KEYS

Approaching ambience

Evaluation criteria for digital artificial reverberators

By Ralph Kessler

(translated by Conan Kirkpatrick and Gerd Räther)

Reverberation, produced naturally or artificially is one of the most important effects for any musical or spoken performances. This goes for concerts as well as for film or sound material that is prepoduced program material.

The purpose of this article is to explain the main criteria, for evaluating artificial reverberators and for understanding their parameters. First we'll provide a physical-technical introduction of the various ways they work and we'll show their typical applications. Above all, we'll discuss the problems of the "realism" of digital devices.

To avoid any misunderstandings: this is no comparison overview. Don't expect anything like this: "Device X of manufacturer Y sounds absolutely horrible." Evaluating reverberators is a very subjective matter and, moreover, depends a lot on the material to be reverberated.This article is more of a guide to help the reader judge the aforementioned statement for himself, with the addition "...with these parameters for this application!"

Readers with very technical understanding will disagree with the statement that there is no need for discussions like that after going through the manufacturer's manual, since it contains all relevant information, such as range, signal-to-noise ratio, frequency response, harmonic distortion, crosstalk attenuation and phase characteristic. Unfortunately, this is only partly true. Although an understanding of the aforementioned technical data is necessary, it is not sufficient for an authentic sounding reverberator.

Learning to hear

In this case, you can't use the same technical criteria than with other audio devices, like tape recorders or mixers. It is also important to take additional qualities into consideration, which you also want when choosing a musical instrument: Listening is the key factor here. If possible (works best in a quiet atmosphere inside a familiar studio).

- in connection with different program material for testing.
- when comparing several reverberators directly.

The program material should be adapted to once current and perhaps future needs. Generally it is safe to say short percussive sounds such as claves, a metronome, or a simple fingernail scraping make very good test signals. But synthesizers, drums and especially speech are very suitable too.

Also bear in mind the transparency of mixed sounds (see below). At this point it is recommended to go ahead and use a complete, dry mixture as a test signal. Check here whether the individual instruments inside the stereo image are still localized or whether the results are blurred. A good reverberation, contrary to many expectations, results in improved transparency and ambience! It shouldn't sound like a foreground effect; instead it should stand out for the fact that you may not be aware of it on a conscious level, but that you will miss it, when it is faded out.

A rule for the stereo distribution of the reverberation output: if you listen to it without the source part, you shouldn't be able to hear the famous hole in the middle (see below). An additional rule for devices with stereo input: the individual instruments must assume constant, but different positions in the stereo reverberation image - assuming, of course, the source has such positions. The reverberation must produce additional perspective arrangement, a phenomenon which can't be produced by devices with a simple mono input.

For comparisons to other reverberators, the test wiring should be arranged in such a way that it's possible to switch from one device to another, without any long crossfading, like by mute on the mixer. This direct A/B comparison prevents psychological masking effects during crossfading.

Again, we're only interested in devices which produce realistic ambience and not in "effect reverberators". The latter can't be subjected to any criteria. It is all a question of the designer's or the user's personal taste. Artists/producers who want thin or metallic or unnatural reverberation characteristics shouldn't hesitate to try out devices in the lowest performance categories - if they can live with the noise.

Some basics on room acoustics

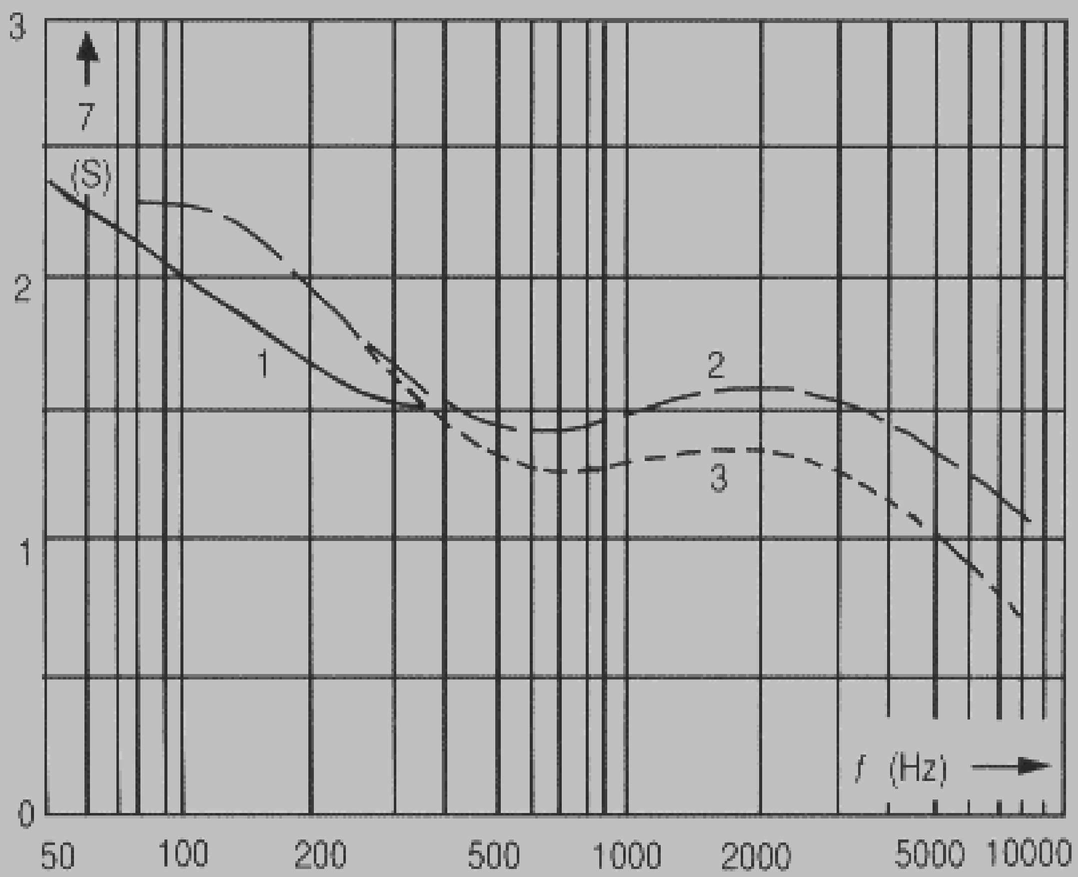
Judging the "realism" of reverberation requires a little basic knowledge about acoustics, in addition to a trained ear. Basically, the acoustics of a room may be described with two base parameters: reverberation time and density of the eigenresonance

The echo time and dimensions based on it were defined, researched, and put into a formula as early as 1923 by Mr. W. C. Sabine, a simplified version of which is shown here:

$$\mathbf{T = 0.163 * V / A}$$

T (also known as RT6O) represents time in seconds, in which the sound pressure level, after the sound source is turned off, drops to 60 dB, that is, to one millionth of the original sound density, **V** represents the room volume in cubic meters, **A** the sound absorbing capacity in square meters. Mathematically, it is derived from the sum of all adjacent areas, in each case weighted with the absortion level **ALPHA**, which can be obtained from tables for various materials.

An essential result was, that **A** is frequency-dependant: every material absorbs high frequencies better than low ones. Thus, high frequencies have smaller echo time than low frequencies.



The diagram shows the frequency dependant echo time for a music studio (curve 2) and for a music studio with curtains (curve 3).
The ideal flow is seen in curve 1.

Later this formula was modified by adding the socalled air absorption coefficient**o**, times the room volume **4*V**, to the numerator, producing a certain saturation effect: Large rooms, high frequencies and low humidity put a limit on the reverberation time, which cannot be raised, even with additional de-attenuation and an increase in room volume.

In this perspective, we take into consideration volume and surfaces, but not the geometry of the room. Now, everybody knows that a tube-shaped room sounds different than a hemisphere. The shape of the room will influence the number, distribution and amplitude of the eigenresonances (also called eigentones or modes), comparable with those of an instrument. In a way, this eigenresonance density is a fingerprint of the room. A good room distinguishes itself in that his eigenresonances do not blurr the sound.

The eigenresonance density can be measured by spectral analysis of pulse responses. The room should not be excited by a sine wave, in order to prevent wrong modes. Instead use filtered noise or simply a bang.

What sounds "good"?

Research was done in what rooms a music or speech reproduction was perceived to be especially pleasant. It became clear that the kind of program material such as organ music, synphonies, speech or pop music, has a considerable influence on the "ideal" echo time. For instance, the BBC recommends 0.3 - 0.4 sec for their voice studios, while for music studios, the tolerance is higher: 0.9 - 1.8 sec, depending on the style of music. Although these "ideal values" do depend on the room size, they depend more on the program.

The frequency dependency of the reverberation plays a vital role too. It has been discovered that reverberation is perceived to be at its most pleasant when the echo time rises at low frequencies and remains constant or drops at higher frequencies. This drop in the high frequency area may be obtained by variable damping of the walls.

In order to avoid inadvertent sonic coloration, the eigenresonance density should be as high as possible (around 1-3 per Hz in large rooms) and as frequency-independant as possible. Coupled with the mode density is the aforementioned pulse response or the reflection pattern as well. In this regard, it is necessary to mention the phenomenon "attack". This refers to the reflection density which increases with time and leads to an exponential rise in sound pressure after excitation, similar to the excitation time of an instrument.

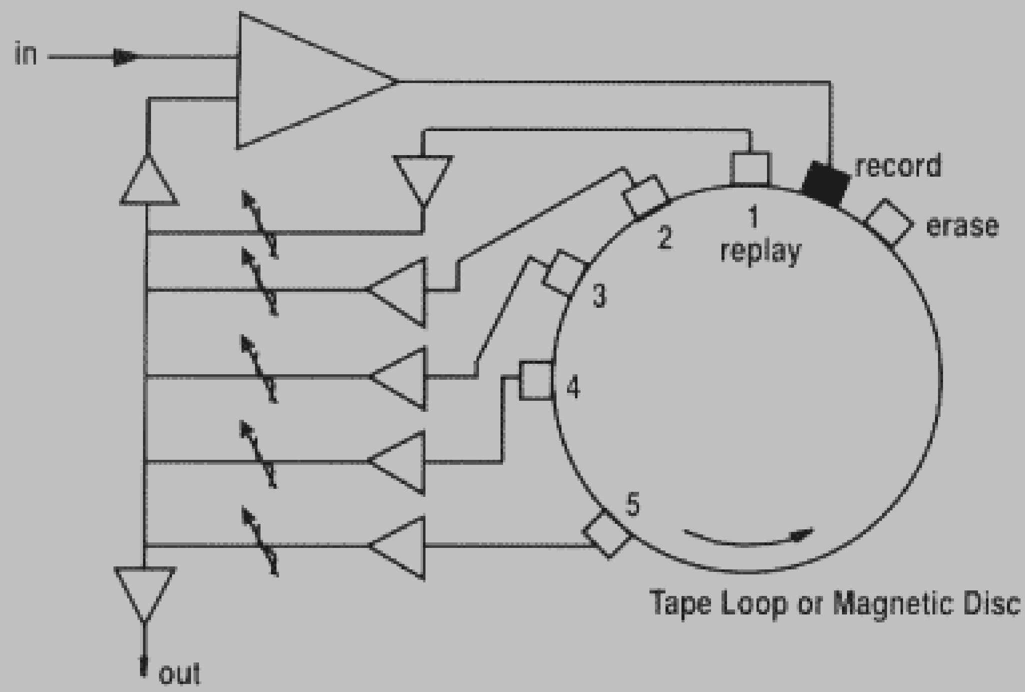
The process of artificial echo

In the early days of sound recording, people made do with real rooms. Microphones and speakers were set up in a room, specially arranged for that purpose. The setup was restricted to mechanical adjustments and the number of problems that arose varied accordingly.

A later way of creating reverberation, accepted to this day, is the plate. A steel plate was excited with electromagnetic converters, the result was sent back by pick ups; reverberation time could be adjusted by a variable-damping device. An intrinsic influence of the sonic characteristics was not possible. However, for these heavy monsters, roughly 2.5 m long and 1.5 m high, a vibration-free place was required.Their size was reduced by the introduction of a gold alloy. The sound results turned out very good, so they are still used today. By the way, the German company EMT and their EMT140/240 have created a patented quasi standard in this matter.

Instead of plates, other manufacturers used springs, saving room. Their sensitivity to vibrations was even put into creative use, by musicians like Keith Emerson.

The invention of magnetic sound recording gave rise to different technical approach. Short delays with feedback, cleverly connected made a rough approximation of real reflection patterns. This was realized through feedback of several replay heads to the record head. A drawback were the copying losses which reduced the signal-to-noise ratio for each renewed input of the signal to the feedback system and a very obvious periodicity in the "echo".



The diagram of an old tape echo device: a recorded signal is "replayed" by the five successive heads, with the delay time being determined by the distance of the heads from each other and the tape speed.

In the 70ies, with the the rapid technological development in micro electronics, the mechanical approach was finally completely replaced. At first, with modest success. Plates still sounded better than the new "cold" digital technology. At the beginning of the 80ies at last, some pioneers achieved spectacular developments: Chris Moore (and his Ursa Major Spacestation, which was actually a delay system from the 70ies) and David Griesinger (Lexicon 224 and 480) in the U.S. and Wolfgang Schwarz (and his legendary Quantec Room Simulator) in Germany.

Due to their excellent signal-to-noise characteristics, a multitude of available setups and, last but not least, due to their size, all-digital devices have conquered the market these days. Decoupling the hardware, which determines purely technical data, from the software, in charge of sonic aspect, makes for very flexible possibilities to determine real reverberation or create other effects.

The hardware basically consists of

- analog/digital converters mit mirror frequency filters,
- a computer as powerful as possible
- digital/analog converters mit smooth-out filters

Technical data are determined by converters and filters. If all you want is real generation of echo, a frequency response of up to roughly 8 KHz is absolutely sufficient, due to absorption in the air. What's much more important is that the frequency response be as constant as possible, that it makes no phase shifts (often underestimated, but it is important for the localization) and that the internal machine precision is high enough (eigenresonances can have an enormous range). 16 bit is no problem for today's converter. This makes roughly up to 93dB possible. The machine precision should top 24, maybe even 32 bit.

Digital simulation

There are two different approaches to digital reverberation simulation. The mathmatical signal theory guarantees an uncompromising solution. Since reverberation is a linear system, this is the rule: once you have the fingerprint, that is , the pulse response of a room, all you have to do is "only" convolute the input signal with the response (see also (B90)) to obtain the amount of simulated reverberation. This "convolution" is a complex mathmatical process that has to be carried out via Fast Fourier Transformation or, in the time region, via powerful computers, or via fast signal processors.

There are two drawbacks, though: the first one is that the delay time of the calculation will always be at least two or three times as long as the reverberation time, itself. Even more powerful computers can't do anything about that. Hence, this approach is not suitable for real time applications, although meanwhile it is applied in other fields (see also (R89).) However, the price of the necessary hardware are in the neighborhood of six digits - the second drawback.

The second approach is more promising. Reflexion patterns are produced with the help of delay feedback. The groundwork is based on research carried out by reverberation pioneer Prof. Dr. Manfred Schroeder of the 3. Physikalisches Institut in Göttingen, Germany, in 1962 ((S62)). Simply put, delay line feedback is actually an echo. If the delay is decreased to milliseconds, the reflection pattern turns increasingly dense. However, at the same time, you get a combfilter which produces severe sonic discolorations. If the feedback approaches one, you get more reflections (and more reverberation time with it), but also more discoloration.

A further problem is the periodicity of the result. It has a harsh, metallic sound to it.

Apparently, real reverberation is more subject to the laws of chaos than to any fixed structures. That's why Schroeder suggested arranging four combfilters with different delay times in a parallel circuit, in order to reduce periodicity, as well as arranging two all-passes in series. All-passes produce an additional increase in reflection density. Furthermore, they have the advantage of leaving the frequency response untouched, which is why they can be arranged in series without any problems.

This configuration sounded very good already, and so the fondation was laid. Don't forget that Schroeder had no hardware back then, that could carry out all calculations in real time. He was working with huge mainframe computers and sometimes had to wait for hours before he could listen to one minute of processed sound.

Many institutes and companies carried out further work. In addition, some were also working on first discrete reflections, which exist in nature, but which also have chaotic pre- and post echo (see also Haas Effect) which are hard to simulate. The theory of initial reflections (->localization) also covers a very broad area and is beyond the scope of this article.

Fact is, the attack of the simulated room is a very important component for the realism of the latter and that a good echo makes additional first reflections unnecessary - except in the case of special effects like train station halls or mountains, where a real discrete echo is heard.

Another way of getting the metal out of the sound of small rooms in particular is a slight modulation of frequency. This method may seem very efficient at first, but it has the disadvantage that the reverberation inevitably varies in the pitch. In a real acoustical situation this would only occur if the performer constantly moves to and fro from the audience, always at random, which is unlikely. More like a random phaser, it is strictly rejected by "purists", especially since the perception of constant frequencies tends to be very subjective. You can hear this effect very well during piano recordings, hence the term "wow-free piano".

Particular attention should also be paid to the correlation of the stereo output. As already mentioned, no hole may appear in the middle of the stereo during reproduction, as would happen when applying two separate machines (0% correlation) for the left and right channel or even simply produce the right signal, phase-shifted, from the left one by 180 degree. 50% is a good value that still guarantees monocompatibility.

In order for stereo input to be compatible to dummy heads, the signals may not simply be mixed inside the device but instead have to be led into the "room" separately. It's the only way to prevent interferences from negatively correlated input signals.

Our studies reveal that real-sounding simulation takes a whole lot of patience, readiness to experiment, good ideas and, above all, a set of sensitive ears. The hardware of the computer has to be optimized too. Although digital signal processors are becoming faster and faster, their p/e ratio does not make them good assistants.

Obviously, the manufacturers protect "their" algorithm like the holy grail. But you can't blame them. If you were to visit Steinway and politely ask them about the ingredients of the paint they use on their pianos, they'd likely tell you everything, but not that.

The parameters

Opinions differ strongly here. Some think that the flexibility of a device can be determined be the number of it's parameters, while others believe that two's company, three's a crowd: "reverberation time by itself works fine for me!"

But away from the merry polemics and on to the different applications. If the idea is to simulate real acoustics, the parameters reverberation time, room size (rough for the eigenton density) and an opportunity to influence the frequency dependence of the reverberation time (here the qualities differ already). The control behavior in the high frequency region corresponds to variable damping.

Additional features for effect reverberation, depending on the manufacturer:

- predelay (for creating more distance or depth)
- diffusion and density (eigenton density fine)
- time and amplitude of the first reflections (the better the reverberation, the more unnecessary),
- absorption by the air (saturation effect, see above: room acoustics) and
- shape, propagation and decay of the envelope.

A quick way of getting your desired setup is to take a program with well-suited music samples and to modify that.

Gate- and reverse reverberation are new ways of which, unfortunately, there is too much in almost all mixes of pop music. It is an interesting effect which has been suffering from a clear case of overkill. Then again, gate reverberation (simulation of a reverberation which is sent through noise gate) offers the interesting opportunity for some devices to not cut off the reverberation completely after the gate time but to crossfade it into another echo chamber. Both echo chambers should be freely defined. This opens the door to some tremendous opportunities!

For concerts, reverse reverberation is of little use, and in studios many people still prefer the good, old "Play the multitrack the other way around"-method.

Applications

Reverberators should always be used with aestetical aspects in mind. What basically applies to a stage setting, whose quality depends on each component, also applies to sound reproduction: music and its environment must be in perfect sync with each other. In my opinion, a good and well-applied reverberator shouldn't be heard at all. If anything, a good mix should sound ambient. In order to create a vivid experience, it has to show many levels of depth and as much transparency as possible.

In a studio atmosphere, especially in the field of movie and TV adaptation, it is actually possible nowadays to create acoustical illusions, with the right devices and their sophisticated application. However, there is nothing worse for an artist's product than unwanted, disillusioning elements (to use the example of the stage scenery again: how do you like it when a marble bank creaks?) This is what separates cheap products from Lucasfilm productions (the contacts of which we'd rather not discuss here, but which really are excellent productions, right down to every detail).

Yet, it isn't only the effort that counts, but also clever use and intelligent ideas. Due to too much daily routine, many music mixes today tend to sound very barren, certainly a result of run-of-the-mill ways of listening. Moylan ((M87)) came up with some interesting and inspiring methods for systematic aural training.

In the concert and PA field, a reverberator may be used to compensate for poor room acoustics. Sound control in the sum may be the conventional solution but it is useless, as far as the range behavior of the room resonances is concerned. A feature that increases the eigenton density of the room and decreases frequency dependency, resulting discoloration-freer sound in the whole room is a lot more efficient.

Future trends...

are, for example, a higher number of in- and outputs (see (G88)). There are debates right now on how many sound channels HDTV(High Definition TeleVision) should get; in any case, the number will be greater or equal to four. More inputs will provide better ambient separability of several sound events.

User panels will need better ergonomical design. The central, graphical use of several devices that can also be coupled will be made possible by use computer interfaces. This means a growing demand for ergonomical software.

For the PA field, there are several projects for the correction of the existing room acoustics. One of these, ACS(Acoustic Control System), already installed in Germany, is based on the process of "acoustic holography" (see(B89) and(M9oc)). Another process, introduced by Mr. Griesinger at the *Tonmeistertagung* in Karlsruhe, Germany, will make it possible to raise the feedback limit of a concert room to roughly 12 dB.

Audio engineers and architects can already obtain simulation programs for mainframe and home computers, with which they can reach conclusions about the sound of a room in any place, via a simulation with input regarding room geometry, surface materials, sonic sources and loudspeaker positions - all this before the room is actually built. This runs under the key word CAAD (Computer Aided Acoustic Design, see (M9OA) and (M9OB)). The systems of Mr. Richter and Mr.Persterer ((R89)) even allow you to listen to the result right away.

Conclusion

Explaining the scientific basics and functions of modern reverberators describes criteria, which makes it easier for the reader to judge the quality of artificial echo and to use it as an acoustic means of creation. These are: frequency dependence of the echo time and its envelope, eigentone density, reflection pattern (caution: metal!), attack, stereo base (correlation, mono compatibility), compatibility to dummy heads (stereo input), wow-free piano and the ability to form unreal interesting effect rooms. Technical data of the hardware, usability and, last but not least, the enhancement should be considered too.

Always think of reverberation and the instument or the mix as a unit. Hence, "one-size-fits-all" arrangements are not acceptable in sound-esthetic terms and only produce a dull, run-of-the-mill sound. Present technology allows for simultaneously varying the ranges of the room acoustics on several acoustic levels and experimenting with them, which, unfortunately, is done much too seldom. Well-dosed range modifications of the acoustic environment are found too seldom in mixing plans or even in the scores of modern componists.

The author

German engineer Ralph Kessler, born on Dec 4, 1957, majored in electrical engineering at the University of Saarland, Germany, concentrating on audio and visual technology, as well as on digital signal processing. For several years, he worked as a freelance audio engineer in the studio and concert business. Cofounder and chief developer of the enterprise PLS GmbH/Saarland (production and hiring of lighting and sound equipment). For three years, he was project manager of the research and development team of Quantec Tonstudiotechnik GmbH in Munich. Since February 1989, he has had his own company, Pinguin Computersysteme und digitale Studiotechnik, Hamburg. Consulting and project development. Last publication: AES, Montreux, Francen in 1990: Optimizing the work surface in digital mastering studios.