10.8 Effects

10.8.1 Reverb

In every (interior) space♣, floor, ceiling and walls reflect sound. Individual reflections with a temporal distance of more than about 50 ms are perceived as individual echoes. Reflections arriving with smaller intervals create a perception of reverb. Since sound propagation in a room (i.e. in air) is – with very good approximation – a linear process (LTI-system), the system "room" may be described by its impulse response (**Fig. 10.8.1**), or by its transfer function.

In order to obtain the **impulse-response of a room**, the room is acoustically excited via a sound-impulse: this could be an electrically generated spark (spark-plug), or a bursting air-balloon, or a hand-clap, or something similar. In reality, such an excitation signal is not the Dirac impulse known from systems theory but a real sound impulse of a duration larger than zero and an amplitude smaller than infinity. A microphone picks up the sound pressure at the measuring location and its magnitude is depicted as a graph over time (**Fig. 10.8.1**). Since there are any number of excitation- and measuring-locations in a room, there is also a corresponding multitude of impulse responses. With passing time, the reflections become weaker and their density increases. The first reflections (early reflections) serve the auditory system to obtain information about the size of the room. The speed at which the reflections decrease is a measure for the absorption in the room. Slow decay results in a *reverberant* impression, the opposite impression is called *dry*.

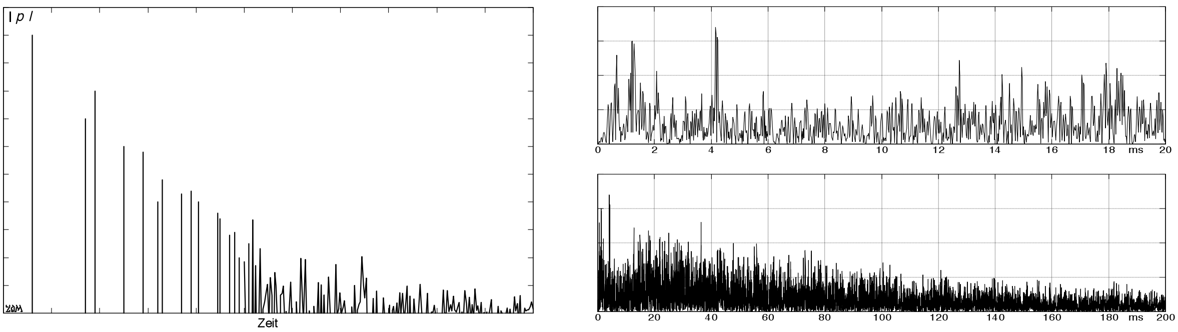

Fig. 10.8.1: Impulse response of a room (reflectogram). Left: model, right: examples for a real room.

Reverb-springs, reverb-plates, magnetic-tape devices or electronic delay-systems are used to simulate real room reverb. For guitar amps, the reverb spring has established itself as a standard from the early 1960s. After being available for some time in a stand-alone device only, it was first integrated into Fender's Vibroverb-Amp from 1963. Two steel-wires wound into helical springs serve as delay lines with mechanical waves running back and forth within them. The basics of this delay-principle had been investigated at Bell Labs, and employees at the Hammond Company had developed it into a product ready for series production. The "reverb can" (or "reverb pan" or "reverb tank") as it is manufactured today by Accutronics holds steel wires of a diameter of 0,4 mm that are wound into a helix of 4.2 mm outer diameter. On the side of the actuator, an electromagnet creates forces within a small permanent magnet that deforms the wire; the sensor side operates similarly: the movement of a permanent magnet induces a voltage in a sensor-coil.

♣ In the following, the term „room" is used; it is always associated with a space having reflecting boundaries (room, hall).

The originally used reverb system included 2 subdivided springs; newer systems are also available with 3 springs (**Fig. 10.8.2**). The connecting point (which is not located exactly in the middle) between the subdivisions creates additional reflections. As a current flows through the actuator coil, the permanent magnet creates torsion in the steel wire that results in a flexural wave. The latter runs along the wire and reaches the other end after about 30 – 40 ms. With the two springs having slightly different dimensions, the delays differ as well (43 ms and 41 ms, according to the manufacturer).

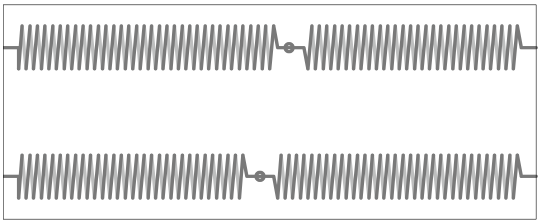

Fig. 10.8.2: Reverb system with 2 subdivided springs (Accutronics). Alternatively, systems with 3 springs are also in use. The flexural waves travel along the springs and are reflected at both ends. At the connecting points, reflections are created, as well – these are, however, less pronounced.

Compared to real room reverb, spring-reverb shows a significantly different behavior: sound propagation in space is three-dimensional and non-dispersive, while in the spring, it is one-dimensional and dispersive. If we had only a single spring without subdivision, we would get merely a sequence of echoes (e.g. after 30, 90, 150, 210 ... ms). These echoes would be equidistant, separated by the time it takes the sound to travel back and forth in the spring. Conversely, the average reflection density in a real room increases with time squared t^2. Each reverb spring consist of two parts connected via a ring. If we take the spring to be a mechanical line (compare to Chapter 2), the ring acts as mass loading which reflects in particular the higher-frequency waves. Thus, two echo-systems are connected in series, and the reflected wave obtains a t-proportional component at high frequencies. The second (subdivided) spring connected in parallel, on the other hand, merely doubles the density of the reflections without changing the exponent of the time-dependency.

Fig. 10.8.3 shows spectrograms of the impulse response: in the left picture for the two-spring system, and in the right picture for the same system but with on spring clamped down (such that no vibration could be formed). Clearly, there is not really any <u>one</u> delay-time per spring. Rather, a frequency dependent group-delay is created due to the dispersive propagation: high-frequency components require about 50% more time to arrive than low-frequency components.

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Fig. 10.8.3: DFT-spectrograms of the impulse response: two reverb-springs in parallel (left), one spring (right).
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The usable upper frequency limit of a spring-reverb is about 5 kHz – fully adequate for a guitar amp. **Fig. 10.8.4** depicts a third-octave analysis taken from an excitation with pink noise [3]. The different curve shapes in the high-frequency range result from different loading of the sensor coil: the inductive source impedance acts, in conjunction with the load capacitance, as a second-order low-pass. This low-pass can generate a slight resonance peak at 4,5 kHz (----) given the appropriate dimensioning. A measurement with a sine sweep enables us to take a closer look at the fine structure but requires consideration of the extremely long attack and decay times. Even using a sweep-duration of 2 minutes, the system cannot actually "settle": the exact position of the maxima and – in particular – the minima depends on the measurement parameters (resolution, sweep duration).

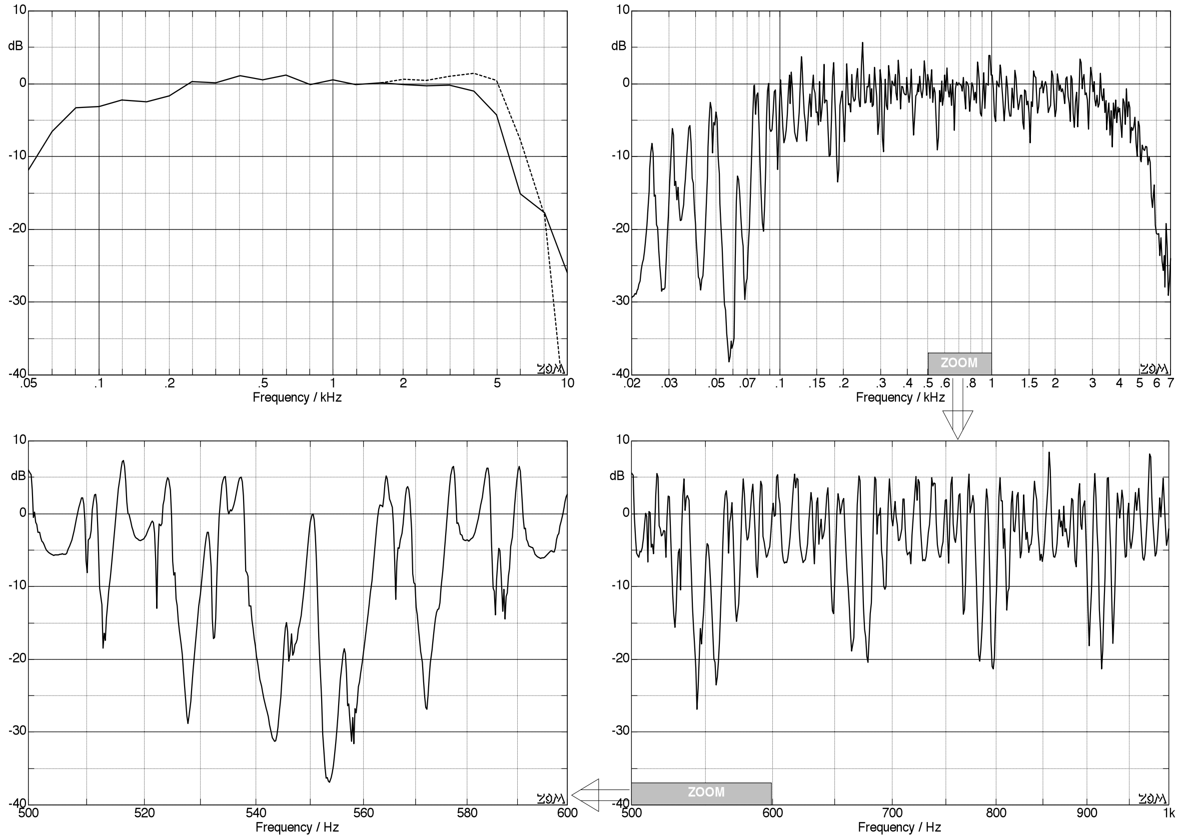

Fig. 10.8.4: Accutronics reverb 4AB3C1B, current input. 1/3-octave-analysis (upper left), sweep measurements.

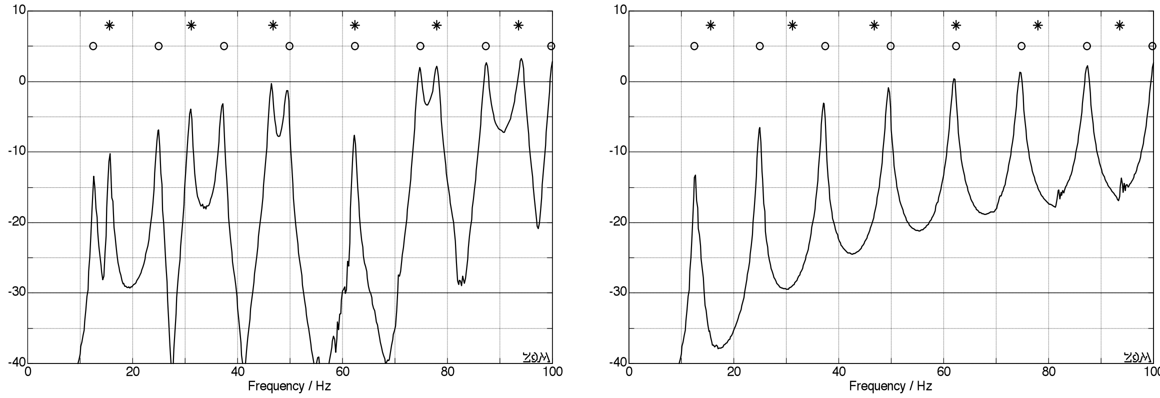

Fig. 10.8.5: 4AB3C1B, sweep-analysis; 2 springs in parallel (left), one of the 2 dampened (right).

In **Fig. 10.8.5** we see the lower-order natural frequencies with enlarged scaling. The lowest of these natural frequencies are located at 12,5 and 15,6 Hz, respectively, for the two parallel springs, and the higher natural frequencies are found at multiple integers thereof (circles and stars in the figure). At 63,5 Hz, there is an interaction of the 5th and 4th natural frequencies, the result being a beat-effect of a very long periodicity. The right hand section of the figure clarifies that the minima are the effect of destructive interference: with only one spring active, the comb-structure is much more even. This regularity also supports the hypothesis (on the basis of transmission-line-theory) that the ring positioned in the middle of the string (and connecting the string subdivisions) works as a scattering body predominantly at high frequencies.

The decay of the reverb is usually expressed as the **reverberation time** T_{60}: this is the time it takes for the (1/3-octave) level to decrease by 60 dB after switching off the excitation signal. In the high-frequency range, we find a by-the-book behavior (**Fig. 10.8.6**): the level decreases linearly with time. The reverberation time is 2,5 s. At low frequencies two superimposed decay-processes reveal themselves: an early fast decay and a subsequent slower decay. In such cases the perception-relevant **early-decay-time** is specified as six times the duration it takes the signal to decay from -5 dB to -15 dB. The reference level is the averaged level for the steady state excitation. We can see in the figure that this EDT (T_{10}) is, again, 2,5 s for the chosen 1/3-octave band – the subsequent slow decay can be attributed a reverberation time of about 12 s. In guitar amps, the frequency range below about 300 Hz does not have a particular importance (for the reverb signal): usually a high-pass will effectively dampen the lows in order to suppress any annoying booming.

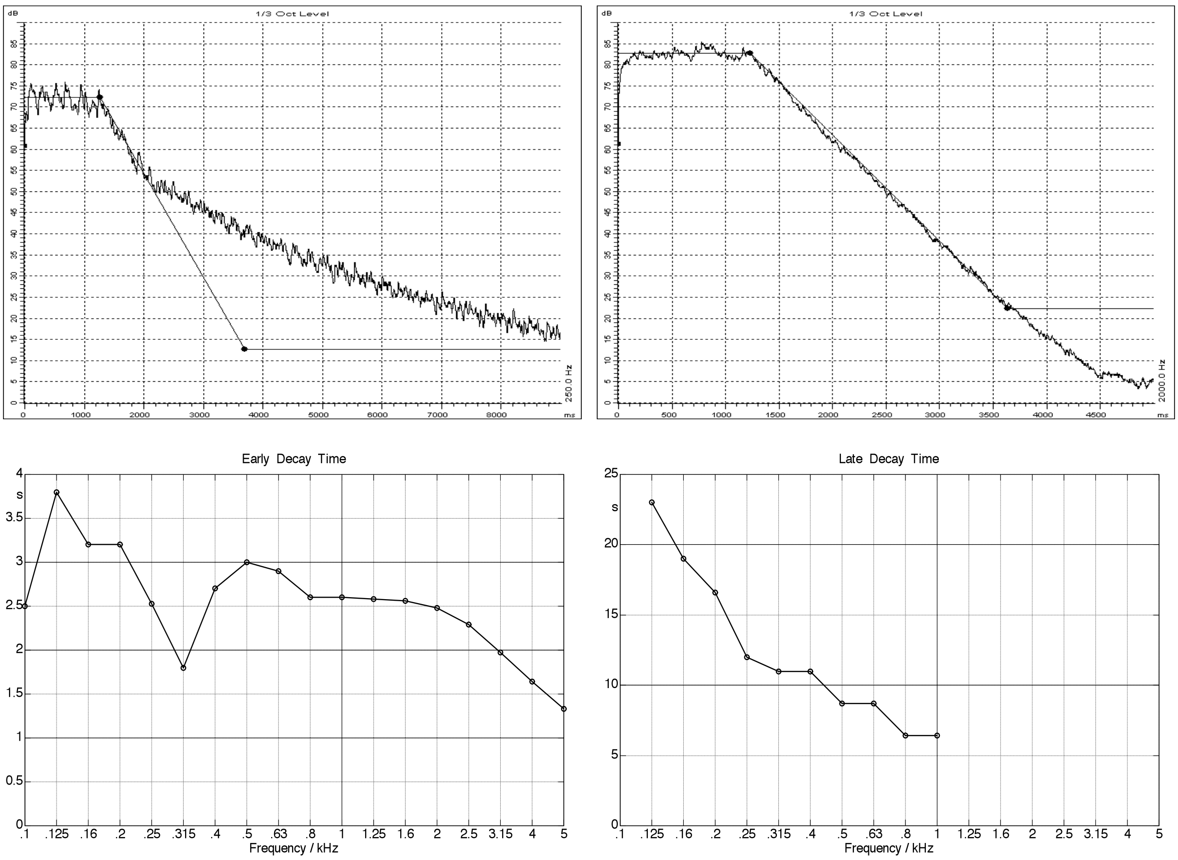

Fig. 10.8.6: 1/3-octave decay analysis, Accutronics spring-reverb 4AB3C1B. 250 Hz (left), 2000 Hz (right). The lower pictures show the frequency-dependency of the reverberation times.

The **kink** in the reverberation curve is probably an effect of the scattering mass in the middle of the spring. The waves reflected at it reach the (partially absorbing) end of the spring twice as often as the waves passing through, and we get two types of standing waves. This hypothesis, however, was not extensively tested via experiments – a full dampening of one of the two springs did in any case reveal clearly that the kink is not a result of different absorption in the two springs: the *individual* (subdivided) spring shows the same kink-behavior as shown in Fig. 10.8.6.

In order to achieve a reasonably frequency-independent transmission with the investigated Accutronics reverb system, driving it with a **stiff current source** is conducive (Fig. 10.8.4). With a voltage source, a strong treble-loss would occur due to the substantially inductive input impedance of the actuator coil. A high impedance source is automatically made available by tubes in a common-cathode circuit. However, the Accutronics system has such a low input impedance (8 Ω at 1 kHz) that an extreme mismatch would result. The optimum current drive happens with a source impedance of about 100 Ω – this is obtainable from a tube plate only using a transformer. Impedances are transformed with the transformation-ratio squared: a transformer with a 25:1 ratio will yield – for a tube output impedance of 62,5 kΩ – the appropriate secondary source impedance of 100 Ω. Fender's stand-alone reverb unit **6G-15** employs a 6K6-GT to drive the reverb-transformer; this low-power pentode features an internal impedance♣ of 90 kΩ (the 6V6-GT has 50 kΩ). If the reverb is integrated into a guitar amplifier, it is almost always a **12AT7** that is deployed; it has an internal impedance of merely about 40 kΩ per triode. Since both triodes in the tube are connected in parallel (!), the source impedance drops to 20 kΩ. On top of this, the reverb transformer 125A20B has a transformation ratio of 50. As a consequence, the reverb system is effectively driven from a voltage source above 1 kHz, and a corresponding treble-loss.

If the reverb system were a linear device (in the sense used in systems theory), we could insert a corrective filter at any point in the amplification chain and boost the missing treble. However, both the reverb spring and the tubes are **non-linear** devices, distorting at high signal levels, and generating noise and rumble in the small-signal range. Filter-design therefore always includes a component of dynamics-optimization, as well. In a typical Fender amp, predominantly the low frequencies are attenuated ahead and after the reverb spring – the treble-boost-enabling current-drive is only rudimentarily taken advantage of.

In **Fig. 10.8.7** we see the transmission frequency response from the ECC83 ahead of the reverb branch up to the plate of the ECC81. The filtering is done via two sections: the RC-high-pass ahead of the ECC81 (f_g = 320 Hz), and the inductive input impedance of the reverb-transformer. Since the double-ECC81 has quite a low output impedance, we see the reverb driven by a current source only up to about 1 kHz – in the upper frequency range, the tube acts as a voltage source. The voltage transmission-factor of the reverb system is shown in **Fig. 10.8.9** – in contrast to the situation with a current source (Fig. 10.8.4) we find a pronounced treble-loss. The connection to the reverb-potentiometer is done via a 3-nF-capacitor resulting in another high-pass filtering (350 Hz). The overall reverb-branch has a bandpass characteristic centered around 600 Hz while the direct signal only receives a mild treble-boost. The circuit depicted in **Fig. 10.8.8** is typical for many Fender amplifiers – some do have a 2-nF-capcitor connected in parallel to the reverb-pan output in order to create a small resonance peak (Fig. 10.8.4).

♣ Data-sheet specifications. ♠The 12AT7 (= ECC81) is working at an atypical operating point!

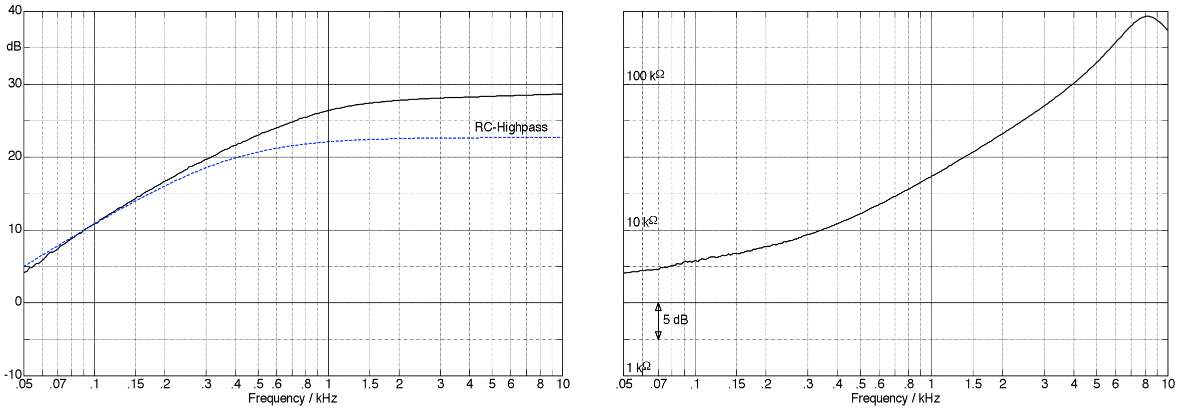

Fig. 10.8.7: Transfer characteristic from the ECC83 up to the ECC81 (left); transformer input (right).

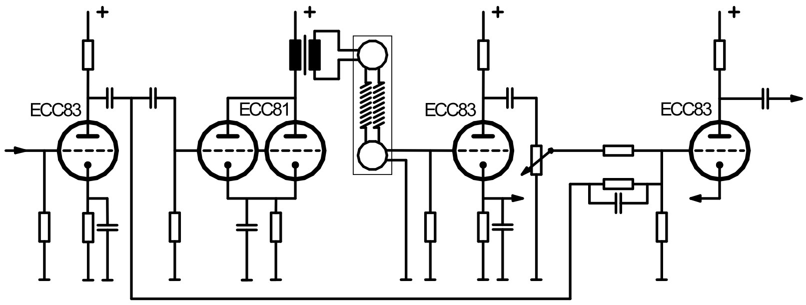

Fig. 10.8.8: Circuit of a typical Fender reverb. The reverb-spring input is of low impedance, the reverb transformer has a 50:1-ratio.

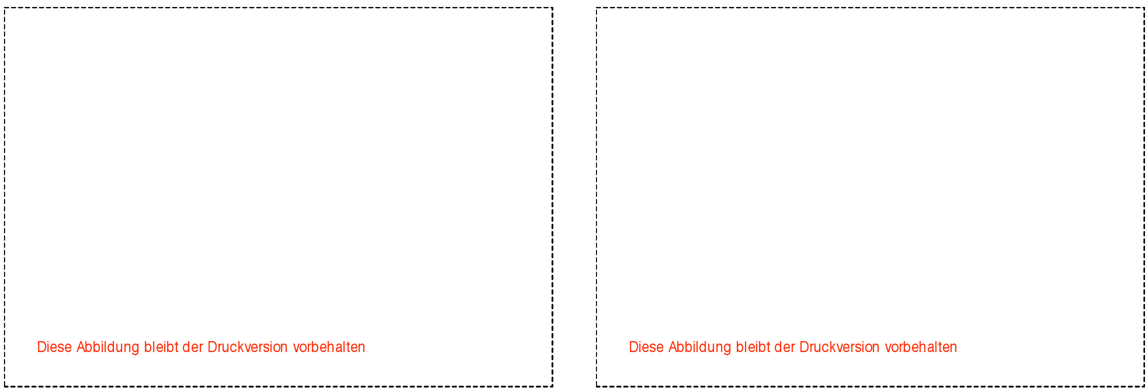

Fig. 10.8.9: Smoothed voltage transmission factor of the spring-reverb system (left). Transmission factor of the overall reverb-branch (right ——), and of the direct signal (right -----). Both curves in the right-hand picture show the transmission from the plate of the ECC83 ahead of the reverb branch up to the last ECC83 in the reverb branch. (Fig. 10.8.8). **These figures are reserved for the print-version of this book.**

The **Accutronics reverb-pans** are coded with 7 characters, e.g. 4AB3C1B. The individual characters indicate:
1st position: type. At the time of writing the Types 1, 4, 8 and 9 are available.

Accutronics reverb-pan Type 1 and Type 4			Accutronics reverb-pan Type 8 and Type 9		
2nd position = Z_{in}		3rd position = Z_{out}	2nd position = Z_{in}		3rd position = Z_{out}
A = 8 Ω	D = 250 Ω	A = 500 Ω	A = 10 Ω	D = 310 Ω	A = 600 Ω
B = 150 Ω	E = 600 Ω	B = 2250 Ω	B = 190 Ω	E = 800 Ω	B = 2575 Ω
C = 200 Ω	F = 1475 Ω	C = 10 kΩ	C = 240 Ω	F = 1925 Ω	C = 12 kΩ

4th position = reverberation time: 1 = short, 2 = medium, 3 = long.
5th position = chassis connected to: A = Input + Output; B = Input; C = Output; D = chassis insulated.
6th position = pan lock: 1 = no lock.
7th position = preferred mounting orientation: A = ⊔; B = ⊓; C = ⊏; D = ⊐; E = ▯; F = ▯;

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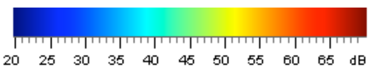

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1: Accutronics (4AB3C1B), 2 springs, subdivided.
2: Accutronics (9AB2C1B), 3 springs, subdivided.
3: low-cost, 2 springs (17 cm), not subdivided.
4: low-cost, 2 springs (14 cm), not subdivided.
5: digital reverb, "Spring-Reverb".
6: digital reverb, "Spring-Reverb".
7: digital reverb, "Room-Reverb".

Fig. 10.8.10: Various reverb spectra.

In **Fig. 10.8.10** we find a comparison between the reverb spectra of various reverb systems. The two Accutronics tanks represent the **spring-reverb** standard: they measure about 36 cm and are subdivided in the middle, with clearly visible individual reflections and dispersion. The spring-reverbs 3 and 4 are only of half the size compared to the Accutronics; they have no subdivision. The dispersion here is much stronger compared to systems 1 and 2, and the bandwidth is smaller, as is the reflection density – and the price is, of course, lower also. The pictures 5 to 7 show spectra of **digital reverb systems**. System 5 is marketed as “Spring Reverb” but has little similarity to an actual spring-reverb. In system 6, we may at least surmise that the developers sought to model the spring-typical dispersion although the result is not very authentic. System 7 is offered as “Room Reverb” and it does differ from the previously shown systems in that the strong periodicity is gone. The limited bandwidth of only 2,5 kHz is probably due to the computation power: the larger the bandwidth, the more load on the signal processor. The reverb spectra of a real room (**Fig. 10.8.11**) show, in comparison to the models, a much higher reflection-density and no discernible periodicity. The spectrogram has only limited meaningfulness here: since the DFT on which it is based cannot provide a high selectivity at the same time in both the frequency-domain and the time-domain. Nevertheless, the spectra shown enable us to get a basic insight into the individual reverb structures.

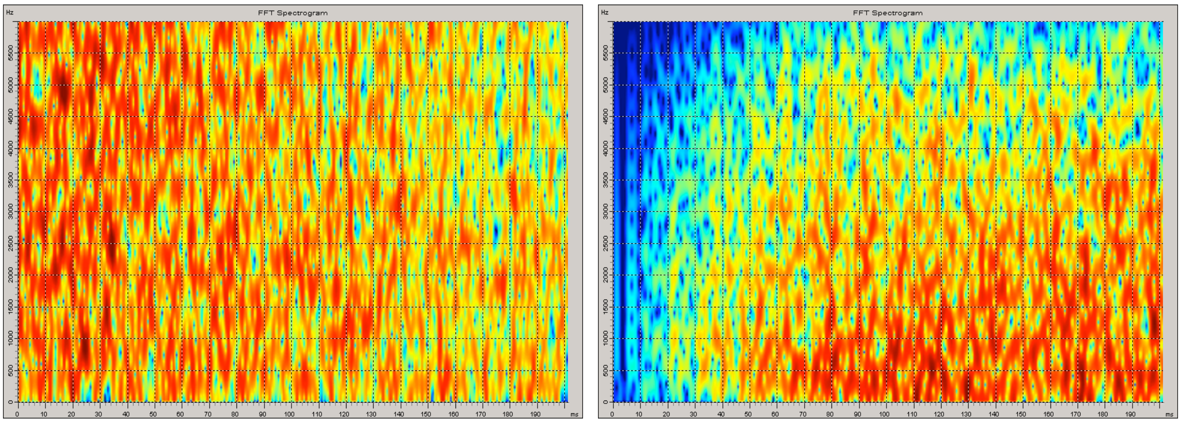

Fig. 10.8.11: Broad-band spectrum of a real room (left). Digital reverb of a studio-grade effects-processor, “large room” program with pre-delay and slight treble attenuation (left)

We should stress that every one of the reverb-systems discussed here can serve to generate a quite useful reverb for guitar. The responses following a short impulse may sound somewhat strange, but with a guitar such an excitation does not normally occur. Of course, compared to a real room, the wavering wash created by a spring-reverb has a somewhat outlandish sound at the first moment. However, the “room reverb” sounds just as peculiar in comparison if we have just listened to a Fender spring-reverb. There is a good reason that professional reverb devices offer a multitude of special reverb parameters to adjust such that the sound can be taylored to individual tastes and needs. In most cases, it is real rooms that are to be modeled (living room, hall, church, stadium), while an authentic digital simulation of a spring-reverb is not found that often. Maybe this is the reason why there are still “real” spring-reverb system and devices on the market.

 Translated into English by Tilmann Zwicker