SEEBURG acoustic line

active systempanel 2

owner´s manual

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1 Introduction

Your new acoustic line SP2 is a very flexible device. Due to its wide range of adjustable parameters it can also conveniently be used as an add-on for existing PA systems.

Read this owner's manual carefully before putting your SP2 into operation, please!

The SP2 system controller becomes the central device of your sound reinforcement system as soon as the signal leaves the mixing console (see **fig. 1**). This one piece of audio gear incorporates a 2-way frequency dividing network, a dynamics controller and a four band EQ (optional) - all in stereo and fully configurable - as well as a connection panel and still takes up only one rack space.

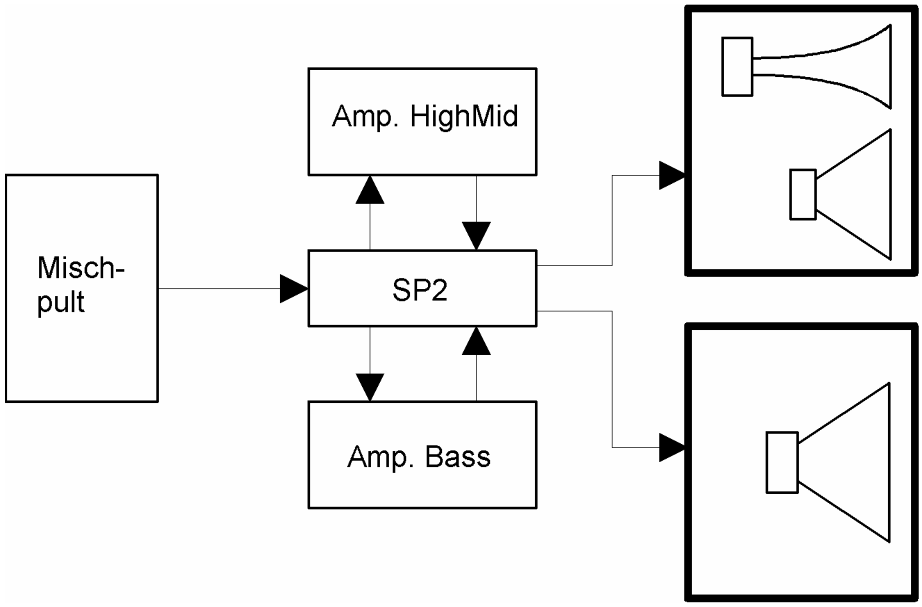

Fig. 1 How the SP2 is hooked up (only one channel pictured).

A detailed illustration of the SP2's functional blocks, which can be seen in the following figure (**fig. 2**).

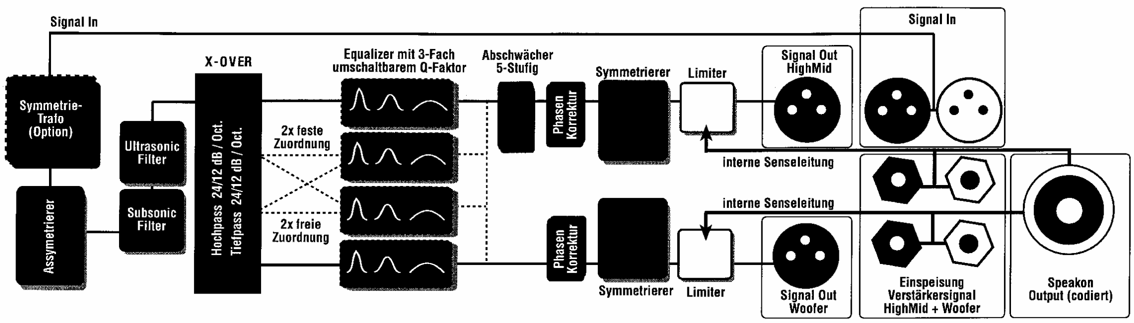

Fig. 2 Block diagram of the SP2

If you bought the SP2 together with or as an upgrade for an acoustic line PA system it is already set up for optimum performance with this specific system. For use with all other brand names there is a modifiable standard configuration which is described in appendix 4.3.

In order to change the SP2's parameters it is necessary to open the device and exchange one or more SIP resistor arrays and/or change some jumper positions. If you are not sure how to perform these tasks properly, please talk to your SEEBURG acoustic line dealer first.

Caution: Before opening the device must be turned off and the power line must be disconnected! The manufacturer will accept no responsibility whatsoever for any damage caused by improper handling of this device.

1.1 How to use this manual

This owners manual is written to assist you in setting the SP2 up in a way that lets you get the most out of it. To this end we think it is useful to give a concise description of the SP2's general mode of operation. However, we are not trying to write a textbook on audio engineering where everything can be covered in detail. Therefore some of the terminology used in the text is listed in a glossary found in the appendix. If you should still encounter any trouble in finding a setup to suit your needs, please talk to your dealer.

2 The controls and connectors

On the front panel (see **fig. 3**) you find the following connectors and indicators:

- **audio in** channels A and B: The SP2's input channels A and B, which are connected to a mixer's stereo outputs.
- **link to next panel** channels A and B: Feed-through of the input signals, which is provided for connecting another SP2 or a power amp.
- **Power on-LED:** Indicator lights up when ac power is present.
- **Signal A/B-LED:** Indicators light up when the input signal level rises above -20dB.
- **Limit-LEDs Woofer/HighMid:** Indicators show when the limiter is in operation.
- **output to speakers:** Speakon jacks to connect the speaker system.The highmid signal is present on pins 1+/-, the bass signal on pins 2+/-.

The back panel's connectors (see **fig.4**) are as follows

- **Signal to Amp jacks for HighMid A/B and Woofer A/B**: Jacks to connect your power amp inputs. These outputs deliver the wet SP2 signal.
- **Power Connection from Amp**: Binding post pairs which are connected to the power amp outputs. **Caution: Never connect or disconnect any power amp output while the power amp is in operation! Always turn off the power amp first!**

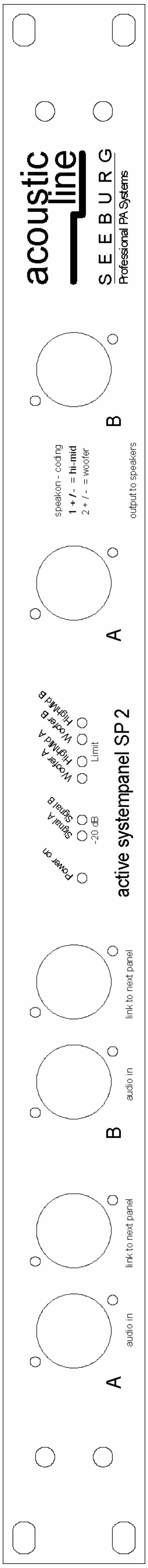

Fig. 3 Front panel facilities

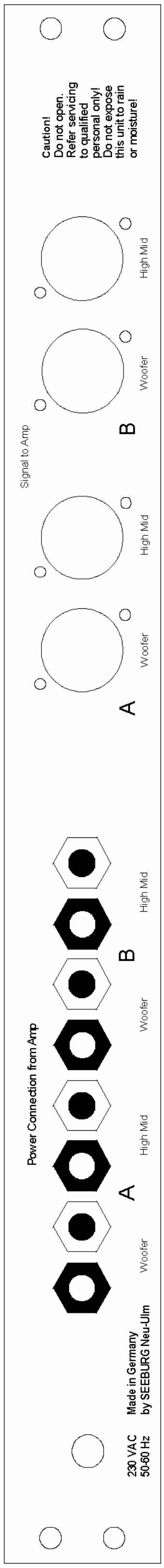

Fig. 4 Rear panel facilities

All SP2 inputs and outputs are electronically balanced. A balancing transformer can be installed as an option if DC isolation is a requirement. If you wish to have this modification installed in your SP2 please contact your authorized SEEBURG acoustic line dealer.

3 Operating the SP2

3.1 Active frequency dividing network

An active frequency dividing network is placed in between a mixing console and a power amp. It serves to divide the source signal into several frequency ranges. These signals are sent to seperate power amps with their respective woofers, midrange drivers and tweeters connected. The SP2 incorporates a 2-way network with the ability to adjust its crossover frequency via exchangeable SIP-resistor-arrays. The highpass filter for the woofer frequency range and the lowpass filter for the high-mid range both show a Linkwitz-Riley frequency and phase response.

The following parameters can be adjusted:

3.1.1 3 dB point

The characteristic breakpoint of the high- and lowpass filters' frequency response can be adjusted by insertion of a SIP-resistor-array. Choosing different 3dB-points for the two filters results in a varying amount of overlap of the complementary frequency response curves. Please choose your SIP values according to **table 1** in the appendix.

3.1.2 Filter falloff

In addition to the filters' 3dB points you can also adjust the steepness of the filter falloff in their respective passband-to-stopband-transition ranges. It can be chosen separately for each filter to yield either 12dB/octave or 24 dB/octave. The adjustment is performed by moving the blue jumpers on the PC board. In the highmid range you can also set this jumper to 0 dB/octave, effectively putting this filter out of operation.

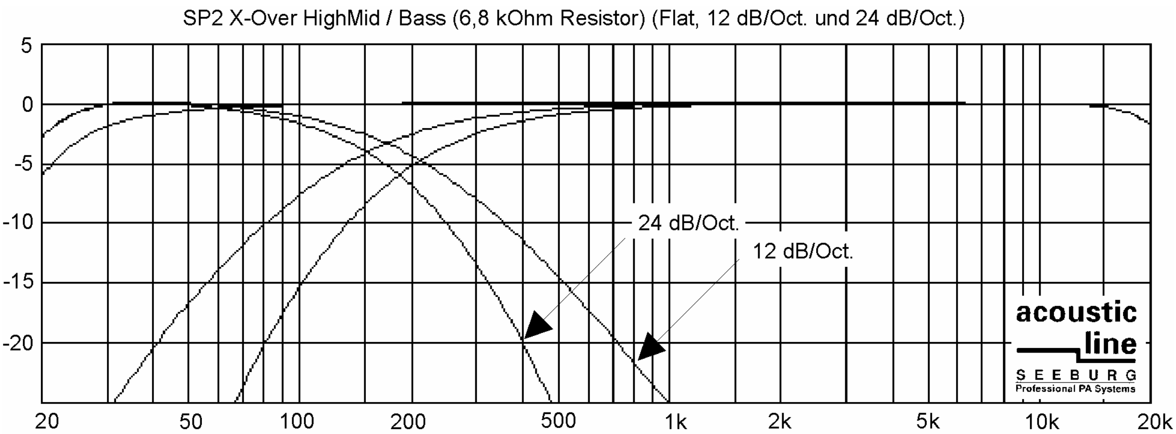

Fig. 5 Measured frequency responses of the SP2 frequency dividing network.

3.1.3 Phase correction

To compensate for phase differences between the frequency divided signals, the SP2 gives the opportunity to set the woofer phase angle to 0°, 60 ° or 120° via the light green jumper. Together with the highmid phase angle, which can be set to 0° or 180°, it is possible to adjust the relative phase in 60° steps over a full 360° range.

This phase correction may be necessary in order to compensate for speaker arrangements which show a horizontal displacement of the loudspeakers' acoustical centers (see **fig. 6**). The geometric difference in the acoustical signal path translates into a phase difference of frequencies which are transmitted by both speakers, leading to severe frequency response distortions in the crossover frequency range [1]

[1] see glossary in the appendix

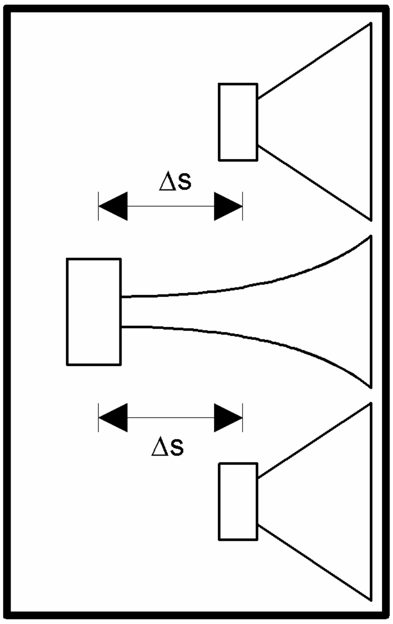

Fig. 6 Displaced Speakers

The phase correction angle (Φ in degrees) for a known speaker displacement is given by

$$\Phi = \frac{\Delta s \cdot 360° \cdot f}{c}$$

Δs: Speaker offset in m

c: Speed of sound in air. (Given an air pressure of 101.3 hPa and a temperature of 20° Celsius c is 340 m/s. The temperature coeff icient is 0.6 (m/s)/°C).

Example: You want to control a monitor with two 12''speakers and a 2''driver with the SP2. The 2'' driver's acoustical center is 0.115m behind the 12''speaker's acoustical center. The crossover frequency is 1500 Hz. The necessary phase correction then amounts to

$$\Phi = \frac{0{,}115m \cdot 360° \cdot 1500\,\frac{1}{s}}{340\,\frac{m}{s}} = 182{,}64° \approx 180°$$

3.1.4 Highmid attenuator

The attenuator function makes it possible to reduce the highmid gain relative to the woofer gain in steps of 2.5 dB. This feature comes in handy if your bass speakers are less efficient than your highmid system. The adjustment is performed by moving the red jumper to the respective position. The attenuation is printed on the PC board (see **fig. 7**). The first position on the socket corresponds to 0 dB.

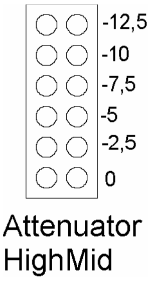

Fig. 7 Highmid attenuation jumper socket.

3.1.5 MONO-Bass option

If you intend to use a system with a single bass speaker cabinet, use the SP2's MONO-Bass function. To this end you have to latch the two blue wire breakers which are labeled 'Mono-Bass'. The effect is an identical bass signal at the woofer output connectors.

3.2 Limiter operation

The SP2 incorporates 4 independent limiters (HighMid CH. A and CH. B, Bass CH. A and CH. B). For each of these you can preset one of 3 different attack/release time constants as well as the maximum allowable speaker output power.

For proper limiter operation it is mandatory to connect the speakers to the 'output to speaker'-speakon connectors, and not directly to the power amp outputs. Those must be connected to the 'power connection from amp'-inputs. By passing the power amp signal through the unit it is possible to detect the actual power delivered to the speakers. This power detection is indispensable for proper limiter operation (see also **fig. 1**).

The time constants are set by placement of the blue jumpers in the limiter section. The jumper sockets are labeled A, B and C corresponding to fast, medium and slow attack/release times. The actual time constants are different for the two signal paths:

Bass:	A = 20ms,	B = 100ms,	C = 200ms
HighMid:	A = 10ms,	B = 40 ms,	C = 100ms

The maximum allowable power is adjusted via the four 5-switch DIP-arrays. **Table 2** in the appendix shows the power amp thresholds for various speaker cabinet impedances. Limiter operation is visible on the front panel by two red LEDs. The limiting characteristic of the SP2 is shown in **fig. 8**: the diagram shows a dynamic characteristic curve where the signal is not abruptly limited above a certain threshold, but rather compressed with a compression ratio[2] that increases with increasing power.

[2] see glossary in the appendix

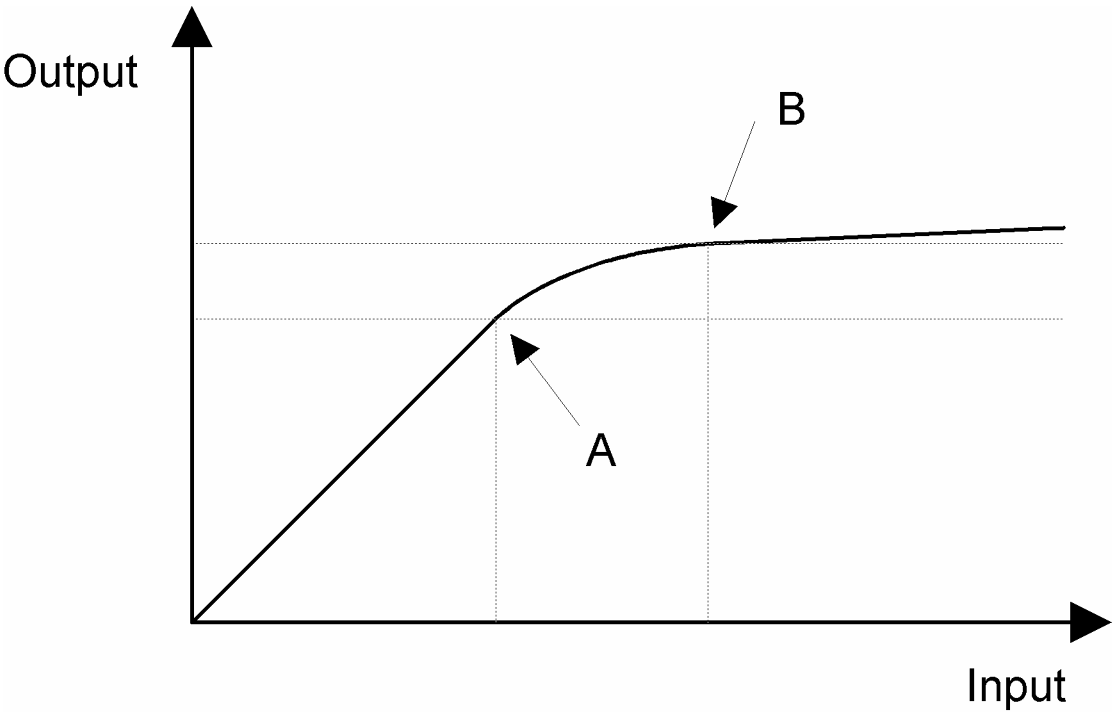

Fig. 8 Characteristic curve of the SP2 limiter. The limit indicator LED lights up at point A. The compression ratio increases with an increasing input signal (area between A and B). The maximum compression ratio of 1:20 is reached beyond B.

3.3 Subsonic and ultrasonic filters

The subsonic filter is a highpass filter with a falloff of 18 dB/oct. The ultrasonic filter shows the same steepness with a lowpass characteristic. The 3 dB points are at 25 Hz and 18 kHz, respectively. The filters are used to block frequencies below and beyond the normal hearing range. They are always in operation and cannot be turned off.

3.4 Optional equalizer PC card

If you choose to purchase a SP2 without the 2x4 band equalizer card, you still have the option to install one as an upgrade. With this card it is possible to equalize your system's frequency response at adjustable center frequencies with a free choice of the signal path (woofer or highmid) where you want to attach each EQ band to. The only limitation here is a maximum of 3 (out of 4 available) EQ bands per signal path. The insert points for the individual filter sections are labeled 'EQ-Insert' on the PC-motherboard. Unused insert points must be short circuited with a jumper. (These yellow jumpers are placed in 'park sockets', which can be found right next to the EQ insert connectors.)

3.4.1 Adjustment of the EQ center frequencies

The available EQ bands are divided into several frequency ranges (see **fig. 9** through **12**). Out of these ranges the exact center frequencies are chosen by installing SIP resistor arrays. The resistor values and the corresponding center frequencies are listed in **table 1** in the appendix for the E12 resistor series, which has been expanded by a 20 kΩ value from the E24 series. The SIP insert connectors are labeled 'Frequency' on the EQ card.

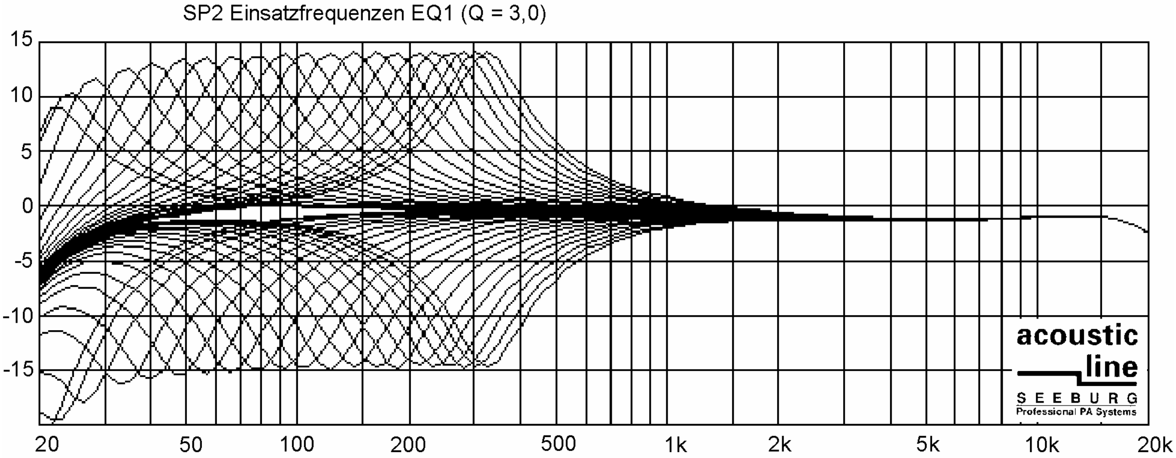

Fig. 9 Measured frequency response curves for EQ1

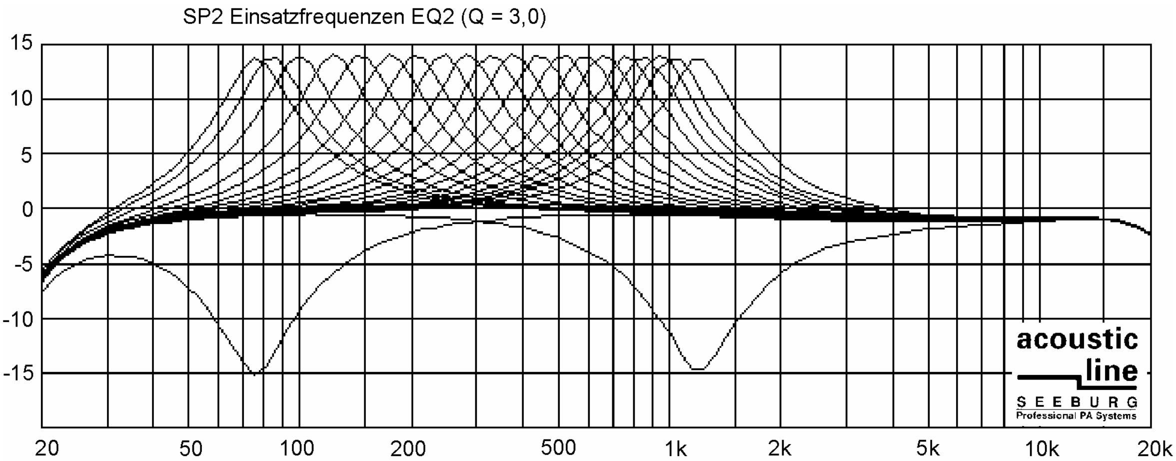

Fig. 10 Measured frequency response curves for EQ2

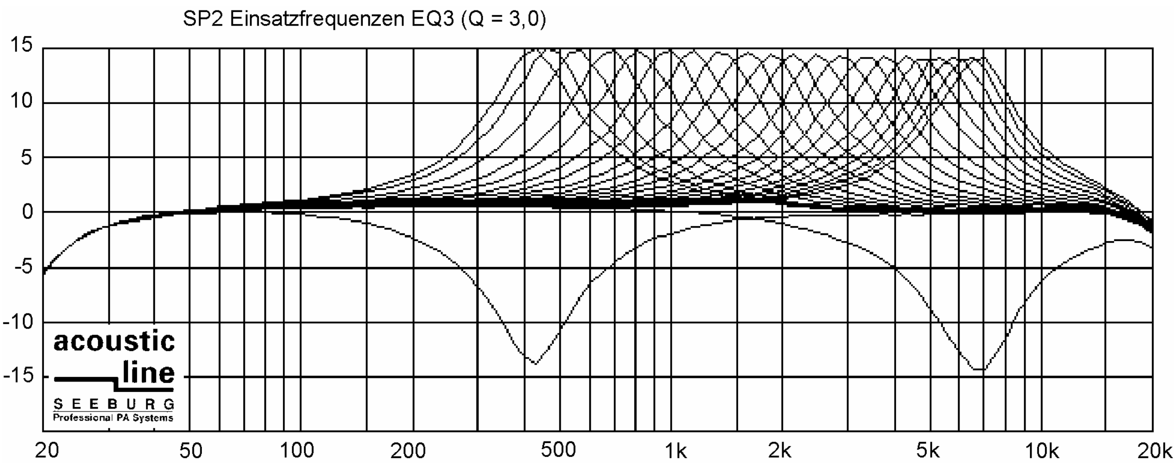

Fig. 11 Measured frequency response curves for EQ3

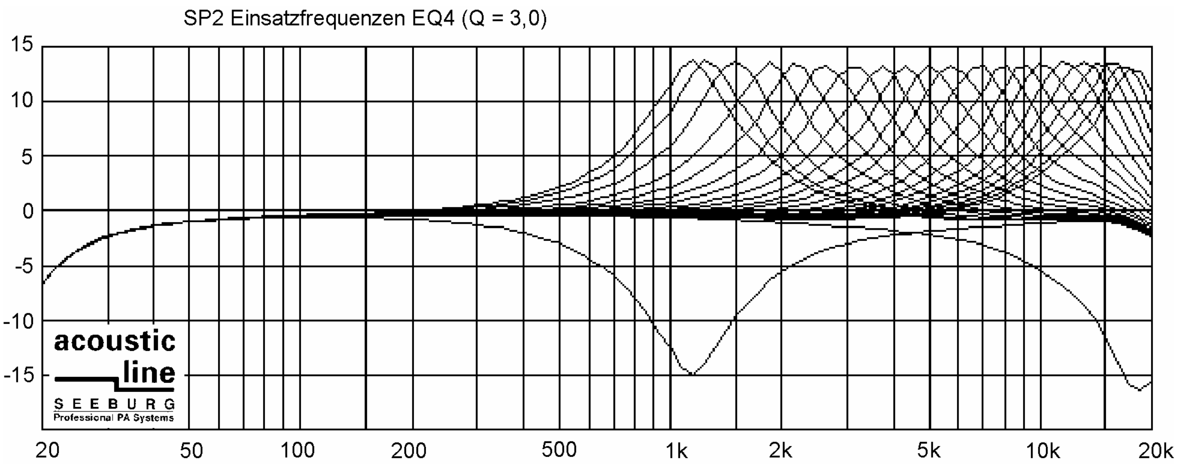

Fig. 12 Measured frequency response curves for EQ4

As can be seen in **fig. 9** through **fig. 12,** there is a falloff of the zero amplitude curves below 50 Hz and above 15 kHz. This is due to the subsonic/ultrasonic filters, which are always in operation (see section 3.3).

3.4.2 Adjustment of the filter Q

The quality factor of the filters (filter Q[3]) can be set by positioning the yellow jumpers on the EQ PC board. The corresponding connectors are labeled 'Bandwidth'. There is a choice of three filter Qs:

- **Peak (Q = 12):** yellow jumper left
- **Normal (Q = 3):** yellow jumper center
- **Shelve (Q = 1,5):** yellow jumper right

The resulting characteristic curves are shown in **fig. 13**.

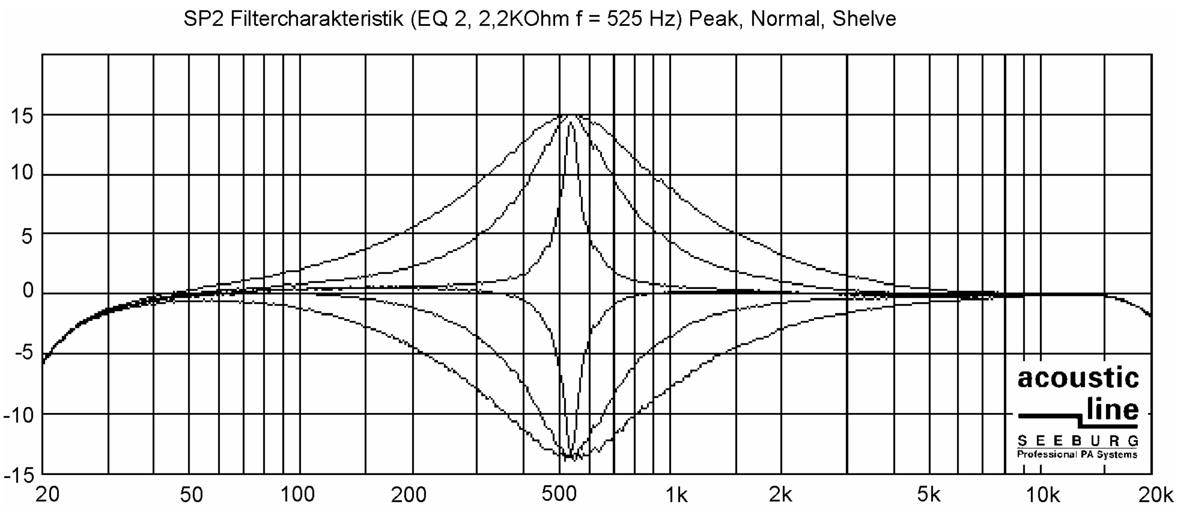

Fig. 13 Characteristic curves of the EQ filters with three different Q factors preset.

3.4.3 Filter gain

The filter gain for each EQ band is adjusted with the trimmers on the EQ card from -12 dB (maximum attenuation) to +12dB (see **fig. 9** through **fig. 12**).

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[3] see glossary in the appendix

4 Appendix

4.1 Tables

SIP-resistor-array	X-Over	EQ1	Frequency EQ2	EQ3	EQ4
470	2400	330	1250	7000	19000
560	2100	305	1100	6400	17500
680	1600	280	1000	6000	16000
820	1400	290	920	5700	15500
1K	1200	240	880	5000	13500
1,2K	1000	205	800	4250	11500
1,5K	800	185	700	3800	10500
1,8K	750	162	610	3200	9200
2,2K	530	140	525	2850	8000
2,7K	430	120	450	2500	6750
3,3K	370	105	400	2200	5850
3,9K	310	90	350	1800	5000
4,7K	260	76	300	1600	4250
5,6K	220	65	250	1400	3700
6,8K	170	55	210	1200	3100
8,2K	150	45	175	1000	2700
10K	130	38	150	825	2250
12K	105	32	125	700	1800
15K	85	28	100	565	1500
18K	70	25	85	460	1250
20K	65	20	75	430	1150

Table 1 SIP resistor values and the corresponding 3 dB points of the frequency dividing network and EQ center frequencies.

Resistor values given in ohms (Ω).

Frequencies given in hertz (Hz).

The resistor values are usually coded with three numbers on the SIP housing. The two most significant digits represent the resistor value. The third number determines the magnitude by defining the number of zeroes that must be attached to number one and two.

Examples: 222: 2200 Ω = 2,2 kΩ

153: 15000 Ω = 15 kΩ

Configuration					Threshold				
S 1	S 2	S 3	S 4	S 5	U	P 16	P 8	P 4	P 2
OFF	OFF	OFF	OFF	OFF	12	9	18	36	72
ON	OFF	OFF	OFF	OFF	15	14	28	56	113
OFF	**ON**	OFF	OFF	OFF	18	20	41	81	162
ON	**ON**	OFF	OFF	OFF	21	28	55	110	221
OFF	OFF	**ON**	OFF	OFF	24	36	72	144	288
ON	OFF	**ON**	OFF	OFF	28	49	98	196	392
OFF	**ON**	**ON**	OFF	OFF	32	64	128	256	512
ON	**ON**	**ON**	OFF	OFF	36	81	162	324	648
OFF	OFF	OFF	**ON**	OFF	40	100	200	400	800
ON	OFF	OFF	**ON**	OFF	44	121	242	484	968
ON	**ON**	OFF	**ON**	OFF	48	144	288	576	1152
OFF	OFF	**ON**	**ON**	OFF	52	169	338	676	1352
ON	OFF	**ON**	**ON**	OFF	56	196	392	784	1568
OFF	**ON**	**ON**	**ON**	OFF	60	225	450	900	1800
ON	**ON**	**ON**	**ON**	OFF	64	256	512	1024	2048
OFF	OFF	OFF	OFF	**ON**	68	289	578	1156	2312
ON	OFF	OFF	OFF	**ON**	72	324	648	1296	2592
OFF	**ON**	OFF	OFF	**ON**	76	361	722	1444	2888
ON	**ON**	OFF	OFF	**ON**	80	400	800	1600	3200
OFF	OFF	**ON**	OFF	**ON**	84	441	882	1764	3528
ON	OFF	**ON**	OFF	**ON**	88	484	968	1936	3872
OFF	**ON**	**ON**	OFF	**ON**	92	529	1058	2116	4232
ON	**ON**	**ON**	OFF	**ON**	96	576	1152	2304	4608
OFF	OFF	OFF	**ON**	**ON**	100	625	1250	2500	5000
ON	OFF	OFF	**ON**	**ON**	104	676	1352	2704	5408
OFF	**ON**	OFF	**ON**	**ON**	108	729	1458	2916	5832
ON	**ON**	OFF	**ON**	**ON**	112	784	1568	3136	6272
OFF	OFF	**ON**	**ON**	**ON**	116	841	1682	3364	6728
ON	OFF	**ON**	**ON**	**ON**	120	900	1800	3600	7200
OFF	**ON**	**ON**	**ON**	**ON**	124	961	1922	3844	7688
ON	**ON**	**ON**	**ON**	**ON**	128	1024	2048	4096	8192

Table 2 DIP-switch configuration for adjusting the limiter threshold.

U in volts
P 16 in watts delivered to a 16 Ω impedance
P 8 in watts delivered to a 8 Ω impedance
P 4 in watts delivered to a 4 Ω impedance
P 2 in watts delivered to a 2 Ω impedance

4.2 Note - chart for SP2 configurations

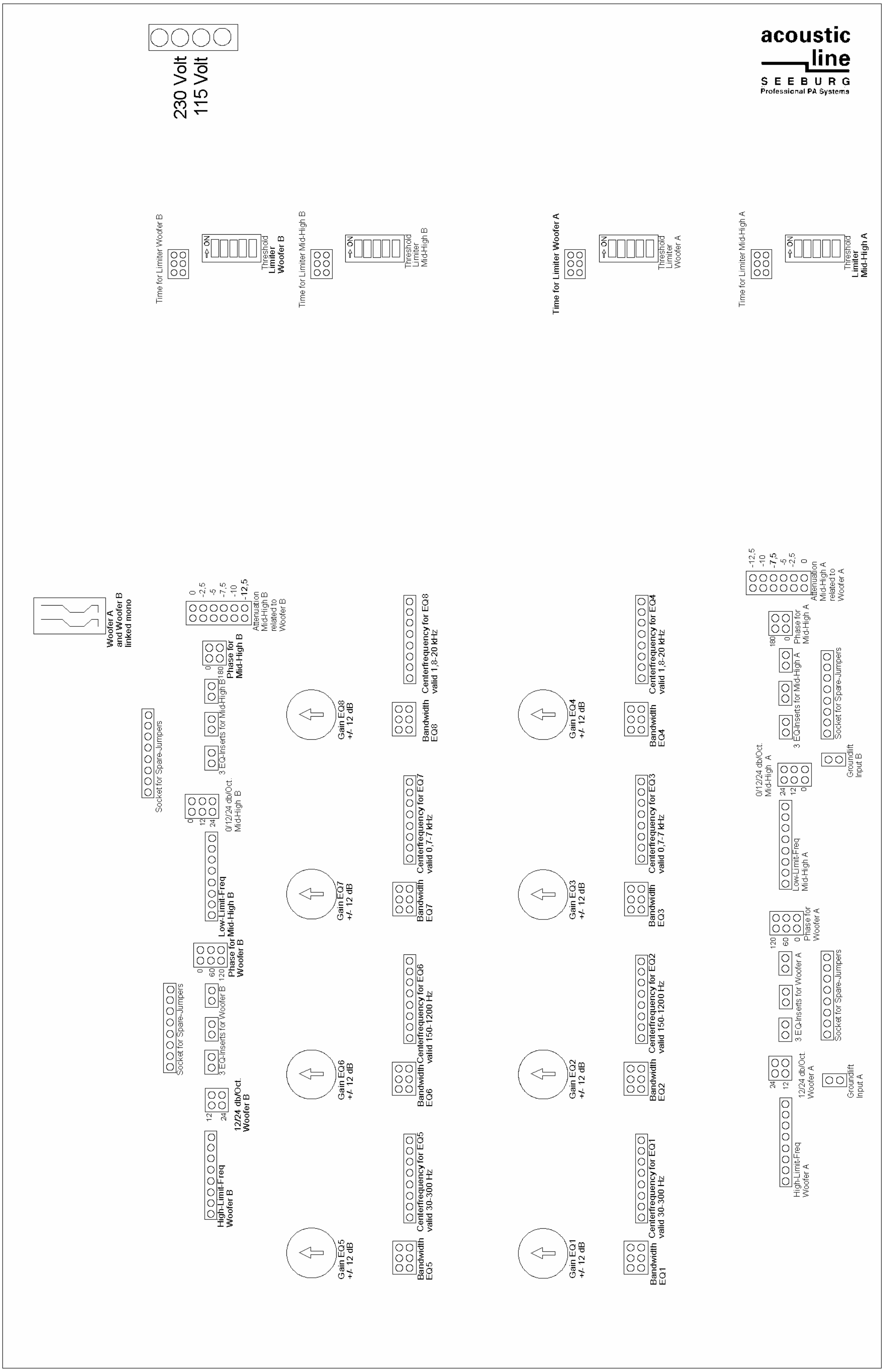


Fig. 14 Simplified SP2 layout can be used as a note chart for controller configurations.

4.3 Factory presets

SP2 units which have not been delivered as part of a SEEBURG acoustic line sytem are factory preset as follows:

X-over: 3 dB point (SIP):		105 Hz (12 kΩ)
	Filter falloff:	24 dB/oct.
	Phase angle HighMid and Woofer:	0°
	Attenuation HighMid:	0 dB
	Mono-Option:	off
EQ 1:	Center frequency (SIP):	55 Hz (6,8 kΩ)
	Insert:	Woofer
	Bandwidth:	normal (jumper centered)
	Gain:	0 dB (trimmer in center position)
EQ 2:	Center frequency (SIP):	210 Hz (6,8 kΩ)
	Insert:	HighMid
	Bandwidth:	normal (jumper centered)
	Gain:	0 dB (trimmer in center position)
EQ 3:	Center frequency (SIP):	3200 Hz (1,8 kΩ)
	Insert:	HighMid
	Bandwidth:	normal (jumper centered)
	Gain:	0 dB (trimmer centered)
EQ 4:	Center frequency (SIP):	13500 Hz (1 kΩ)
	Insert:	HighMid
	Bandwidth:	normal (jumper centered)
	Gain:	0 dB (trimmer centered)

Limiter: all limiter-DIP-switches on (limiter not in operation)

4.4 SP2 specifications

SNR (at 6 dB gain):	> 103 dB (EQ included)
THD (at 6 dB gain):	< 0,04 %
Headroom:	22 dBU
subsonic/ultrasonic filters:	-3 dB points at 25 Hz and18 kHz
	-18 dB/octave
Output impedance (Signal to Amp):	600 Ω
Input impedance (Audio In):	40 kΩ
AC mains / power draw:	230 Volt/AC/10 VA (EQ inc.)
Dimensions (w x h x d):	482 x 440 x230 mm
Weight:	3,4 kg

4.5 EQ card specifications

Gain:		±12 dB
Frequency bands	EQ1:	20 - 330 Hz
	EQ2:	75 - 1250 Hz
	EQ3:	430 - 7000 Hz
	EQ4:	1150 - 19000 Hz

(Specifications subject to change without notice)

4.6 Glossary

- **Compression ratio:** Compressors limit the audio signal amplitude whenever it surpasses a certain preset threshold level. The characteristic curve's slope equals 1 below this threshold, meaning that the output amplitude is the same as the input amplitude. At the threshold point the curve has a knee, beyond which the slope decreases with increasing compression ratio. Here, the output grows slower than the input (see **fig. 15**). The compression ratio is the slope of the curve above the compressor's threshold. Stand alone compressor units feature an adjustable compresion ratio, as opposed to a limiter, which usually shows a fixed compression ratio of 1:∞ (i.e. a slope of 0: the output amplitude is constant over varying input levels).

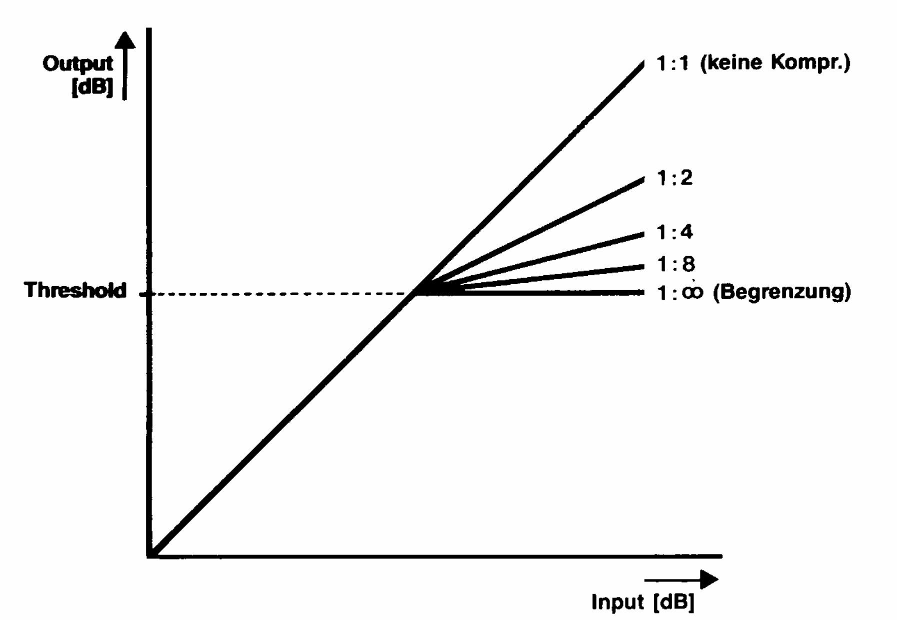


Fig. 15 Illustration to help explain the term compression ratio.

- **Decibel (dB):** In technical acoustics all kinds of signal amplitudes, power levels and sound pressure levels are usually expressed in decibels. Since a decibel is a logarithmic measure, it can only express the ratio of two signals, and no absolute values.

By definition, the ratio of two power levels is:

$$\frac{L_p}{dB} = 10 \cdot \log\left(\frac{P}{P_0}\right)$$

With the help of a known impedance, this can be expressed in voltages as follows:

$$\frac{L_u}{dB} = 20 \cdot \log\left(\frac{U}{U_0}\right)$$

The reference voltage U_0 in audio electronics is by convention 0.775V.

The reference power level P_0 is usually assumed to be 1 mW, which equals 0.775V into 600 Ω.

- **Phase shift:** The superposition of two identical waveforms of the same frequency and amplitude that cross zero at the same time (i.e. they are in phase) results in the same wave with its amplitude doubled (see **fig**. **16a**). Likewise, if they are 180° out of phase, the wav e is extinguished (see **fig. 16b**). The time delay between the two waves is called their phase shift. The effect of the amplitude amplification/attenuation depends strongly on the amount of phase shift. It is not necessary for the two signals to have the same amplitude, it is just that then the effect is at its extreme. But even lesser effects due to different amplitudes (at the same frequency) can be very noticable in audio reproduction.

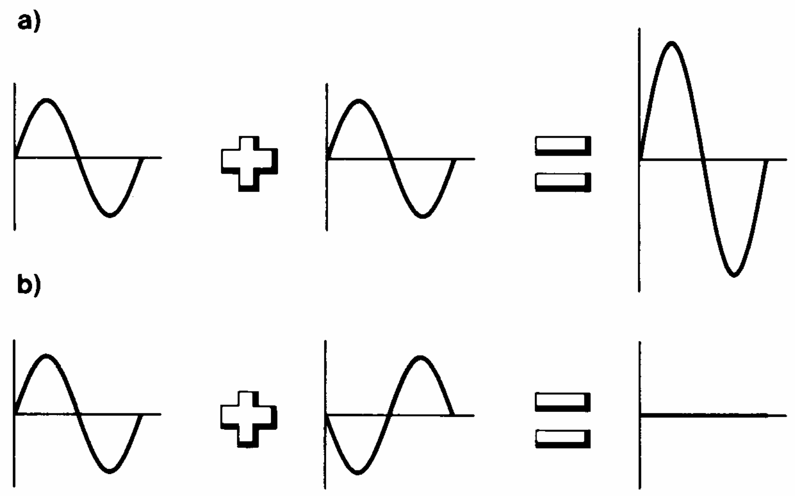


Fig. 16 Two waves of equal amplitude, frequency and phase add up to a wave of twice the amplitude (a).

- **Filter Q:** The quality factor (Q) of a resonant filter is a measure for the sharpness of the resonance peak that can be realized with this circuit. Every equalizer contains a number of electric resonant filters. A equalizer with a low Q will affect the frequencies in the vicinity of the resonance peak stronger than one with a high Q. So, if you want to control only a very narrow frequency band you have to choose a very high Q filter. However, in many applications it is desired to affect a wider range (for example a broad bass boost between 50 and 150 Hz in a PA system), which can be achieved with a low filter Q (see **fig. 17**).

The filter Q for electrical resonant circuits can be expressed as:

$$Q = \frac{f_0}{B}$$

B = fu-fl: Bandwidth. fu is the upper 3 dB point, fl is the lower 3 dB point.

fo: Center frequency.

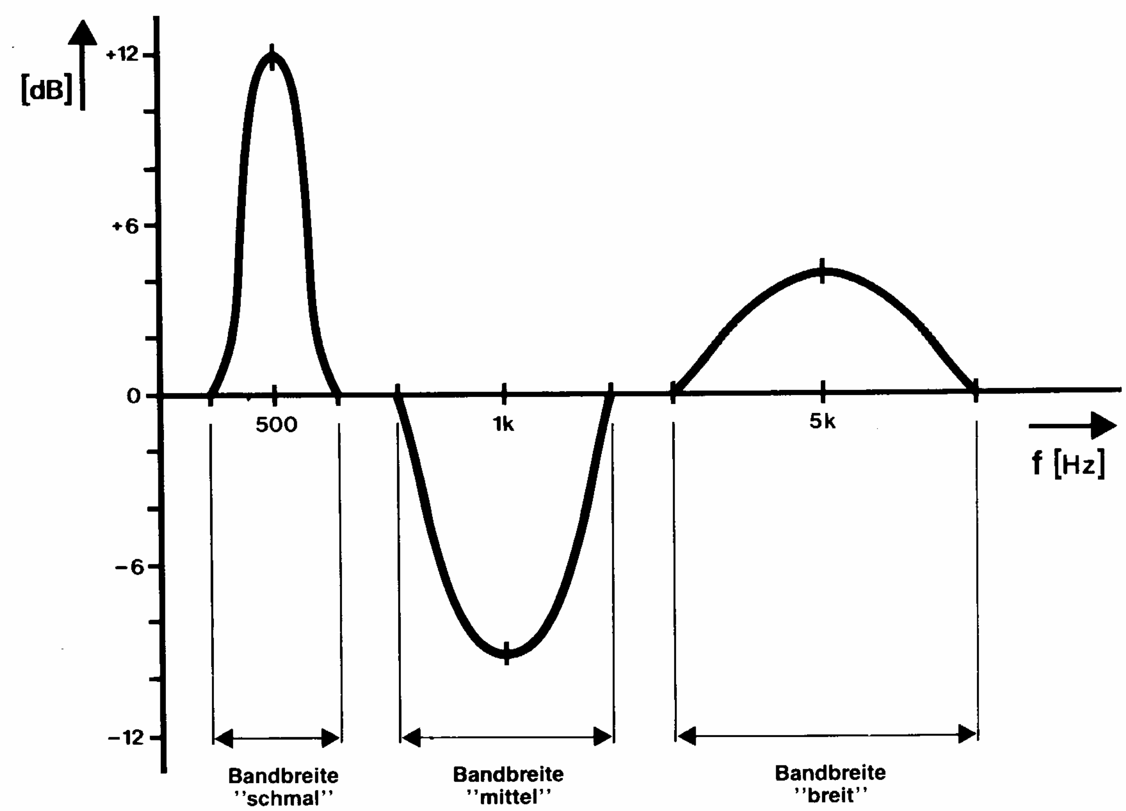
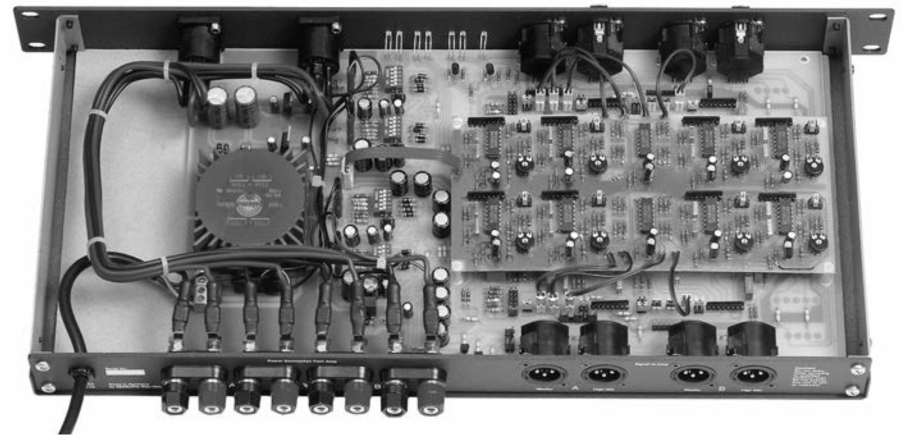


Fig. 17 Filter Q and Bandwidth.



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