently, a buffer under-run will never occur. In contrast, the clock in the real-time scheme will be driven by a hardware interrupt, as shown in Figure 3. Hence, buffer under-runs are possible if the clock is advanced prior to completion of either of the two chains. The utility of a real-time synthesis is to allow real-time changes to the aircraft trajectory and/or operating conditions, such as would be necessary if ASoNG were coupled to a flight simulator. As real-time operation has not been the focus of the development, this capability is only partially implemented with the hardware interrupt simulated in software.

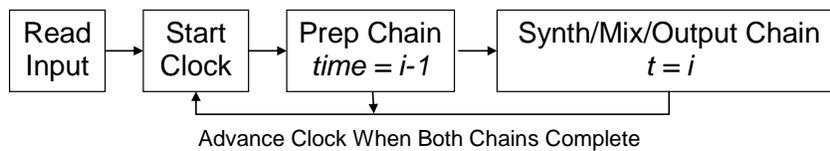


Figure 2: Non-real-time double buffering scheme.

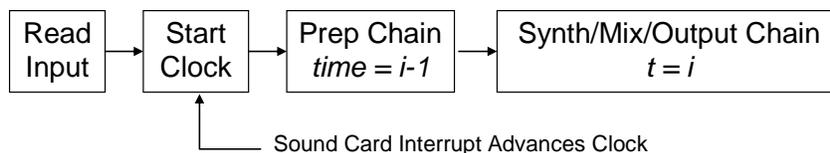


Figure 3: Real-time double buffering scheme.

The ASoNG program was written as a multi-threaded application. This allows the program to automatically take advantage of the computational efficiencies offered by hyper-threading and multi-core processors without having to restructure the program. Each processing chain consists of multiple threads.

The input file contains information concerning the aircraft, source and components, digital signal processing (DSP) parameters, low frequency oscillator (LFO) parameters, trajectory data, and listener position information. Input data is read prior to the start of the preparation chain, as shown in Figure 2 and Figure 3.

2.1.1 Preparation Chain

The preparation chain performs up-front calculations needed in the subsequent Synthesis/Mix/Output chain. A depiction of the preparation chain is shown in Figure 4.

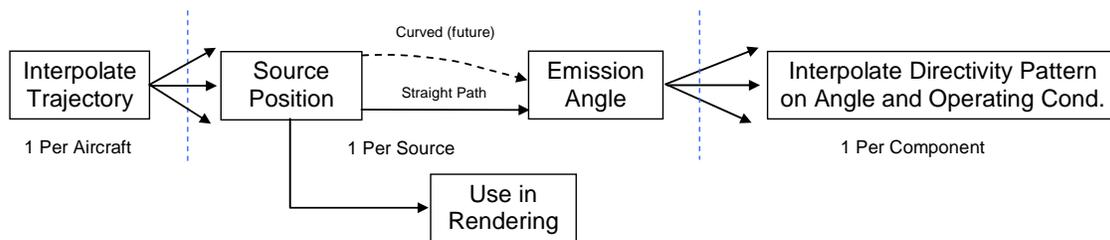


Figure 4: Flow of operations in the preparation chain performed during each time buffer.

Prior to the start of the clock, ASoNG loads the prediction database. For non-real-time synthesis, the prediction databases for each component and aircraft trajectory are pre-loaded as soon as the input file has been read. The operational state of each component, e.g. RPM of port side engine forward fan, is included with the trajectory data. For future real-time syntheses, the prediction database would be pre-loaded, but the trajectory data would come from another source, e.g. a flight simulator. In either case, the buffer-to-buffer interpolation of trajectory data is not performed until the synthesis is running. This will allow the injection of real-time trajectory information without restructuring the code. During each processing buffer, the following procedures are performed:

- The aircraft trajectory including operational state is interpolated based on current simulation time. Because sources from multiple aircraft may be synthesized at a time, each aircraft trajectory must be interpolated.
- The absolute position of each source is next calculated using its position relative to the aircraft datum. This information is also used during rendering. At present, the straight-line path between each source and the listener is then used to determine the azimuthal and fore-aft emission angle. In the future, an auxiliary code, such as the NASA LaRC Spectral Attenuation Model (SAM), which uses the aforementioned ray-tracing code [13, 14], could be used to determine the curved path. However, doing so will likely preclude the possibility for real-time synthesis.
- For each component with emission characteristics defined in the frequency domain, i.e. those of type broadband, narrowband or pure tone, the predictions database is interpolated over emission angle and operational state to yield an instantaneous directivity pattern for use in the Synthesis/Mix/Output chain. Broadband components are specified in terms of sound pressure levels in decibels (dB) for each 1/3 octave band. Narrowband components may be expressed as sound pressure auto spectra or

power spectral densities. Tonal components are specified in terms of sound pressure levels at each tonal frequency. An alternative to the above interpolation scheme would be required for components with emission characteristics defined by time-domain predictions. This might entail cross-fading waveform predictions across emission angle and operational state.

In the above, component directivity predictions are performed in the pre-processing stage and are made over a range of discrete independent variables which define the operational state, e.g. throttle setting and Mach number. These predictions are performed without regard to the aircraft position. The last operation in the preparation chain linearly interpolates the component directivity between all relevant operational states to obtain the instantaneous directivity at emission time. The onus is on the user to ensure that the prediction database is sufficiently populated to characterize the directivity between operational states in this manner.

2.1.2 Synthesis/Mix/Output Chain

The Synthesis/Mix/Output chain actually consists of two phases. The synthesis phase generates a buffer of pressure time history data for each component, while the mix/output phase mixes all components associated with each source and writes the output. The synthesis method is based on the type selected, that is, broadband, narrowband, pure tone, and wavetable. Both the synthesis and mix phases are targets for various forms of modulation, as explained below. A depiction of the synthesis/mix/output chain is shown in Figure 5. Each component of a source has its individual thread in which the core of the synthesis operations is performed.

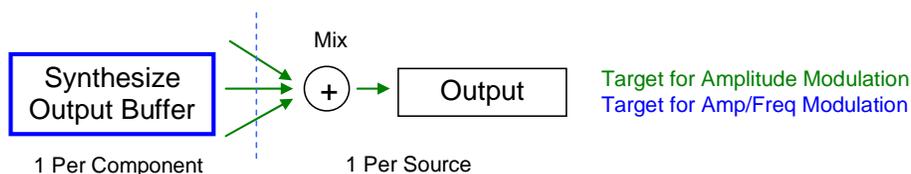


Figure 5: Flow of operations in the synthesis/mix/output chain performed during each time buffer.

2.1.2.1 Broadband and Narrowband Synthesis

The broadband and narrowband synthesis is performed using a subtractive technique akin to a phase vocoder [15]. A flowchart for each type is shown in Figure 6 and Figure 7, respectively. Each synthesis thread performs the following steps:

- For either noise type, the thread first pops the interpolated directivity data computed during the previous buffer of the preparation phase.
- For broadband types, the total pressure in each 1/3 octave band is computed. This is simply a dB to pressure conversion.
- Previous work by the authors [7-9] indicated that the time-invariant spectra produced by essentially time-averaged predictions lack temporal variations found in real near-field source noise recordings. An analysis of such data [6] indicated the presence of sub-audible low frequency oscillations (< 20 Hz) about the mean level. In this processing step, a LFO derived from a short-time Fast Fourier Transform (FFT) analysis of empirical data, may optionally be used to modulate the amplitude of either the 1/3 octave band levels or the narrowband levels.

- For broadband types, the modulated spectrum is converted into a narrowband spectrum by dividing it into 2^n evenly spaced narrow bins. In order to capture the low frequency response, the spacing of the bins is small, so the number needed to fully span the 1/3 octave spectrum is typically large, e.g. 8192 points. This process yields the FFT amplitudes of a finite impulse response (FIR) filter. For narrowband types, the modulated spectrum itself constitutes the FFT amplitudes of a FIR filter.
- Because the filters are large, convolution with a white noise source is performed in the frequency domain. An inverse FFT is performed to reconstruct the time domain signal. Each subsequent buffer is combined with previous buffers using an overlap-add technique [15]. In this manner, the synthesized sound smoothly transitions with changes in the source directivity. Samples of a synthesized broadband jet component with and without application of the LFO can be found on the web site <http://stabserv.larc.nasa.gov/flyover/>.

2.1.2.2 Pure Tone Synthesis

The pure tone synthesis is performed using an additive technique as shown in Figure 8. Unlike the broadband and narrowband syntheses, the pure tone synthesis is performed in the time domain. Each synthesis thread performs the following steps:

- The thread first pops the interpolated directivity data in the form of frequencies and amplitudes for each harmonic. These were computed during the previous buffer of the preparation phase.
- Like the broadband and narrowband techniques, the pure tone synthesis may optionally apply a LFO to independently modulate each tonal amplitude and frequency. In the absence of an analytical model, analysis of empirical data is needed to determine the attributes of the LFO.
- A sweep function smoothly shifts each tone from the buffer start frequency to the buffer end frequency, while a fade function controls the amplitude variation.
- The sum of buffers from the individual harmonics forms a single component output buffer.

2.1.2.3 Wavetable Synthesis

For source emission predictions made in the time domain, in particular those associated with some components of rotary wing aircraft, an alternative synthesis technique will be required. One possibility is to generate short time histories (wavetable) at a number of discrete emission angles during the pre-processing stage. During the preparation phase, the looped samples would be cross-faded to the instantaneous emission angle. In the synthesis stage, each sample could be amplitude modulated if necessary, and combined with previous buffers in an overlap-add technique. Wavetable synthesis is an area of future work.

2.1.2.4 Mix and Output

The last phase of the synthesis/mix/output chain is for mixing and output. There is a separate thread for mixing the component output buffers to generate a single output buffer for each source. Since the components of each aircraft source are collocated, the rendering stage operates on source data, not on component data. Regardless of the synthesis technique employed, the steps of the mix and output phase are as follows:

- The current output buffer from each component is popped from its respective synthesis thread.
- Gains are applied, and output buffers from each component within the source are summed to form a single output buffer. Gains may be in the form of a constant gain or a modulating gain, and can be applied per component, per source, or per aircraft.

- The source output buffer is written to a 32-bit IEEE floating point WAV file in the non-real-time scheme, and loaded on the audio server for rendering. In any future real-time implementations, the source output buffer would be written directly to digital audio hardware on the synthesis computer, then transferred to the audio server as part of an additional buffered operation. Because the output is generated per-source, an aircraft with two engine sources would have two separate output files, both of which need to be simultaneously played back during the rendering stage.

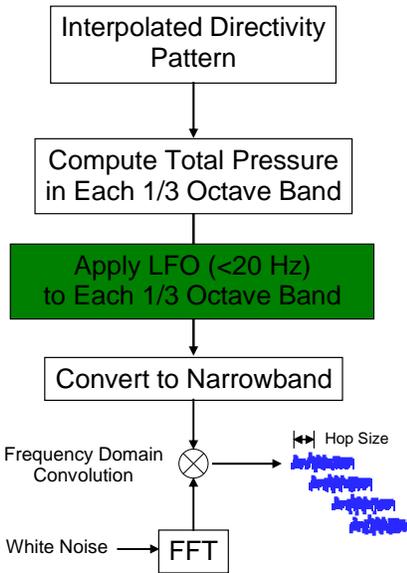


Figure 6: Flowchart of broadband synthesis.

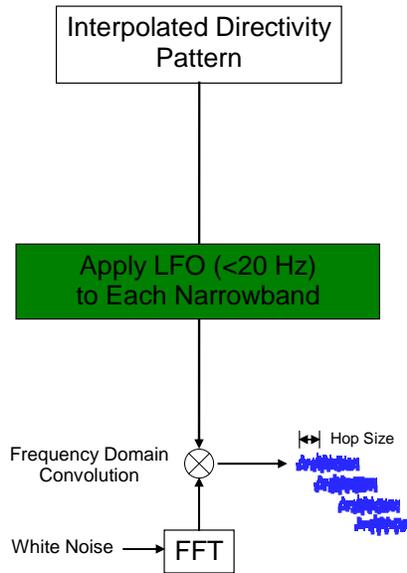


Figure 7: Flowchart of narrowband synthesis.

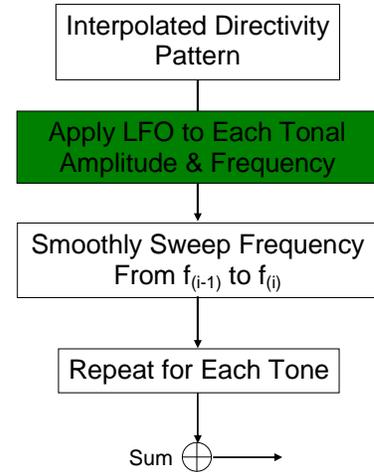


Figure 8: Flowchart of pure tone synthesis.

2.1.2.5 Performance

There are two applicable measures of performance – latency and CPU time. Latency is applicable to the real-time buffering scheme. It is determined as the sum of each buffered operation. In the real-time scheme, there are minimally three buffers. Two buffers are associated with the synthesis stage: one for the preparation chain and one for the synthesis/mix/output chain. A third buffer is associated with output through the digital audio hardware. Depending on the hardware, an additional buffer may exist on the input side between the input source (simulator) and the synthesizer. Since both synthesis chains and the digital audio hardware operate at the same audio sampling rate of 44.1 kHz, the minimum system latency for a typical buffer size of 512 points is equal to 34.83 ms ($3 \times 512 / 44,100$).

CPU time is the applicable measure of performance for the non-real-time buffering scheme. As an example, a timing check was made of a synthesis performed on an older 1.7 GHz single core Pentium 4 computer with 1 GB RAM. The sound was synthesized for a straight and level flyover of an aircraft with a single source and a single broadband component, and included a broadband LFO and three independent variables for interpolation (Mach, thrust, and fore-aft emission angle). The buffer hop size was 512 samples, the sampling rate was 44.1 kHz, and the FFT length was 8192. A synthesis time of 22 seconds was required to generate a 41 second WAV file. While these figures are in no way authoritative, they nevertheless indicate that real-time performance can be achieved even with modest computing resources.

3.0 RENDERING

The rendering stage is performed in real-time on the AuSIM3D audio server [11] under the control of a client application running the overall simulation. The details of the simulation environment are discussed in Section 4.0. The audio server performs DSP on the input signal, either as synthesized source noise or ground based recording, through application of a time-dependent gain, time delay and filter, as depicted in Figure 9.

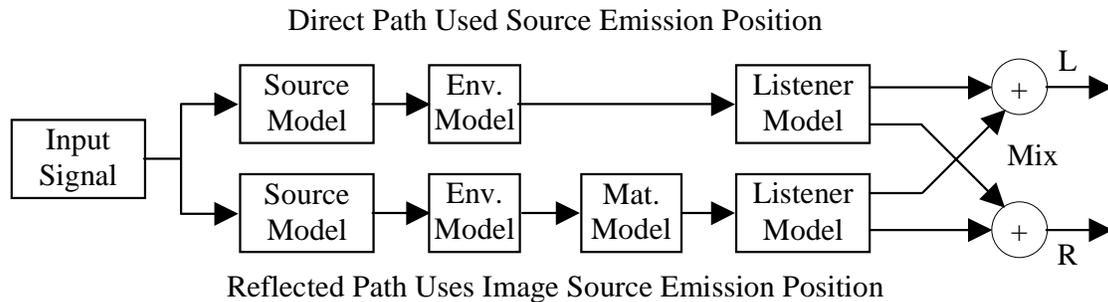


Figure 9: DSP performed during the audio rendering stage of synthesized sources for auditory display over headphones.

3.1 Dynamic Models

The particular gains, time delays and filters applied to the signal are dynamically computed using various physical and empirical models, including source, environment, material and listener models, as indicated in Figure 9. The client application controls the attributes of the models, e.g. time delay on or off. The attributes used are dependent on the particular simulation and source material, i.e. synthesized or recorded. The gain, time delay and filters returned from the various models are concatenated and applied in 2^n size buffers by the DSP engine to obtain the output stream(s).

3.1.1 Source Model

The AuSIM3D source model returns the direction-dependent gain for each source. For recorded flyovers, the source directivity pattern is implicit in the recording. The recording may thus be treated within the source model as a monopole having uniform directivity. For synthesized flyovers, the source directivity pattern must be explicitly taken into account. However, the paradigm adopted here takes the source directivity into account within the synthesis stage. Therefore, like the recorded flyover, the input audio stream implicitly contains the effect of source directivity and the source may be treated as a monopole. Hence, the gain in the source model is only used to adjust level.

3.1.2 Path Dependent Models

The path utilized in all path-dependent models is based on the emission position, that is, the position of the source at the time the sound is emitted. For synthesized flyovers, synthesis occurs at the emission position. For recorded flyovers, the emission position must be calculated with knowledge of source trajectory and the propagation path, i.e. straight or curved. Because the AuSIM3D DSP engine assumes a constant speed of sound, current capabilities are limited to straight-line propagation. The use of an off-line processor for curved path simulation is discussed in Section 3.2.

3.1.2.1 *Environment Model*

The environment model returns the time delay, gain and filter corresponding to the absolute delay, spreading loss and atmospheric absorption, respectively. For straight-line propagation, the absolute delay between when the sound is emitted and when it is heard by the observer is the emission distance divided by the constant speed of sound. The emission distance is the distance between the source and the observer at the emission time and is a function of time. The time rate of change of the absolute delay simulates Doppler shift. Simulation of spherical spreading loss is performed through application of a time varying gain, which is a function of the emission distance. The atmosphere absorption filter used is a NASA-developed plug-in model to the AuSIM3D engine. The model pre-computes the attenuation as a function of distance and elevation angle for uniform and non-uniform horizontally stratified atmospheres. In the case of a non-uniform atmosphere, the path is still considered as straight line even though the temperature variation would indicate otherwise. A more consistent treatment will require implementation of the curved path simulation. The attenuation is expressed in the form of FIR filters, typically on the order of 128 taps. Additional details regarding the environment model and NASA atmospheric filter plug-in may be found in reference [9].

For recorded flyovers, the gain, time delay, and filter are all disabled because the effects of spreading loss, absolute delay, and atmospheric absorption are part of the recording. All effects are required for simulation of synthesized sounds, as this stage propagates the sound from the flying source to the listener.

3.1.2.2 *Material Model*

The presence of a ground surface produces a reflection, which interferes with the direct sound. The interference produces a comb filter effect, which alters the spectral content in a time-varying manner as the aircraft moves along its trajectory. Instead of implementing ground reflection as a comb filter [16] within the environment model, an image source is used to spawn a secondary path with its own emission position, as indicated in Figure 9. The image source is positioned at a location symmetric about the ground plane. For an observer above the ground plane, the reflected path is longer than the direct path. Hence, the delay obtained from the environment model for the reflected path is greater than that of the direct path. The comb filter effect manifests itself at the mixing stage. The material model specifies attributes of the ground plane. In its current implementation, a simple gain is applied. In the future, a custom plug-in could be written to establish FIR filters which model the impedance of different ground surfaces, e.g. grass, asphalt, etc. The comb filter effect and its importance to flyover simulation can best be appreciated by listening to the audio clips on the web site <http://stabserv.larc.nasa.gov/flyover/>.

If a recording is made at head height, the effect of ground reflection is automatically realized, so a virtual source is not generated and the material gain is not applicable. If a recording is made at a hard ground plane, it is possible to back propagate the source to the emission time and position, then treat the sound as it if were synthesized. That is, use both the environment and material model with an image source and correct the level for pressure doubling. Such a procedure would permit a single recording to be used for simulating different types of ground impedance.

3.1.3 **Listener Model**

The listener model returns time delay, gain and filter required to position the source in the virtual three-dimensional space about the observer. Rendering over headphones utilizes binaural simulation [17] which applies a head related impulse response function (HRIR) independently for the left and right ears. The HRIR function accounts for interaural time delay (ITD), interaural intensity difference (IID), and scattering about the pinnae, head and torso. The latter is represented with minimum phase FIR filters, typically on the order of

128 or 256 taps in length. Rendering over speakers utilizes a derivative of vector base amplitude panning [18]. In this case, the dimension of the listener model is not two but n , where n is the number of speakers. For both headphone and speaker rendering, the particular set of filters used is dependent on the emission position of the source relative to the observer's position and orientation, and on the path. These dictate the elevation and azimuth angles at the receiver location as a function of time. Through the use of head tracking, real-time changes to the observer's orientation are conveyed via the client application to the audio server, allowing the observer to interact with the simulation.

3.2 Simulated Recordings and Off-Line Rendering for Curved Paths

If real-time control of path-dependent model attributes is not required, the time delay, gain and filter of the listener model may be disabled, and the synthesized sound can be pre-rendered and recorded at a head height position. The resulting simulated monaural recording could subsequently be rendered in real-time through the source and listener models in the same manner as an actual recording.

By extension of the above, a curved propagation path could be rendered through an off-line version of the AuSIM3D engine. For each processing buffer, the time delay, gain and filter associated with the curved path would be specified, instead of that generated by the constant speed of sound environment model. The curved path data could be generated by the SAM or other comparable propagation code. Additionally, the emission and receiver angles would differ from the straight-line path and these would need to be used in the synthesis and rendering stages, respectively. In the same manner as the above, a simulated monaural recording to a head height position could be made, and rendered later in real-time. What is lost in this scenario is real-time control of the path-dependent models. This does not necessitate that conditions along the path be invariant with respect to time. For example, wind gusts would be permissible, but their effects on the path would have to be calculated and specified *a priori*, not via real-time input during the course of the simulation.

4.0 ARCHITECTURE

The CNoTE system was developed as a framework for performing subjective testing in virtual environments. It consists of three main modules: the Controller, the Quizzer and the Simulator, as shown in Figure 10. The function of the Controller is to specify the test sequence via a user-defined script and to coordinate the activities of the Simulator and Quizzer modules. The function of the Quizzer is to present questions to subjects following a stimulus, and retrieve the response for logging by the Controller. The questions the Quizzer displays are conveyed to it via the Controller as part of the test sequence. The Quizzer uses a small PC device for display and data entry. The heart of the CNoTE system is the Simulator module, which renders three-dimensional graphics and audio, and processes real-time tracking data. The action of the Simulator is determined via commands issued to it by the Controller module. In particular, the Controller sets the Simulator master clock and enables/disables pre-loaded flyover events on the audio server.

The elements of the Simulator module are shown in Figure 11. A master simulation clock running on the graphics server (audio client) synchronizes the visualization with the audio rendering such that a listener in the virtual environment will see the aircraft at the visual position and hear it coming from the emission position. At each update cycle, the graphics server interpolates the visual trajectories according to the current simulation time, updates the visual scene graph with the interpolated position, and updates the listener orientation with the current tracking information. The listener orientation is used for both graphics and audio rendering.

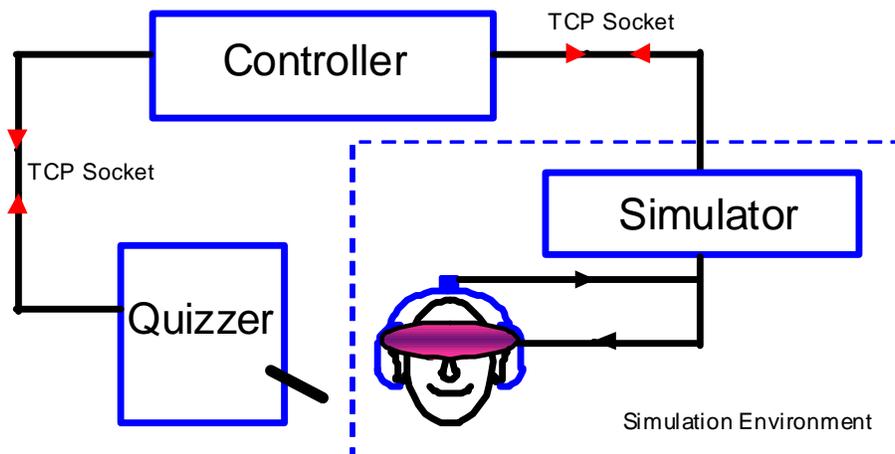


Figure 10: Schematic of the CNoTE system architecture.

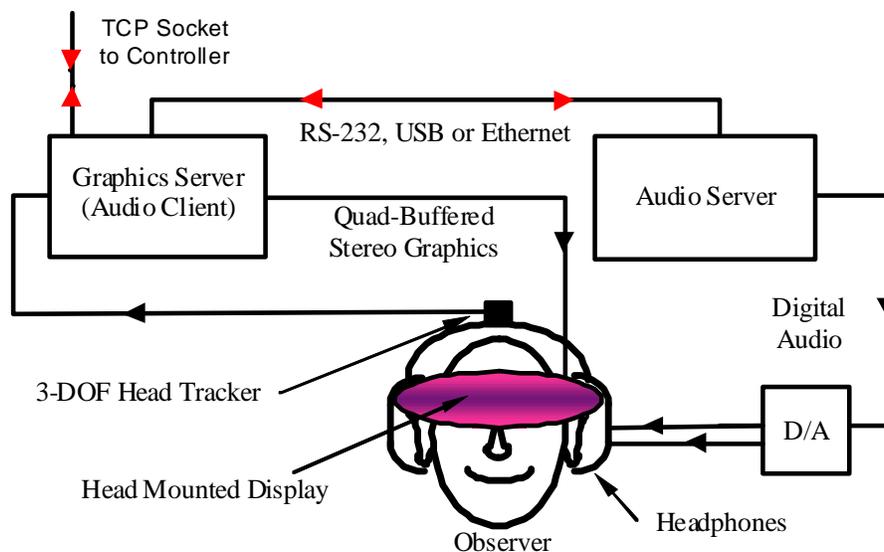


Figure 11: Schematic of CNoTE Simulator module.

When the aircraft trajectories are known in advance, the emission positions are predetermined and loaded on the audio server as a sequence of time-stamped position locating events. The sequence is triggered by an internal clock running on the audio server, which is synchronized with the graphics server master clock, allowing the graphics and audio to run independently but synchronously. If, in the future, the Simulator module was coupled to a flight simulator, the real-time trajectory data would need to be provided to both the graphics and audio servers. Of course, in this case, real-time synthesis would also be required to generate the appropriate source content. An example rendering from the CNoTE Simulator is available in the form of an AVI movie on the web site <http://stabserv.larc.nasa.gov/flyover/>.

Finally, as mentioned in the Introduction, both the graphics and audio simulations are scalable across different display devices. The target device is selected via configuration files without the need to alter source code. The audio displays used to date at NASA LaRC have been limited to headphones. The frequency content of subsonic fixed wing aircraft sources is such that headphones are adequate. However, as the research focus shifts to rotary wing aircraft, headphones will no longer be adequate because of the low frequency content associated with such aircraft. To prepare for this, the Exterior Effects Room (EER) at the NASA LaRC is being upgraded to provide audio rendering over speakers. The EER is an auditorium previously used for flyover noise research [19-22]. The upgrade entails installation of 27 satellite speakers and 4 subwoofers, as shown in Figure 12. The AuSIM3D engine will utilize vector panning to deliver 31 independent channels of positional audio. Expected performance is greater than 90 dB (flat weighting) down to the 16 Hz 1/3 octave band. In addition to the audio upgrade, a 3D graphics system capable of generating active or passive stereo-graphics will be installed with projection on a large front screen. The facility modification is scheduled for completion in the spring of 2009.

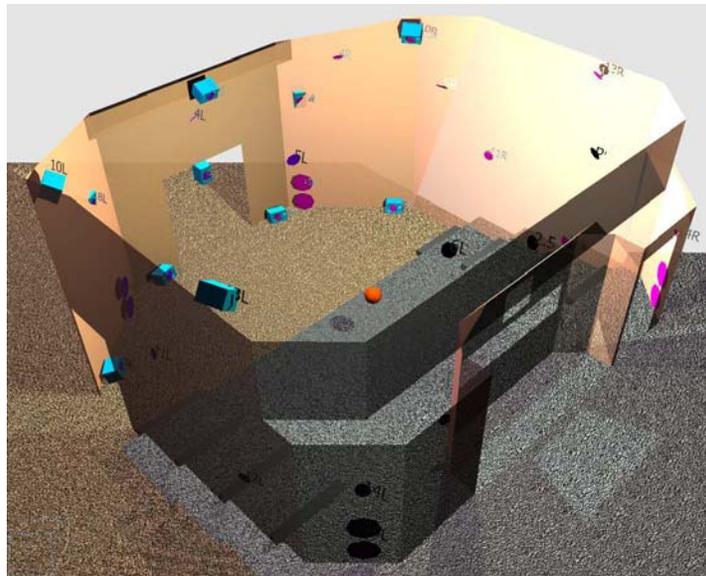


Figure 12: NASA LaRC Exterior Effects Room modifications.
Not shown are six satellite speaker locations in the ceiling.

5.0 CONCLUDING REMARKS

New capabilities for the subjective study of aircraft community noise have been recently developed at NASA LaRC. While the CNoTE system is capable of rendering recorded flyovers in a virtual 3D audio-visual environment, its true potential is best realized when used in conjunction with source noise synthesis. The combination of synthesis methods embodied in the ASoNG program with the simulation capabilities of CNoTE allow for subjective assessments of aircraft and their operations not yet in existence. The way forward requires additional developments for synthesis of rotary wing source noise and curved propagation paths. Feasible approaches have been proposed to address these requirements.

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