10.8.2 Vibrato / Tremolo

From a systems-theory point-of-view, a vibrato- or tremolo-system is a modulator with time-variant transfer-characteristics – it changes the signal-amplitude or -frequency. **Leo Fender's** usage of these terms for his guitars and amplifiers has created a big mix-up that the music world has not really recovered from even today; a clear assignment between the terms and the respective function has not reestablished itself: does the vibrato effect in fact change the pitch – or is it the volume that varies? Fender's Stratocaster, protected since 1954 by US-Patent No. 2741146, according to the patent description holds a "*tremolo* device" to change the pitch. 50 years later, Fender brochures still use the term in the same sense. However, in a Fender service manual from 1968, the corresponding unit on a Mustang guitar is suddenly called *vibrato*, although on the same page the term *tremolo* is used for the Stratocaster and the Bronco. Similar confusion happens with the amps: *vibrato* is originally used for amplitude-modulation (change in loudness), and the *Vibro*lux amplifier indeed includes this function. How about the *Tremo*lux? Same – it's the identical effect. Does that feel complicated? Yep, without a doubt: there is a Tremolux with a tremolo-pedal♣, and also a later one with a vibrato-pedal. And, sure enough, there is a Vibrolux-version with a tremolo-pedal – and also one with a vibrato-pedal. The circuit that generates the effect, is always based on the same principle: originally it was a time-variant grid-bias that varied the amplification factor; later a light dependent resistor (LDR) illuminated by a blinking light – in any case the typical amplitude-modulator that was most often termed "vibrato" in the Fender brochures. Not always, though: what does the 1968 Fender brochure designate the built-in amplitude modulator for the Princeton Reverb (sporting a "Vibrato" pedal)? Right you are: it is called a tremolo.

Fender did offer not just this one modulation effect: in 1959, the **Vibrasonic** amp received a special circuit generating a mixture of frequency modulation (FM) and amplitude modulation (AM). In the mid-frequency range there was mainly FM, and in the treble and bass ranges an AM working in opposite directions: as the treble got louder, the bass got softer, and vice versa. This same circuit could be found at the beginning of the 1960s also in the Concert, Bandmaster, Pro, and Super amps, and in a slightly modified version in the Showman and the Twin. Its reign was short, however: it soon was replaced by the LDR-amplitude-modulator. With one exception all these effects were designated "vibrato" at Fender; just for the Princeton reverb the same effect was called "tremolo" – as mentioned above.

In summary: at Fender, "tremolo" is often (but not always) used for FM, and "vibrato" often (but not always) stands for AM. The classical (and scientific) definition is the other way 'round: tremolo = AM, and vibrato = FM.

How are these two effects **perceived** differently in our **hearing**? Surprisingly: not to a big degree – as long as the modulation is not too strong. The reason is that pure FM does not occur in normal situations: due to selective resonances in speakers and, especially, in the rooms we listen in, FM always generates an additional AM [e.g. 3]. The latter may even be detected (for small modulation indices) more easily by the hearing system. It is difficult to generate FM with a strong modulation index while it is much easier for AM. Here we may find the reason why Fender says good-bye to FM in the early 60s, and fits the low-cost and highly efficient LDR-modulator into all his amps. The following circuit descriptions focus predominantly on the well-documented and trend-setting Fender amps – well aware that other manufacturers have also developed and successfully marketed vibrato/tremolo-circuits.

♣ Label of the footswitch-jack

In your typical tube amp, a triode generates the low-frequency signal (**LFO** = low frequency oscillator) while the modulation itself happens in another tube (or more tubes), or in the **LDR**. The LFO is of a relatively simple build: a tube in common-cathode configuration with frequency-dependent feedback. The tube inverts the signal from plate to cathode, and consequently the feedback circuit needs also to invert; both inversions result in a phase shift of 2π, which is the requirement for self-excitation. In addition, the loop gain needs to be larger than one – easily achievable with a tube. **Fig. 10.8.12** shows a circuit as it is frequently utilized (Fender, VOX, and many others). The feedback branch consists of a 3rd-order high-pass with a variable resistor that adjusts the oscillation frequency between about 3 Hz and 11 Hz. Since there is no amplitude control, the generated signal is not of perfect sine-shape – the system is non-linear and therefore there is, strictly speaking, not really a transfer function as such. This should not be seen as a problem, however, since the approximation achievable with the linear model is perfectly practice-oriented and therefore adequate for the present context.

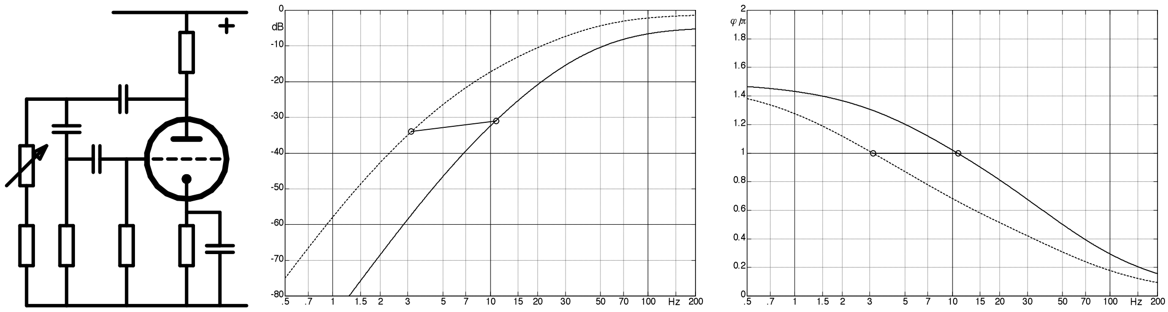

Fig. 10.8.12: LFO-circuit in a tube amplifier; magnitude- and phase- characteristics of the feedback network.

From about 1963, Fender amplifiers were fitted with an amplitude-modulator that used an opto-coupler: an **LDR** intermittently illuminated by a **glow-lamp**. The required control signal was tapped (with high impedance) from the circuit described above, and fed to the glow-lamp via the second half of the double triode (ECC83). Due to the operating point chosen for this second triode, a significant current is flowing only during a relatively short part of the LFO-period, and the glow-lamp lights up only for a short time. The resistance of the LDR decreases when lit and causes – integrated into the parallel branch of a voltage divider – a signal-attenuation (**Fig. 10.8.13**). Significant slurring of the envelope occurs due to the relatively long recovery time of the LDR – this is, however, rather beneficial to the auditory perception.

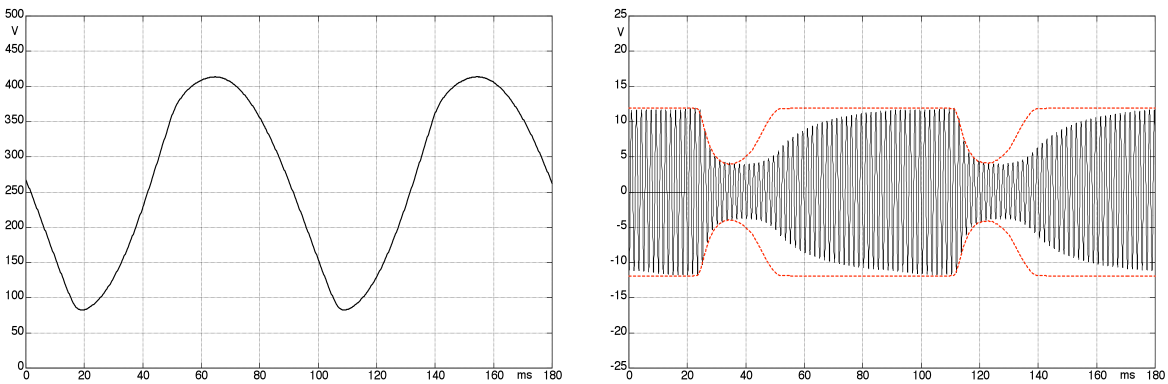

Fig. 10.8.13: LDR-modulator. LFO-signal at the plate of the oscillator-tube (left), 600-Hz-sine-tine modulated by the LDR (right). Dashed: imaginary effect of a modulator with zero recovery time.

Preceding the LDR-era, Fender deployed tube modulators. There are three types of circuits: screen-grid modulation in the power-stage, modulation in the phase-inverter, and, for single-ended power amps, modulation in the intermediate amplifier stage. The amplitude-modulation is achieved simply by shifting the operating point: a superposition (addition) of an AC voltage of very low frequency periodically moves the operating point into the end ranges of the characteristic curve. Here, the slope of the latter (and thus also the gain) is smaller than in the middle range: the gain becomes time-variant. The non-linear signal distortion also created could be accepted as an additional effect; the low-frequency parasitic signal also occurring (even without input signal from the guitar), however, requires including additional high-pass filters. For push-pull output stages, there is an elegant workaround: since the output transformer constitutes the difference of the two anti-phase signals, all common-mode signals cancel each other out (as it happens in every differential amplifier). The guitar signal is fed out-of-phase into the two halves of the push-pull stage while the LFO-signal is fed in-phase to the two sections. The result is that the guitar signal is doubled while the spurious LFO-signal is cancelled.

The **control-grid voltages**♣ of the power tubes offer themselves as the “last possibility” to achieve the mentioned shifting of the operating point; it is implemented e.g. in the Tremolux 5G9). Synchronously pushing both grid voltages into the negative makes both tubes block: the audio signal is attenuated. Apparently, this power-tube control was seen as superior. It is found in several Fender amplifiers, and it superseded the **driver-stage control** (e.g. Tremolux 5E9-A) introduced a few years before and feeding the LFO-signal to the cathode of the phase-inverter. In both circuits an in-phase excitation of a differential amplifier is accomplished which (ideally) will avoid any LFO-signal coming out of the loudspeaker.

In the Fender Vibro-Champ (AA764), this LFO-compensation does not work because it has a single-ended power amp. Here, the LFO-signal is fed to the **cathode of the driver-tube**, and it is amplified together with the guitar signal, resulting in a low-frequency interference. The high-pass inserted directly ahead of the power-tube provides merely limited relief.

In contrast to the amplitude-modulator described above, the AM/FM-circuit first included in 1959 into the **Vibrasonic** is not understood prima facie. Here, the guitar signal is fed to a **frequency crossover** and separated into a high-pass branch and a low-pass branch♥. The effect is mainly a change in the loudness of the partials, but to a small degree there is also a change in phase, and therefore in pitch. The momentary angular frequency is, in fact, the derivative of the phase angle φ [3]. Since a 1st-order high-pass changes the phase by up to 90°, and a 1st-order low-pass does this by up to -90°, phase-shifts occur – as we change from the high-pass filtering to low-pass filtering – of up to about 120° (in the Fender-typical circuit). A pitch modulation with a frequency-swing of about ±10 Hz is possible with this approach, allowing for definitely audible changes in pitch. The threshold for just noticeable frequency changes is about ±2 Hz for FM-tones [12]. In **Fig. 10.8.14**, we see the magnitude and phase characteristics of the Vibrasonic-circuit (5G13); the schematic is given in **Fig. 10.8.15**. In later amplifiers (e.g. 6G13-A), the resistive voltage divider in the high-pass branch was dropped, with a gain of 7 dB in this branch.

♣ In principle, the screen-grid voltages of the power-tubes could be modulated, as well, but this would require a higher control-power.

♥ Strictly speaking: high-pass and bandpass, but the bandpass center frequency is, at 60 Hz, very low.

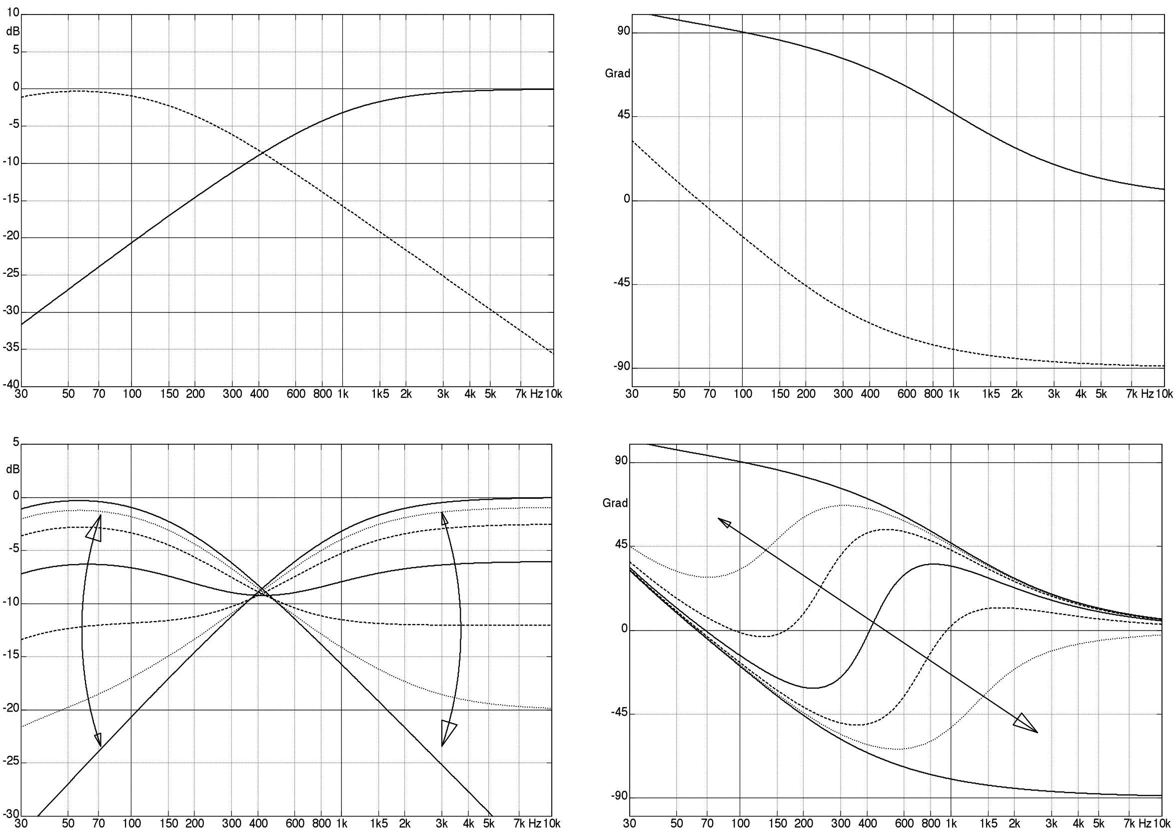

Fig. 10.8.14: Magnitude- and phase-characteristics of the frequency crossover (top), and of the overall system (bottom). 5G13.

This small **frequency modulation** realized in the Vibrasonic et al. was perfected in the **VOX** AC-30 (**Fig. 10.8.15**) using an **all-pass** circuit from the Wurlitzer organ [Petersen/Denney] that generates mainly FM but almost no AM. The required filter network is considerable: it uses 6 capacitors, 6 resistors and 3 amplifiers. It may nevertheless be divided into simple partial systems for a calculation purposes. The schematic shows two active 2[nd]-order bandpass filters of the same structure, differing merely in the values of the components. The signal mapping from U_0 to U_a is easily understood by omitting R_3 and C_3, to start with. What remains is a capacitively bridged voltage-divider determined by 4 components (4 degrees of freedom). One of the latter is the impedance level which tube-typically is chosen to be in the 100-kΩ-range. The second degree of freedom is the attenuation factor (about 3). Pole/zero-compensation yields the third degree of freedom ($R_1C_1 = R_2C_2$), and the cutoff frequency (about 1 kHz) yields the fourth. The result is a passive system of zero (!) order that generates a frequency-independent attenuation (of about 10 dB) across the whole frequency range.

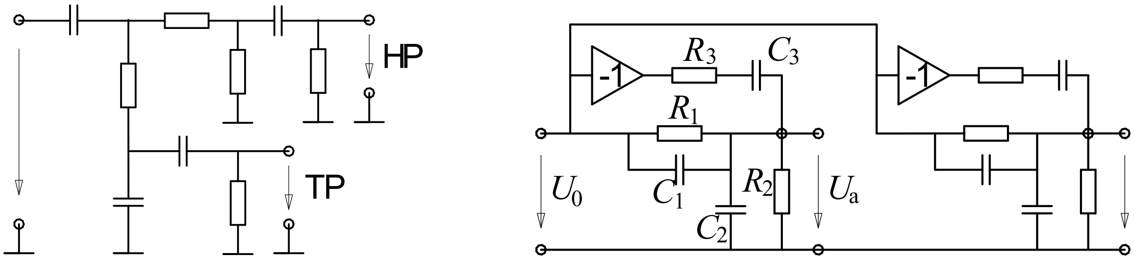

Fig. 10.8.15: Frequency crossover of the Vibrasonic 5G13 (Fender, left), and of the AC-30 (VOX, right).

Now we add R_3 and C_3, driven by an inverter ($v = -1$). Given the correct dimensioning, the output signal (U_a) is of the same phase at low and high frequencies. The active branch (R_3, C_3) cannot have an effect at low frequencies since C_3 is of high impedance relative to the divider-resistors. At high frequencies, no effect is there, either, because here R_3 is of high impedance relative to the divider-capacitors. It is only in the range of the cutoff frequency that R_3 and C_3 determine the transmission and have the effect of a phase shift from 0 to π. The components of the active branch (R_3 and C_3) can be determined such that the magnitude of the transfer characteristic becomes frequency-independent: this corresponds to a true all-pass.

An **all-pass** is a filter that changes only the signal phase but not the signal amplitude. This is achieved if the numerator of the transfer function is the complex-conjugate of the denominator of the transfer function; the magnitudes of numerator and denominator are equal for this condition, the magnitude of the transfer function becomes a constant (i.e. it is not dependent on ω). In the case of the filter circuit described above (Fig. 10.8.15), we get a 2nd-order all-pass the numerator- and denominator-polynomial of which contains p at the most with the power of two.

$$\underline{H}(p) = \frac{ap^2 + bp + 1}{cp^2 + dp + 1} \cdot H_0 \qquad a = c, \;\; b = -d.$$

Second-order all-pass, $p = j\omega$

A 2nd-order transfer-function has 5 degrees of freedom. Two of these are required by the all-pass characteristic: the same behavior for $f \to 0$ and $f \to \infty$ results in $a = c$, and the complex conjugation of numerator and denominator yields $b = -d$. The remaining 3 degrees of freedom are defined by: basic gain (H_0), cutoff frequency (a) and Q-factor (b). The components of the AC-30-filter in the original circuit were chosen such that not a perfect all-pass resulted but a slight magnitude change did also occur (about 3 dB). The reason for this is unknown; possibly the additionally generated AM was desirable.

An all-pass in itself does, however, still not generate a frequency modulation (FM) – it only creates a stationary (time-invariant) phase shift. For this reason there is a second all-pass (Fig. 10.8.15) with a cutoff frequency of a factor of 4,5 lower than the cutoff frequency of the first all-pass (1040 Hz vs. 4700 Hz). There will be a significant phase difference between the output signals of these two all-passes that can be turned into a time-variant phase-shift by a LFO-controlled cross-fading between the two outputs. If we take the phase modulation to be approximately sine-shaped, the maximum of the frequency modulation generated this way corresponds to the product of modulation-frequency (LFO-frequency) and phase-change amplitude: $\Delta f = f_{\text{mod}} \cdot \hat{\varphi}$. With f_{mod} = 10 Hz and a maximum phase-change amplitude of 55° (= 0.3π), we obtain a frequency-change amplitude of Δf = ±9.4 Hz.

We know from psycho-acoustical experiments that the threshold for just noticeable frequency differences is about ±2 Hz at low frequencies; the **auditory system** becomes increasingly less sensitive to absolute frequency changes only at frequencies above 500 Hz [12]. The frequency modulation generated by the AC-30-Modulator is therefore clearly audible; in addition we need to consider that the modulator circuit, and loudspeaker- as well as room-resonances, additionally generate amplitude modulation. In conclusion, it should be noted that in the AC-30, one of the two all-passes can switched-off such that the amplitude modulation becomes the dominating effect.

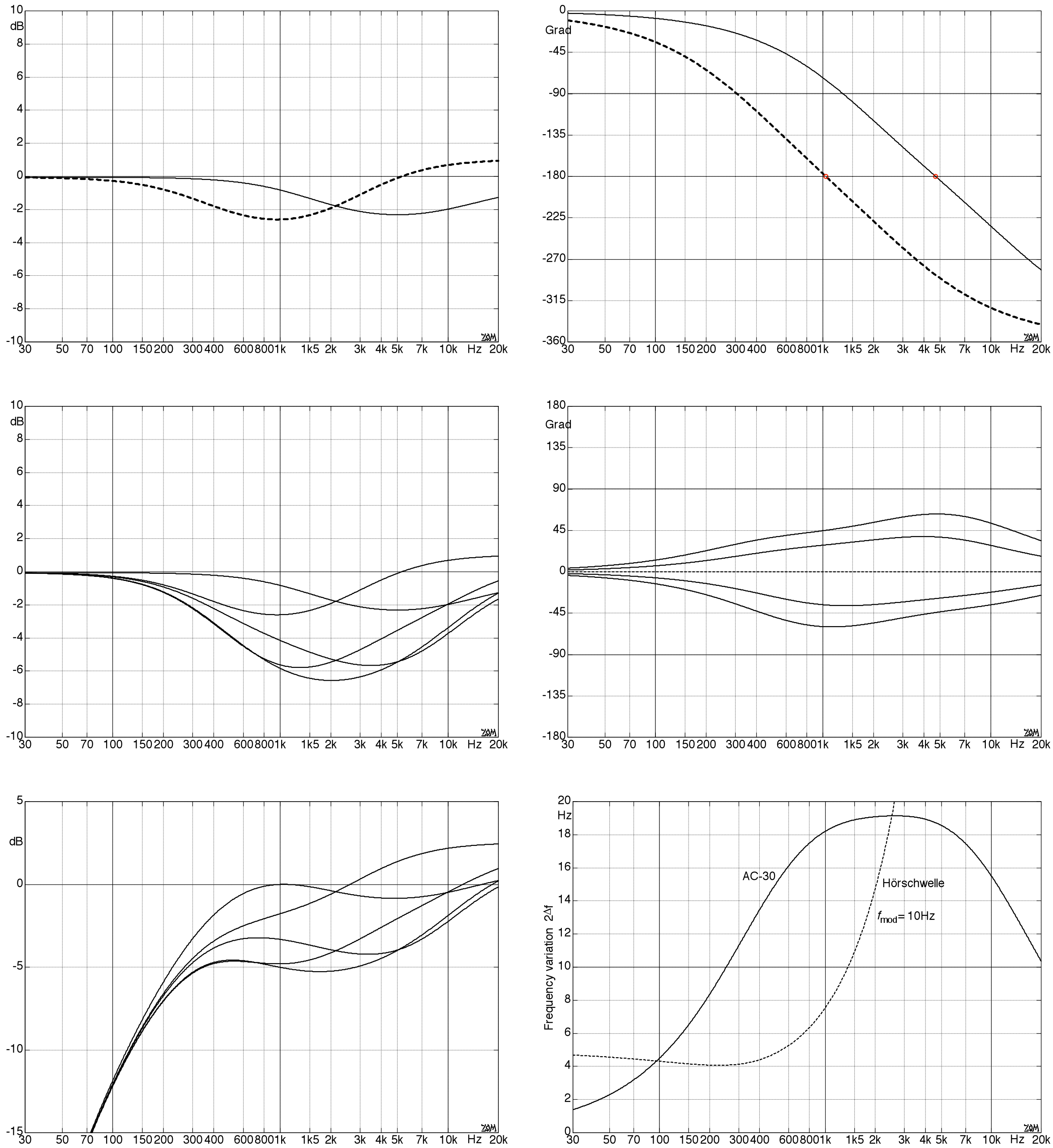

Fig. 10.8.16: AC-30-modulator. Top: magnitude and phase characteristic of both all-pass filters.
Middle: Magnitude- and normalized phase characteristic of the overall modulator.
Bottom: Magnitude characteristic incl. 4th-order high-pass; frequency change $2{\cdot}\Delta f$ achievable with f_{LFO} = 10Hz.
"Hörschwelle f_{mod}=10 Hz": threshold for just noticeable FM at a modulation frequency of 10 Hz

In **Fig. 10.8.16**, calculations regarding the transfer behavior are depicted. The LFO-controlled crossover between the all-passes generates level changes of up to 5 dB, and phase changes of up to 110°, resulting in frequency changes of up to 19 Hz at 10 Hz modulation frequency. In the lower right picture, the FM-perception-threshold is shown (dashed) for comparison [12]; the achieved modulation is clearly above threshold. The 4th-order high-pass added in for the picture on the lower left follows the modulator in the AC-30 to detach the remaining LFO-signal from the guitar signal. A compensation of the LFO-signal is implemented in the summation stage but this can never be perfect due to unavoidable tube tolerances.

