BANG & OLUFSEN A/S
LABORATORY INFORMATION
DESIGN PRINCIPLES
BEOMASTER 4400.

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THE DESIGNER'S DILEMMA:

Balancing a number of choices

The design of a receiver is a highly specialised project. It places demands on at least two areas of specialisation: the amplifier, which is basically audio frequency electronics, and the tuner, which is radio frequency electronics. However, an appreciation and understanding of the design requirements of other elements in the sound reproducing system, particularly of loudspeakers and pickup cartridges, is essential both from a performance point of view (ultimately these components must work harmoniously together) and from a cost point of view.

Cost is an important factor for every designer. His aim ought to be to utilise the least expensive methods to achieve his goal – whether he is concerned with a "state of the art" or a budget design. In other words, the designer's aim must be to secure the best value for money within his design objectives. It should be remembered that apart from costs, circuit design is not usually the major problem – the real problem lies in specifying objectives.

What then of manufacturer's claims of "no compromise" products? The fact is that even if a designer is free to disregard the cost factor, he still cannot make a perfect receiver – if only because of the inherent limitations of the hardware he must use. Every material and likewise every component has its limitations. In turn, the designer's choice of materials and how he uses them determine the product's limitations.

Many audio manufacturers specialise in only one or two of the elements of a sound reproducing system and can, in time, develop a know-how, which enables them to design products, whose measured specification levels approach the limits of those obtainable with current technology. However, such specialisation often implies a less than perfect understanding of the other links so that excellent specifications are often achieved at the expense of compatibility. Thus, when such a component is placed in a system, sound quality is not as good as its specification level suggests.

Furthermore, three undeniable facts add to the designer's dilemma.

One, specifications can be traded off against each other – improving one at the expense of the other. For example, for any given set of components at any level of sophistication or price, wider frequency range or power bandwidth is usually obtained at the expense of signal to noise ratio or vice versa.

Two, specifications can be measured according to various standards. It may often be necessary for a designer to choose the standard for his measurement and consciously design the product or circuit to get the best paper specification according to that standard, even at the cost of subjective sound quality.

Three, despite the industry's increased knowledge about the correlation between objective and subjective evaluations, high measured specifications and good sound quality are not the same thing. This means that inspite of all that is said, there seems to be no absolute relationship or consistent correlation between conventional performance measurements or paper specifications and what actually sound good.

In other words, the design of an amplifier – indeed of any electronic component – is an exercise in balancing a number of choices.

With these in mind, it is the purpose of this booklet to acquaint the reader with some of the design considerations, which lie behind the development of the Beomaster 4400 receiver.

The Beomaster 4400 represents what may be said to be a new kind of reasoning, a new concept in the design of a modern high quality hi-fi receiver.

Internally, and especially at the Bang & Olufsen laboratories, this new kind of reasoning with all its implications was endearingly nicknamed "Project Tango".

In theory, Project Tango embodies the acknowledgement and acceptance of two limitations: those related to the quality of our programme sources – for example records, and those related to hardware. But Project Tango also embodies the acceptance, that measurements are only a tool. It is the ear, that is the ultimate judge of sound performance. This means that "listening" specifications are given higher priority.

In practice, Project Tango therefore represents Bang & Olufsen's efforts to come to terms with a series of compromises, to design products which not only measure well but also sound well – within a reasonable consumer price bracket for quality sound equipment.

Thus, Project Tango is a different way of establishing the design goals of hi-fi equipment for use in home entertainment. This thinking is expressed in the design of the Beomaster 4400. But in reality, Project Tango is nothing new at Bang & Olufsen. The Bang & Olufsen Beogram gramophones with their electronically controlled tangential arms, and the line of Beovox loudspeakers with virtually linear phase response and linear amplitude characteristics are also expressions of the same spirit or concept of approach.

Techniques – or performance?

Often, a manufacturer will claim, that his product is better than others because he uses a new component, method or technology. Or perhaps a reviewer will attribute poor performance to the use of a well known component, construction method or circuit. The wise consumer will be suspicious of such generalisations. In the majority of cases, good or poor performance is not due to components, but to the way in which they are used.

Sometimes a component becomes fashionable because a manufacturer develops it a huge expense for a costly prestige product, with excellent performance. Soon others make use of the prestige with an inexpensive imitation of that component, with performance no better or worse than the conventional. For the conscientious designer it is unfortunate, that components and circuits fall into disrepute because of the appearance of budget designs with poor performance, in which these components are used incorrectly, or below their performance capabilities.

It should be remembered, that while the laws af nature and physics apply for all designs, improvements that can be made to any circuit or component are limited only by the ingenuity of the designer.

It was a wise man, who said: "There are no simple solutions, only intelligent choices". Even so, perhaps both characterise Project Tango. On the following pages we discuss some of the most important choices made by the Design Team of the Beomaster 4400.

Do you really want a concert hall in your living room?

Once upon a time hi-fi manufacturers promised to transfer the concert hall to your living room. Today, we know better. Or some of us do, because the fact is, that tradition dies hard – many customers still want to cherish the illusion, and there are manufacturers enough to feed it. Or maybe the process is vice versa – but that is not the matter in discussion. The point is, that even with today's technology it is gross deceit to talt in terms of hi-fi equipment "bringing the concert hall into anyone's home".

If only from a psychological point of view we must accept that going to a concert – getting dressed, the travel, being in close proximity to perhaps hundreds of people, and the effect of seeing the musicians, in fact all the experiences involved with "being there" is a far cry from sitting in your own home listening to a record, tape or FM broadcast. It is not the job of a high fidelity set to interpret music or the quality of the sound, or in fact make the music more natural or exciting. That is the job of the musicians and recording engineer. The high fidelity set must on the other hand accurately reproduce the recording or broadcast in your home. In fairness, the serious hi-fi manufacturer must respect the reality of the programme source and with it accept to leave the concert hall just where it is.

Similarly, since the whole point of the exercise is listening pleasure it seems wise to include considerations about the human ear at least side by side with conventional measurements in any attempt to judge the quality of a sound reproducing system.

At this stage, a few readers might begin to ask: Is this a sell-out? – all this talk of respecting limitations . . . . . and with today's technology? No, there is no sell-out, despite advanced technology. In a way, it is because of technological potential, that Bang & Olufsen engineers and others stopped to ask certain questions. Technology permits us to construct amplifiers whose power output is say 2 x 1000 watts RMS. But what would anyone in the home entertainment sector do with such power? And even if that can be answered satisfactorily, a more relevant question is: "At what cost?" Cost, both in terms of economics and specification compromise. For example, is enormous power output secured at the expense of reliability? – For it is a well-known fact, that the greater the amount of thermal heat produced, the less reliable the components are likely to be. Besides, where would we get speakers to match the potential of the amplifiers, and if we construct them, would we have space in our living rooms or would we need to build a concert hall to accomodate them? Could our ears hear it, or would the 1000 watts remain an unused prestige symbol? In short, are excellent specifications in one area obtained at the expense of other specifications, others which are perhaps more crucial because the ear is more sensitive to them or because the rest of the sound system or the source cannot attain that standard?

Also, are excellent overall specifications achieved, but at a monetary cost which is prohibitive for avid enthusiasts?

These are some of the deliberations, which preceded the development of the Beomaster 4400. Enthusiasm as well as scientific discipline characterised the development of the project.

Defining the problem

The project was divided into four sections, each of which had a target formulation to enable all the sections to be matched, but where each was optimised within its own design objectives. These are:

Input amplifiers
Tone controls and switching
FM tuner
Power amplifier

The sections were, from the beginning, integrated into a common idea, in which importance was given to the following main points:

Signal path through receiver of a quality compatible with the quality of the source.
Signal paths with handling capacity so that first limitation is output stage.
Maximum possible reliability.
Flexibility in use where necessary, compatible with simplicity and ease of operation.
Production efficiency and ease of service to ensure minimum costs.

The development objectives of the Beomaster 4400 have been largely achieved. We are confident, that the Beomaster 4400 represents a product in which every specification – including price – is optimised to give the best possible sound quality performance. It is a product, which will faithfully recreate the original of the source, a product, which truly deserves that too-often-abused descriptive term – high fidelity.

The Beomaster 4400 is not the ultimate in receivers. We do not claim that. We simply claim, that given today's parameters (discussed in previous pages) it is an optimum solution, which cannot be improved upon without a significant increase in cost and a corresponding increase in the sound quality of our music sources.

In the interest of clarity, each system will be described separately.

THE GRAMOPHONE PREAMPLIFIER

The gramophone input on an amplifier differs from all other inputs because the signal level generated by a magnetic cartridge is very much lower than that obtained from any other sound source. This very weak signal has to be amplified to the same level as the other inputs before it is fed, through the control stages, to the power amplifier. The preamplifier's sole job is to amplify or enlarge the signal and add the necessary equalisation without adding to or subtracting from it. Several factors are important in this process, among them noise or hum is the most crucial.

Dynamic range and noise.

Every amplifying circuit has some noise associated with it, and noise is of particular importance in circuitry designed to handle low level signals. The gramophone preamp has to be able to handle a fairly wide range of voltages in order to cope with the different modulation levels from records, as well as the varying outputs from different makes of cartridges. The difference between the highest level an amplifier can handle, and the noise level is called its dynamic range, quoted in dB.

The problem is that the higher the level the preamplifier is required to handle (without altering basic sensitivity for full power output), the more noise it will generate. Irrespective of the cost or sophistication of the circuitry, the designer is therefore forced to make a choice between the highest signal level the preamplifier is required to handle and the maximum level of noise he will accept from the circuit.

Thermal noise.

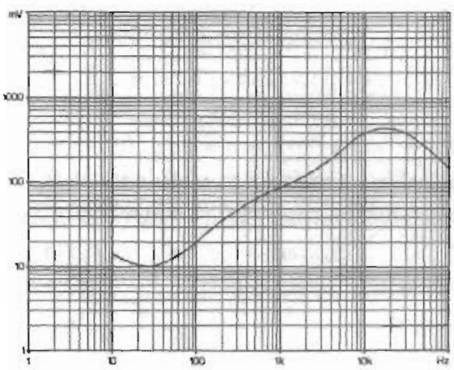
Every magnetic cartridge has an associated impedance. This is a function of the coils and magnetic material of which they are made. Every impedance has associated with it source, or thermal noise which cannot be removed. The lower the impedance, the lower the thermal noise. Thus, thermal noise (which is inherent in all magnetic cartridges) sets a lower limit on the noise level at the output of the preamplifier.

In order to ensure minimum thermal noise, design engineers ensure that the impedance of Bang & Olufsen cartridges is very low. Indeed, among high quality magnetic cartridges of normal sensitivity, Bang & Olufsen cartridges are among those with the lowest impedance.

Phono Overload Voltage.

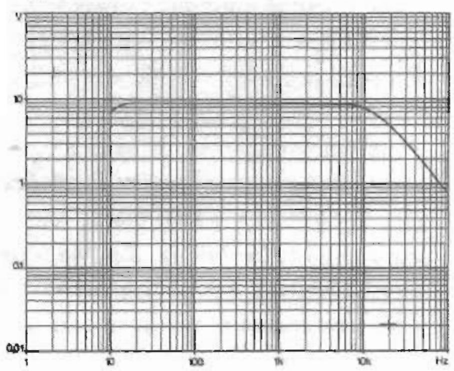
The record itself largely determines the maximum signal level which will be fed to the input of the preamplifier. In many designs, the maximum input before overload has been determined by a guess as to what the maximum value might be. Bang & Olufsen's experience in cutting test records shows that the maximum velocity that a top quality cutting head can be expected to engrave on a record is 40 cm/sec.

Fig. 1.



Phono Input -- Maximum input voltage against frequency.

Fig. 2.



Maximum output after equalisation from phono preamplifier.

A magnetic cartridge is a velocity dependent device, and will put out a constant voltage for a constant recorded velocity. Today, all known high quality magnetic cartridges have sensitivities below 2 mV/cm/sec. Thus, the maximum signal that can be received by a gramophone preamplifier is in the order of 80 mV. Designing a preamplifier which can handle a higher signal level would therefore seem pointless. Unless, of course, it could be done at no cost – both in terms of economic costs and specification compromise. But we already know that the higher the signal level the preamp is required to handle, the more noise it will generate – and that is one major undesirable "cost".

The design team therefore accepted the limit of 80 mV as the maximum signal to be handled, and the noise figure is optimised for the Bang & Olufsen range of cartridges with their typically low impedance.

Since there is a separate preamp for the phono input, there is no switching between the input and the preamp, thus, there is neither measureable nor audible noise which usually characterises this source. The switch is placed at a high level stage after amplification.

The preamplifier itself is a two-transistor circuit with feedback tailored to give the equalisation characteristics required for magnetic cartridges. Strict control over the parameters in this design ensures a noise figure which is not more than 2 dB above the theoretical thermal noise for the impedance of a magnetic cartridge. Noise generated in the preamp remains virtually unaltered with magnetic cartridges of other makes, with impedances in the practical range.

A pair of input sensitivity controls in the bottom cover allows pickups of higher output to be accomodated, or to balance the outputs from the two channels of the pickup. This control should normally be set to its most sensitive position, or at a position where the average sound level from records is the same as that from FM radio.

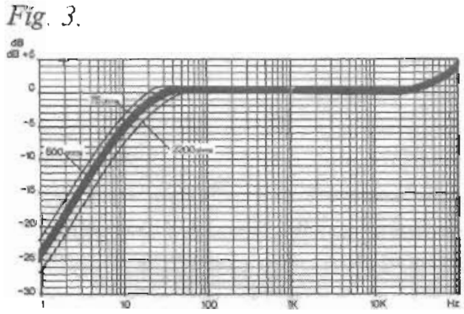
High pass filter.

A high pass filter is built into the preamplifier circuit. This cuts low frequencies below 30 Hz, so that the pickup arm resonance of a good quality turntable is adequately damped and has no audible effect on amplifier performance. Admittedly, this is not enough to remove rumble from poor quality turntables, but we expect that the Beomaster 4400 will be used with a turntable which is its equal. Rumble recorded on poor quality records, or the effects of warped records will not be obtrusive but full quality will be obtained with good quality equipment and records.

Equalisation.

Frequency response, or in this case correct equalisation, is determined by the tolerance of the components in the feedback loop. This is basically a production problem, if the design is right in the first place. With the aid of computer analysis of variation limits due to components in the preamp, we have determined the component tolerances necessary to keep deviation within 0.5 dB of theoretical equalisation, in the 30 Hz – 20 kHz region. This performance can be guaranteed for every receiver.

Fig. 3.



Phono Input equalisation – Deviation from RIAA showing the effect of pickup cartridge impedance on the built-in "rumble" filter. Maximum deviation in a random sample of the Beomaster 4400 will lie within the boundaries of the thick line.

It has been a design objective that distortion at any frequency for any level up to the maximum for which is has been designed, should be less than 0.1 %. This goal has been easily achieved. In practice, total harmonic distortion is below 0.05 %, at any frequency in the audio range.

Thus, because this preamplifier has negligible noise and distortion characteristics, it will handle all signals from any available record with utmost fidelity. The limiting factor will always be the quality of the record: its noise level and dynamic range, distortion and frequency response, and the quality and performance of the cartridge and turntable.

TAPE RECORDER INPUTS AND OUTPUTS

The Beomaster 4400 has sockets for two tape recorders. Both sockets are wired to record from the source or programme chosen via the selector press buttons. The outputs to these sockets are taken from a stage before the tone control circuits, so that while listening to the programme, volume and tone controls can be set to any position without affecting the recording in any way. On replay, the volume and tone controls are fully operative, as on any other input source.

Tape 1 facility.

The tape 1 socket acts either as tape replay or monitor input, in conjunction with the corresponding press button selector. The monitoring facility is used simply by pressing the tape 1 switch, an action which will not release the programme source selector button. While recording proceeds uninterrupted, the recorded programme will be heard on the loudspeakers. Pressing the selected programme source button again will release the tape 1 switch, and the loudspeakers will then continue to reproduce the source of recording without interrupting the recording. The tape 1 output consists of an emitter follower output stage, with an attenuating circuit providing a signal dependant on the input impedance of the connected tape recorder, as required by DIN standards.

A sensitivity control on the bottom cover allows connection to any tape recorder, or the adjustment of balance, on both tape recorder inputs.

Tape 2 facility.

The tape 2 facility in intended for use with a cassette recorder or other tape recorder without a monitoring facility. It was considered, that the inclusion of monitor facilities on both tape sockets would unnecessarily complicate operation, while additional costs could not be justified in everyday use.

The tape 2 socket and switch can also be used as an input for any other source and may therefore be regarded as an auxiliary input, but with an output for a tape recorder. This output is also fed from an emitter follower stage, followed by attenuation to DIN standards.

TONE CONTROLS:

The Volume Control.

The Volume Control does not present a design problem but its position in the circuit plays an important role. In order to ensure that it has no adverse effects on the amplifier's performance, the circuit designer must be aware of the consequences of his decisions regarding the position of the volume control.

For example, the designer must ensure that, for any setting on the volume control, no part of the circuit will overload before the output stage is unable to follow the required level. This may sound fairly obvious, but it is often forgotten, with unexpected results. In the case of the Beomaster 4400, it is a fundamental concept throughout the design, from every input stage to the output.

Consider the two possible positions: either before or after the tone controls. If the volume control is placed after the tone controls, it is quite possible that the signals from the input amplifiers will overload the tone control circuits, when these are set to boost the ends of the frequency spectrum by 12 dB or more. The 12 dB corresponds to 4 times voltage gain, anf if the input receives a signal of say 5 volts, the tone control stage will be expected to handle 20 volts without overloading. This is not practicable – especially from a cost point of view. If, on the other hand, the volume control is placed before the tone controls, it is quite possible that, when operating at fairly low levels, the tone controls will add noise to the circuit. In many amplifiers a compromise is made: the volume control is placed after the tone controls and the designer hopes that the user will be careful in his use of the tone controls.

In the Beomaster 4400 it is impossible to overload any part of the circuit before the output stage, allowing the tone controls to be fully exploited in everyday use. The Beomaster 4400's volume control precedes its tone controls. Very careful design, together with an additional amplification stage for the tone controls ensure that the extra noise from the tone control circuits is of no consequence for the final sound quality.

This is one of the cases in the receiver circuit where is was felt that the added cost of extra circuitry is fully justified by the audible improvement in performance. Careful and detailed analysis alone were not adequate to give the required subjective quality.

Balance Control

The balance control is designed, so that when it is in its centre position, there is no attenuation of either channel, preventing the addition of noise to the programme. Regulation is small immediately on either side of the central position, with increasing regulation further from the centre to ensure that small differences in levels between the channels can be accurately and easily adjusted. At its extreme positions, one channel is completely faded, while the other is at the same level as when the balance control was in its central position. It does not operate on either of the tape recorder outputs.

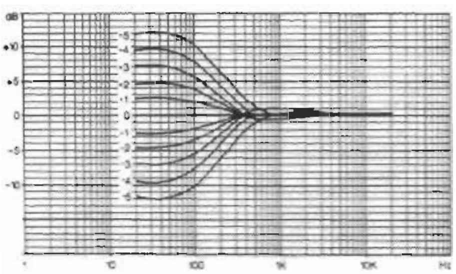
Bass and treble Controls.

The tone controls are designed around active filter circuits and optimised to ensure that only a minimal amount of noise and distortion are added to the signal. Noise levels have been kept well below those required to cover the dynamic range of any known signal source. The tone control curves selected are those which listening tests have shown to be effective for correction, and the least objectionable with regards to side effects. Component tolerances and filter curves are calculated on a computer for consistency and accuracy.

Maximum gain from the tone control is limited to 12 dB at the extremes of the frequency spectrum. This has been done deliberately since no known music source requires compensation beyond these values. Furthermore, higher values of gain lead to unacceptable side effects, irrespective of the circuitry, its sophistication and expense. It should be remembered that tone controls are designed to correct small deficiencies in the frequency response of the programme, or to compensate for imperfect room acoustics. Ideally, with good quality sound sources, the tone controls should be flat.

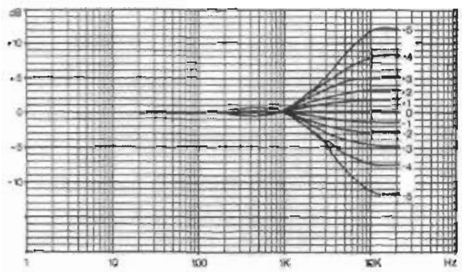
The figure below shows the effect of the tone controls at various cursor settings.

Fig. 4.



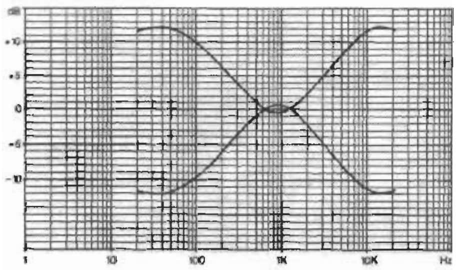
Bass Control only.

Fig. 5.



Treble Control only.

Fig. 6.



Tone controls at maximum setting.

High and Low Filters.

Minor defects in the programme source can be compensated by the use of 2 filters: a lo-filter, which attenuates record or turntable rumble, and a hi-filter, which renders noise or distortion from programmes less objectionable. Neither filter has serious side effects. They will not remove more than is necessary to correct slightly imperfect programmes. However, no filter can transform bad programme material into good sounding material and any attempt to do this can result in the programme sounding even worse than it would without filters.

Mono Switch

A mono switch parallels the input to the two channels so that the combined output is fed to both output channels. It should be noted, that in the case of a mono input received on only one of the stereo channels, pressing the mono switch will attenuate the signal by

6 dB. A small correction to the volume control will allow full volume replay but inevitably, a small amount of noise will be added.

The balance control is fully operative, when the mono switch is in function, and controls the signal level to the output stages. It can be used to feed the signal to either speaker if necessary.

Loudness Control.

A loudness control is included to allow the user to compensate for the ear's reduced sensitivity to frequencies at both ends of the frequency spectrum when listening at low volume levels. This is a feature, which many find useful although with any loudness control, the compensation does not correspond to any counterpart in nature, and is therefore not strictly "Hi-Fi".
is given below.

Linear Switch.

Any electronic circuit, however expensive and carefully designed, will have tolerances and will add some – at best negligible – noise and distortion to the reproduced sound. There is no logical reason why these deficiencies should be imposed on the listener who does not wish to use tone controls. That is why the Beomaster 4400 has a linear switch. This switch bypasses or defeats all tone and loudness controls, filters, etc., and guarantees a perfectly flat frequency response through the amplifier. It simply removes all audible effects of the tone control circuitry.

Loudspeaker Switching

Two press buttons marked LS1 and LS2 allow the permanent connection of two pairs of loudspeakers, which can be selected independantly. In addition, LS2 works in conjunction with the "Ambio" switch on the rear panel, to feed the difference between the stereo signals to a pair of speakers at the sides or rear of the room, to simulate the concert hall ambience in which the recording was made. It should be noted that this is a simulation, and the effect will vary with the recording. No extra circuitry is required, and the extra pair of loudspeakers can be connected directly to the LS2 sockets.

THE FM TUNER

A tuner means different things to different people. To the Hi-Fi enthusiast a tuner is a audio component, which receives and converts radio signals into audio signals, ready to be amplified further. To the circuit designer, however, a tuner is the RF circuit or front end, which selects a particular station from the many which may exist in the particular radio frequency band. In order to avoid confusion, the part which selects a station will be called the front end, while "tuner" will indicate the complete receiving section ending in an audio output, ready to be fed to the audio amplifying stages in the receiver.

A hobby of "radio amateurs", known as DXing on the FM band, is in its essence the attempt to receive very weak or distant stations. The Beomaster 4400 is not designed to exhibit the kind of performance required for the hobbyist, as a primary aim. To be suitable for DXing, a tuner must optimise signal to noise ratio at very weak signal levels, either by narrowing audio bandwidth, and therefore at the cost of audio quality at reasonable signal strength, or at prohibitive economic cost.

In the Beomaster 4400, emphasis was laid on the performance in areas of reasonable signal strength. This does not mean that the tuner section is incapable of receiving weak or distant stations. This will be received on the Beomaster 4400 at as well as on any but the most specialised tuners. But signal to noise ratio under these receiving conditions has not been a primary consideration in the design, whereas providing high fidelity reception was given the highest priority.

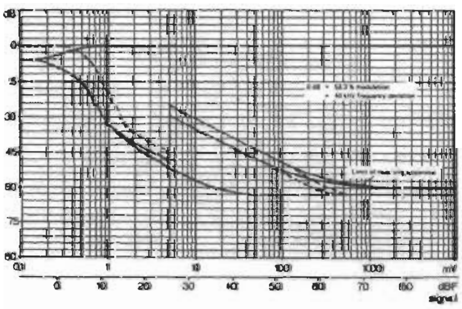
The Front End.

The front end of the Beomaster 4400 is the only part in the entire receiver which is retained from its predecessor. It consists of a well tried and proven circuit, which, as recent tests carried out in the south of Germany have shown, is able to hold its own among the very best and most expensive modern tuners. This area in Germany is notorious for its difficult receiving conditions because of numerous transmitting stations from 3 countries, as well as hills which cause both reflections and signal attenuation.

The front end is designed around two FETs, with varicap diodes accounting for voltage dependent variable capacitance. Voltage is supplied from a regulated power supply, through six preset potentiometers and selector switches, or a potentiometer coupled to the tuning scale. This arrangement allows six stations, which are listened to frequently to be pretuned, subsequent station selection is then at the touch of a button.

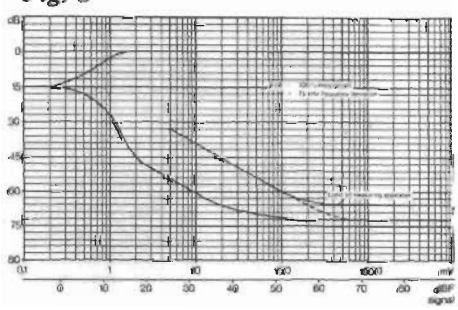
The RF stage has a signal/noise ratio for 50 dB quieting of only 2 μV (18 dBf) for mono, and 20 μV (38 dBf) for stereo, both measured to IHF standards. Ultimate quieting for high signal strength, above 487 μV (65 dBf) is at least 65 dB in both mono and stereo modes.

Fig. 7



Signal/noise ratio against signal stregnth at aerial, to DIN standard. Frequency deviation 40 kHz.

Fig. 8



Signal/noise ratio against signal strength at aerial (75), to IHF standards. Frequency deviation 75 kHz.

The I–F Section.

The tuning circuit is followed by an intermediate frequency gain stage, from which a signal is fed back to the tuner for automatic gain control (AGC). The inclusion of this circuit ensures that the tuner cannot be overloaded by very high signal strength at the aerial, and cross modulation is virtually eliminated. Cross modulation results in spurious stations being heard anywhere on the tuning scale, or two stations at the same time.

From here, the signal is taken via a selectivity block containing three ceramic filters, to an integrated circuit with four stages of amplifier limiting, a double tuned quadrature detector and some secondary circuits.

A signal from the detector is passed through an integrated circuit amplifying stage, and fed back to the tuner, to give automatic frequency control (AFC). This reduces tuning error by a factor of fifteen in the Beomaster 4400, as against approximately five in most other tuners. The AFC circuit, in conjunction with the tuning lamps, is the most effective method of making noise and distortion due to off centre tuning inaudible. AFC is automatically switched off, if a weak station is received, in all positions of the switches.

The secondary circuits of the IF stage supply signal to the tuning lamps, and the field strength meter. This meter shows full scale deflection at 500 μV signal at the aerial and has a logorithmic characteristic. This covers the range where signal strength is too low to give maximum quieting of which the receiver is capable. No further improvement in signal to noise ratio is obtained for signal strengths larger than full deflection on the meter in stereo, or reading of 3 on the scale for mono.

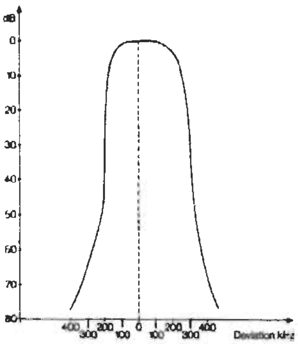


Fig 9

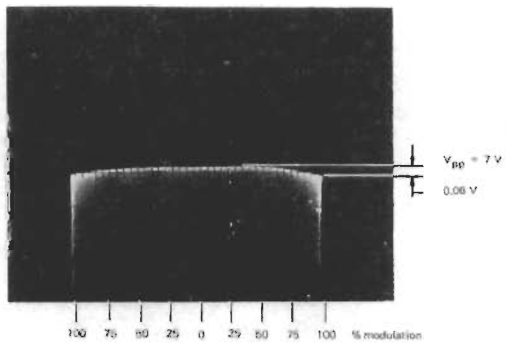
Selectivity to IHF standards.

The phase characteristic of the detector circuit is extremely linear, giving low distortion and good stereo separation, audibly maintained during loud passages where modulation level is high.

Fig. 10a

Tuner linearity for 150 kHz modulation bandwidth by the dual sweep method.

Fig. 10b

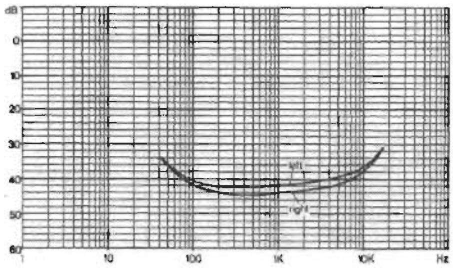


Same as a. but with expanded vertical scale for better resolution.

Stereo Decoder.

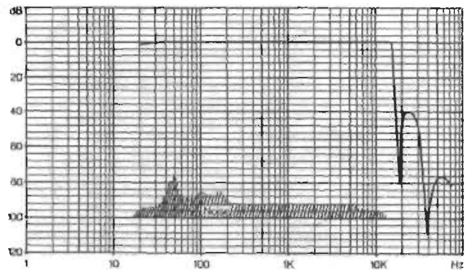
The original FM signal has now been converted to a mono audio signal, together with the encoded stereo component in the frequency band just above the audio frequencies. The next stage is therefore the multiplex stereo decoder, which is of the phase locked loop type, incorporated in an integrated circuit. The phase locked loop principle is relatively new, but the standard integrated circuit is capable of functioning extremely well, provided the signal fed into it has good phase characteristics. The IF strip with the quadrature detector used assures this, and the performance of the decoder closely approaches designs at many times its price. The combination of the detector and decoder give channel separation through the complete tuner section of at least 30 dB from 40 Hz or 15 kHz. However in the USA models, due to phase shift caused by the compulsory SCA filter (= background music service for paid subscribers such as supermarkets), phase linearity cannot be maintained all the way up to the required 53 kHz, and channel separation will not be up to the same standard at high frequencies. Even so, it will be at least 20 dB at 15 kHz, and the difference between the Beomaster 4400 and a tuner many times its price is subjectively marginal.

Fig. 11



Channel separation for 40 kHz deviation.

Fig. 12



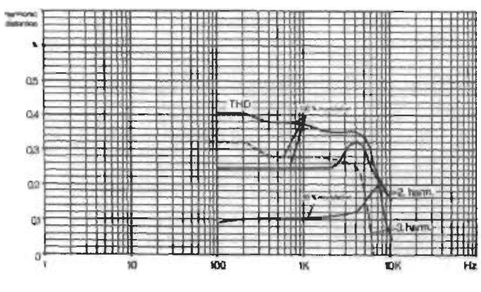
Frequency response after de-emphasis, and response of pilot tone filter. Noise level measured with 1/3 octave filter.

The decoder essentially delivers the two channels of a stereo signal, but superimposed are the stereo encoding frequencies of 19 and 38 kHz. These must be removed by a filter, and the characteristics of the filter determine both the frequency response of the tuner, and suppression of the pilot tone signal of 19 kHz, and the corresponding 38 kHz tone. This is accomplished by a computer designed filter, which gives a minimum pilot tone damping of 75 dB, and of the 38 kHz tone of 100 dB. At the same time, frequency response is maintained from 20 Hz to 15 kHz ± 1.5 dB. The pilot tone is therefore inaudible, while the 38 kHz tone will not interfere with bias frequencies in a tape recorder, or interfere with levels as seen by Dolby Noise Reduction circuits.

The final stage of the tuner consists of a single amplifying stage with cross talk adjustment and signal de-emphasis.

Audio signal distortion through the tuner on either mono or stereo signal is less than 0.3 % at mid frequencies and very near this specification at the limits of its bandwidth.

Fig. 13



THD for 75 kHz and 40 kHz deviation, and harmonic distortion analysis for 75 kHz deviation.

Other important specifications, which may be mentioned, are the tuner's capture ratio of 1.5 dB, and the alternate channel selectivity, which is at least 60 dB.

In short, the tuner section of this receiver has a performance which is very close to "state of the component art" tuners, which may cost much more than the complete Beomaster 4400 receiver. As a tuner for fringe areas, or in difficult receiving conditions, the Beomaster 4400 will pick out stations as well as any other in the market. The audio performance is of a quality where most listeners will be hard put to hear the difference between the Beomaster 4400 and any other tuner/amplifier combination or receiver on the market.

Fig. 14a

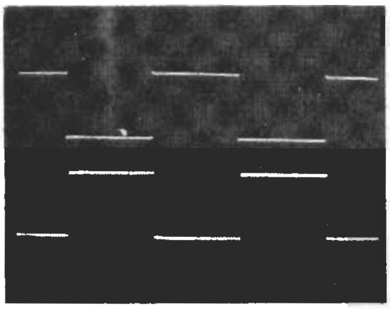


Fig. 14b

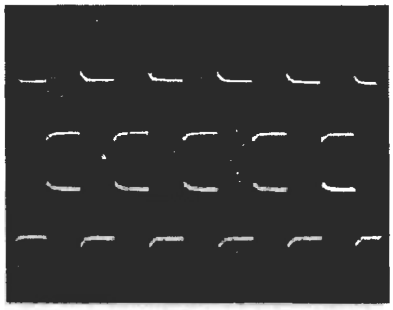
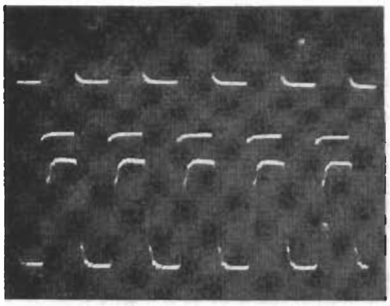


Fig. 14c



Square wave response through tuner. Signal frequency 1 kHz, upper trace shows signal from professional quality modulator. Differences between input and output in fig. c are due to pilot tone filter and de-emphasis.

Fig. 15a

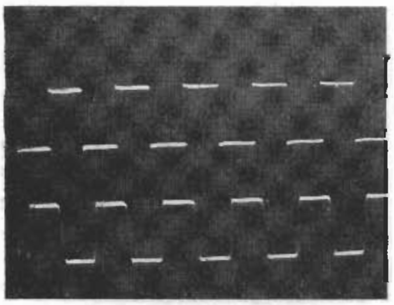


Fig. 15b

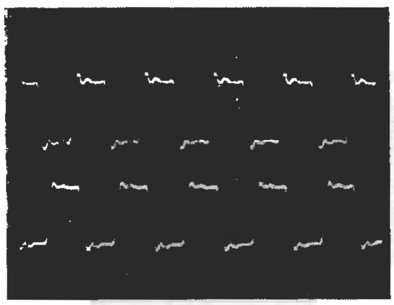
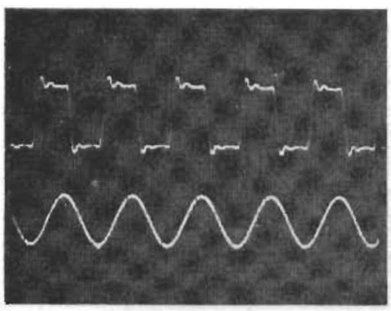


Fig. 15c



Square wave response through tuner. Signal frequency 10 kHz. At this frequency, all harmonics of the square wave are removed from the output (fig. c).

The Muting Circuit

Whenever a FM programme button is pressed, the muting circuit is automatically triggered, so that no noise occurs at change over. At the same time, AFC is defeated, and as soon as the circuit has stabilised, a process that takes a fraction of a second, the muting circuit fades in the station selected.

The same process of course accurs when the FM button for manual tuning is selected. The muting circuit then operates to mute inter-station noise, which also makes all stations below a pre-selected signal strength inaudible. A control in the bottom cover of the receiver allows the level at which this occurs to be pre-selected. A front panel switch defeats the silent tuning circuit, for reception of weak signals, and automatically decouples AFC.

In contrast to normal practice, the muting circuit obtains information not only from the IF stage, but also from the detector. This results in the total elimination of noise when the tuning cursor is close to, but not centred on a station. If the cursor is moved rapidly across the scale, the output is totally muted, only the tuning lamps indicating the presence of a station.

THE POWER AMPLIFIER:

The power amplifier is often called the heart of a hi-fi system. It is often forgotten, that audio quality is dependent on the quality of all the components: from the source of music (the gramophone record, the tape recording or the radio programme), to the loudspeaker. But providing the preceding components are of the same quality as the amplifier, the amplifier does indeed have a major role to play in determining the quality of the reproduced sound.

The importance of specifications.

From one point of view, the power amplifier is indeed different from the rest of the electronic amplifying chain. As discussed earlier, the problem of designing a preamplifier lies not in the circuit design itself but in defining what specification levels the design should fulfil. Once these are decided, design is a straightforward task for the competent circuit designer. However, the matter is not as clear-cut in the case of a power amplifier design. Firstly, because an amplifier has to be specified in many mare ways than a preamplifier in order to obtain the same sound quality. Secondly, all these specifications are so interrelated that improving one is more often than not, at the expense of some other, possibly equally important specification. Thus, even the most competent circuit designer has great difficulty in meeting a given set of specifications. The difficulty in specifying the amplifier in the first place is probably greater for an amplifier than for any other component. This is because so little is known about the importance of various amplifier specifications in relation to audio quality, and because it is known that many amplifier specifications which can be measured, are not those which are most important to the ear.

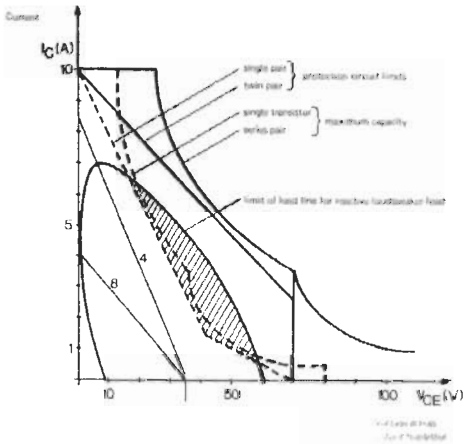
Load Handling Capability

One factor which gives the designer great problem is that the amplifier is expected to drive a component, a loudspeaker, whose specifications are unknown to the amplifier designer. The load presented by the loudspeaker can vary widely according to its type and design. Ideally the amplifier should work well with all of these, but this is not always so. No other component in the hi-fi chain is required to function under such varying conditions.

Commercially available loudspeakers almost never present a purely resistive load to the amplifier. The load varies with frequency, being inductive for most of the frequency range, but capacitive at certain frequencies. The ability of an amplifier to deliver its rated power to any speaker at peak levels in music, is an important characteristic in the subjective evaluation of its overall sound quality.

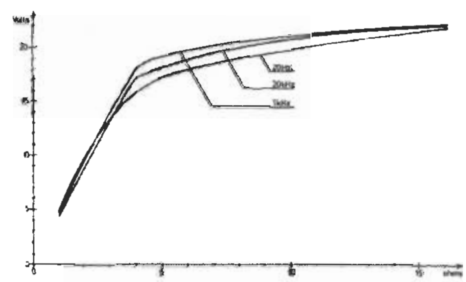
To be able to deliver full power to all loudspeaker loads places demands on both the voltage and current handling capacity of the power transistors in the output stage. Normally a voltage dependant current limiting protection circuit is coupled to the circuit, so that in the event of peak demand exceeding the capacity of the output transistors, they are switched off. The same circuit operates to protect the output devices against a short circuit of the signal output leads. The effect of switching off during musical passages is severe distortion in the short interval during and immediately after the signal peak, in severe cases heard as "buzzes" or "clicks".

Fig. 16



Current limiting protection circuits. With single pair of output transistors, current limiting will occur in the shaded portion, with load line indicated by the ellipse. Full protection can be given with series pairs in the output stage, with no danger of limiting.

Fig. 17



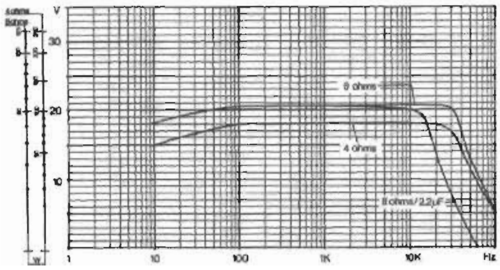
Output capability of Beomaster 4400 showing maximum undistorted voltage for various resistive loads for three frequencies.

In the Beomaster 4400 the problem is solved by using four output transistors instead of the usual two, so that it is never short of voltage or current to supply its rated power into any load above the specified minimum of 4 Ω. The transistors are high current devices connected in series pairs, so that the required voltage capacity is shared between them.

Irrespective of the loudspeaker connected, the load line lies well within the working capacity of the series pairs, and the protection circuit is designed to operate only in cases where it is genuinely required. When the amplifier is asked to handle more than its rated output, the result is virtually pure voltage clipping, which is audible, but nowhere near as irritating as the distortion and other effects due to current limited protection circuits.

The expense of doubling the number of output transistors is well justified by the improvement in sound quality at high listening levels.

Fig. 18



Output capability of Beomaster 4400 showing maximum undistorted voltage at various frequencies and loads.

Transient Intermodulation Distortion

A point, which is often misunderstood, is the relative importance of low distortion and wide bandwidth. The conventional method of obtaining these specifications is the use of negative feedback. While in general any improvement of specification is worthwhile, it is only true, if the improvement is not made at the expense of other specifications. But this is rarely the case. Whenever there is a choice, it is important, that the "right" decision is made, that is, that the more audible specification gets priority over one which may only look good on paper. Thus, in recent times a fruitless search for wider bandwidth and lower distortion has led to the use of ever increasing amounts of negative feedback.

Unfortunately, the use of increasing amounts of negative feedback can cause another kind of distortion. That this is so, was first pointed out by Matti Otala, who called this phenomenon transient intermodulation distortion (TID). A complete analysis of the conditions which give rise to TID and the criteria for its elimination were the subject of a paper by Tom Jelsing of Bang & Olufsen, presented at the 53rd. AES convention (Zurich 1976). This paper is attached at the end of this booklet. Jelsing concluded, that TID is easily eliminated for any given set of components, and TID presents no problems unless one attempts to obtain a level of paper specifications beyond the capacity of the components used. The results of Jelsing's findings have been fully expliofed in the design of the Beomaster 4400 receiver and subsequent receivers from Bang & Olufsen.

What is TID?

TID (Transient Intermodulation Distortion) occurs when, in order to reduce harmonic distortion and increase bandwidth, a small part of the signal from the output stage is fed back to the input to compensate for error between the distorted output and the undistorted input. This is a conventional technique which goes back to the days of valve amplifiers. However, in transistor amplifiers it is fairly easy to apply large amounts of negative feedback without causing instability – or at least instability that cannot be compensated for. This high level of negative feedback allows the designer to obtain any predetermined level of harmonic distortion. As limiting of frequency response through the amplifier causes a difference between input and output signals, applying negative feedback also increases bandwidth. Thus, one can find "super" amplifier specifications, in which distortion may be 0.05 %, together with a bandwidth of over 200 kHz. This level of specification can in fact be measured, but only on sine wave signals. Since we do not listen to sine waves, it is necessary to use the kind of signals of which music is composed in our measurements. On doing this, some peculiarities show up.

The problem occurs due to the very high level of feedback applied to reduce distortion and increase bandwidth, in that feedback attempts to compensate for errors in the forward amplifying chain. The greater the difference between the input and output, the larger the signal the feedback loop will attempt to generate, to compensate for the error, until at some point a level reached which the amplifier is not designed to handle, and distortion results. This phenomenon is particularly marked with transient signals, hence the name, Transient Intermodulation Distortion.

It is not long since the problem was aired, and to date no measurement standards exist. The fault is rarely mentioned in sales literature or printed specifications.

While the distortion due to a peak in the feedback loop overloading the amplifier will normally be too short to be heard directly, it can lead to audible intermodulation by-products, most noticeable in loud passages with appreciable high frequency content. The elimination of TID results in clean uncluttered sound when playing at high average levels, but within the power handling capacity of the amplifier even for peaks in the musical signal.

Elimination of TID

Others who have written on the subject, and quoted design criteria for eliminating TID, have stressed the need for wide open loop bandwidth and low gain, together with methods other than lag compensation to retain stability. The mathematical analysis of the whole problem by Tom Jelsing proved that this was a pessimistic view, and that methods, at a fraction of the cost of those proposed earlier, could be used to give equally acceptable results. TID can be eliminated from conventional amplifiers with lag compensation, provided adequate dynamic range exists at all critical points in the amplifier circuit – a detailed mathematical analysis gives the necessary conditions – and that signals of unnecessarily high frequency, or rise times, are prevented from reaching the amplifier. If this is done, TID is totally removed from the amplifier, not only for music signals, but also for any electronically generated signal that may be used to test for TID, with or without signal boost from the tone controls.

Fig. 19a

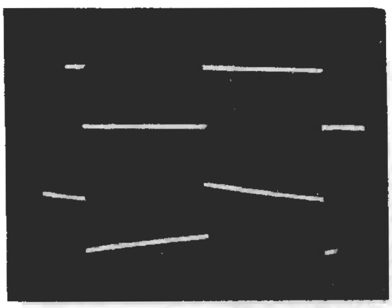


Fig. 19b

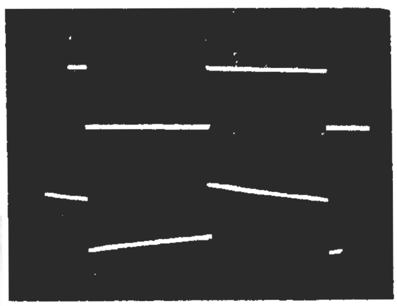
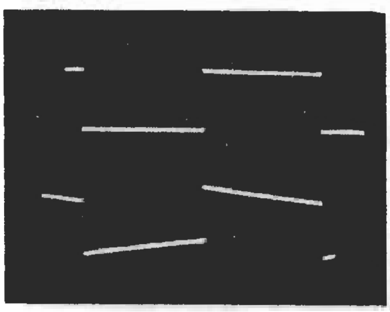


Fig. 19c



Square wave response at 100 Hz.

Note: In this, and all following oscillograms, fig. a, b & c are taken with loads 4 Ω, 8 Ω and 8Ω//2.2 μF respectively.

Fig. 20a

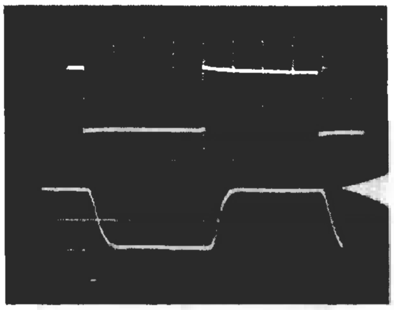


Fig. 20b

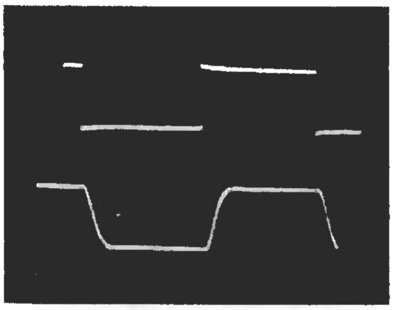


Fig. 20c

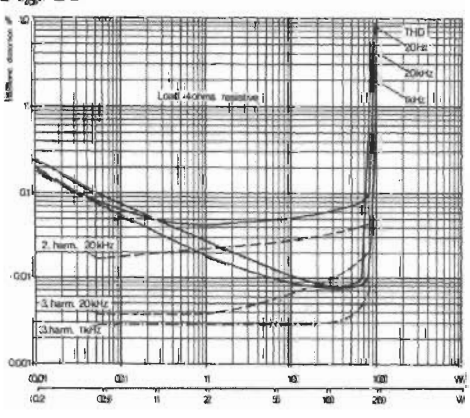
Square wave response at rated power, signal frequency 10 kHz.

In the Beomaster 4400, bandwidth is deliberately limited (– 0.5 dB at 20 kHz, – 3 dB at 50 kHz) a value which is adequate, both for the ear, and to cover the frequency range of all signal sources in domestic use today. This is done by introducing a low pass Bessel filter at the input to the power amplifier, which attenuates unwanted high frequencies, while maintaining a linear phase characteristic at all frequencies. Together with other details in design, this completely eliminates TID since no distortion can occur for any signal – neither sine waves nor transients – in the feedback loop. Along with this, linear phase response is maintained to the limits of the amplifier's bandwidth, giving the amplifier an almost perfect square wave response, indistinguishable from an amplifier with a much wider bandwidth.

Crossover Distortion

Most music is characterised by a fairly low, average-level signal punctuated with sudden, short peaks of high level transients. The average low-level signal required by the amplifier gives rise to other problems. The difference in quality between sound reproduced by valve and transistor amplifiers has been the subject of debate in recent times. The most obvious difference is perhaps that of the distortion characteristics of the two types of amplifiers. For example conventional valve amplifiers have the characteristic where,

Fig. 21



Harmonic distortion against output power at 4 Ω load. It can be seen that while noise remains constant, its ratio (or percentage) rises as power level is decreased. Harmonic distortion falls with decreasing power level.

as the power output decreases, distortion falls at a rate faster than the power output. Thus, if we examine a curve of percentage distortion against power output, the curve slopes upwards as power output is increased, but falls as it decreases until the distortion is drowned by noise. In conventional transistor amplifiers, which have a "class B" output, distortion falls with decreasing power output, to a power level of about 1 watt, below which percentage distortion begins to rise again. The exact level, at which minimum distortion occurs can vary, but the characteristic is different from valve amplifiers. The cause of this distortion pattern is the discontinuity as signal is switched from one output transistor to the other, and the type of distortion due to this cause is called crossover distortion. An analysis of the distortion products of an amplifier with crossover distortion shows predominance of the odd harmonics, when the amplifier is delivering say 1 watt. However, in the Beomaster 4400 the total harmonic distortion level is not far below average, but the even harmonics dominate in the harmonic residue. As the even harmonics are products, which are natural components of musical instruments, these do not audibly affect sound quality. Odd harmonics and thereby crossover distortion is virtually eliminated even at frequencies as high as 20 kHz. Thus, the ear is not subjected to the unnatural sound of a higher level of say third harmonic than second harmonic, or generally of odd harmonics which are louder than even harmonics. This results in an amplifier, which has an uncluttered sound at low levels.

Power output.

Power output is perhaps the most misunderstood amplifier specification. A brief paper on the subject presented by S. K. Pramanik to the 53rd. AES Convention (Zurich 1976) is attached at the end of this booklet. Briefly, power output alone does not determine the level of sound reproduced in a room, but the speaker combination has at least as large an effect. Thus, by itself, power output has very little meaning, expect as an aid to matching speakers to the amplifier. Too little power can result in severe amplifier distortion when trying to reproduce high level music with an inappropriate speaker, while too much power can cause severe speaker distortion, or worse destroy the speaker. Ideal results are obtained when the amplifier can deliver slightly more power than the speaker requires for maximum undistorted output, provided that the amplifier is not used continuously to its maximum level, since this will destroy the speakers.

The worst abuse of the power output specification is the criterion known as "watts per dollar". By its very nature, this specification does not take into account the quality of those watts. And that is really what determines the quality of an amplifier. High rated power output can be obtained quite cheaply if one disregards those specifications which determine the quality of sound obtained from an amplifier. A low power, high quality amplifier will often sound very much better than a poor quality, high power amplifier, which may be considerably cheaper.

The circuitry of the amplifier appears conventional, except to those with detailed knowledge of electronic circuits. Some may notice that the methods used to achieve conventional ends differ in detail. Others may notice, that although specifications do not appear to be outstandingly better under conventional measurements, there are specifications, which under special tests in actual use show marked improvements. Power output is rated at 75 watts per channel into 4 ohms. This means that a minimum of 17.3 V RMS will be delivered to any load provided the total impedance remains above 4 Ω resistive. The load may be inductive or capacitive of any reasonable value without altering the delivered voltage. Of course, as in any other transistor amplifier, available voltage will rise slightly if the minimum load is higher, giving it a rating og 50 watts (20 V RMS) at 8 Ω. This increase in voltage will also apply to high level signals delivered for a short period, thus

Fig. 22a

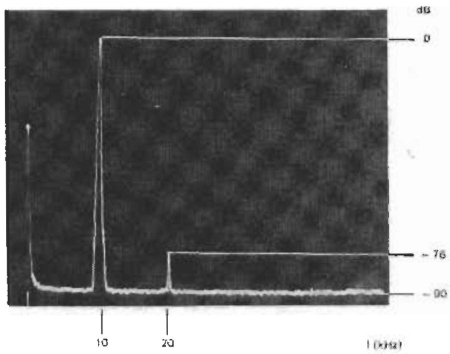


Fig. 22b

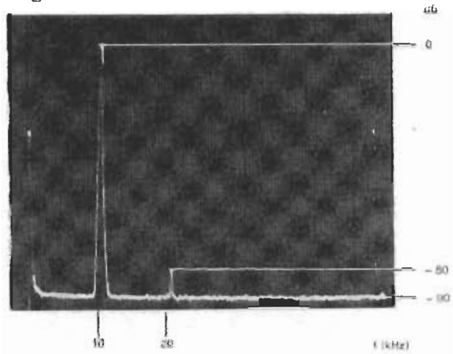
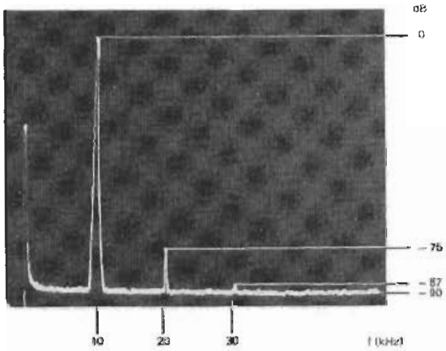


Fig. 22c



Spectrum analysis of harmonic distortion, into resistive and capacitive loads. Signal frequency 10 kHz, output 1 W. Test for crossover distortion.

Fig. 23a

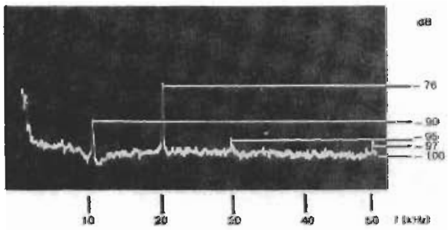


Fig. 23b

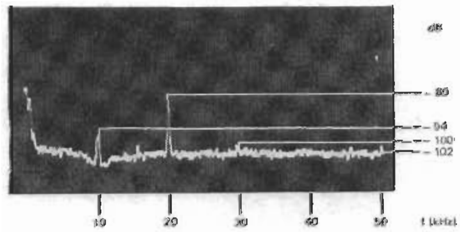
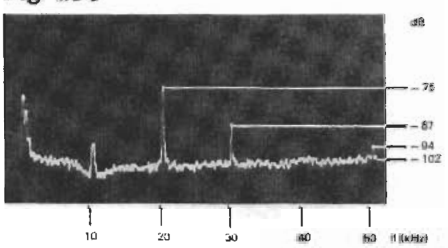


Fig. 23c



Spectrum analysis, same as above with fundamental removed for greater resolution of harmonic components. Analysis shows almost complete absence of odd harmonic components, which are the symptoms of crossover distortion.

giving the amplifier a fairly high transient power capacity. This, and the amplifier's ability to deliver its power into any load, means that the listener will not suffer the irritating experience of transient clipping caused by TID, lack of transient power, or current limiting due to loudspeaker load. The Beomaster 4400 is one of the very few amplifiers, that can boast this feature. The power rating of 75 watts per channel quoted for the Beomaster 4400 is its continuous power rating.

The overload indicator.

Every power amplifier has a limit to the amount of power it can deliver to a loudspeaker. This power level will vary depending on the kind of load the loudspeaker presents, the kind of signal, ambient temperature and the temperature of the heat sinks, etc. For any correctly specified amplifier, the minimum power output quoted in the specification will

Fig. 24a

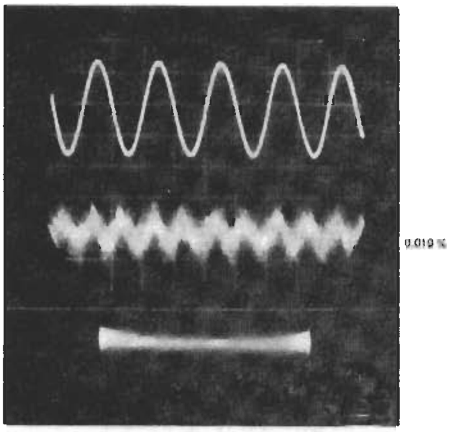


Fig. 24b

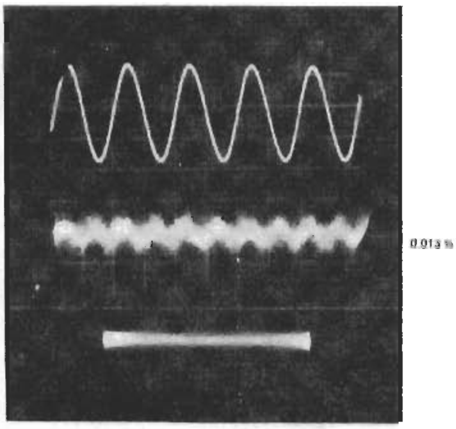
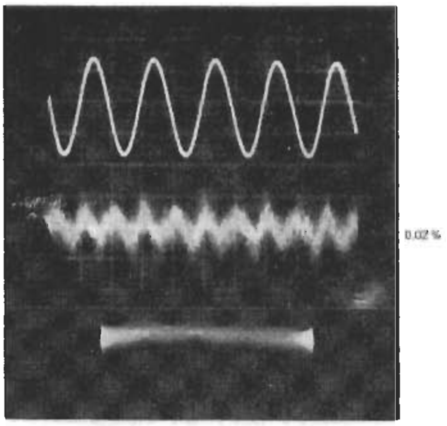


Fig. 24c



*Top: Input signal 10 kHz, output 1 W.
Middle: Harmonic components with fundamental removed. Residue is mainly 2nd harmonic.
Bottom: Harmonic distortion residue on Y-axis, signal on X-axis. The absence of any loops in the plot indicates the absence of odd harmonics, and therefore of cross-over distortion.*

Fig. 25a

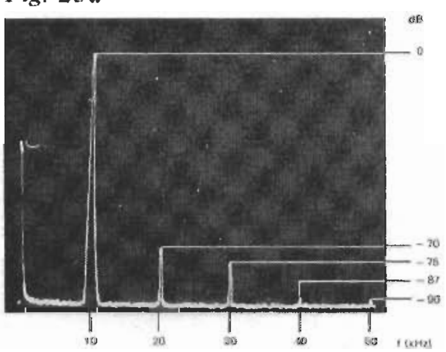


Fig. 25b

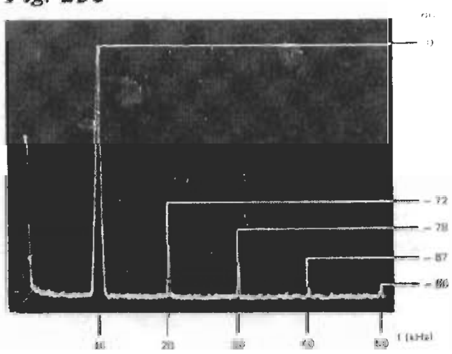
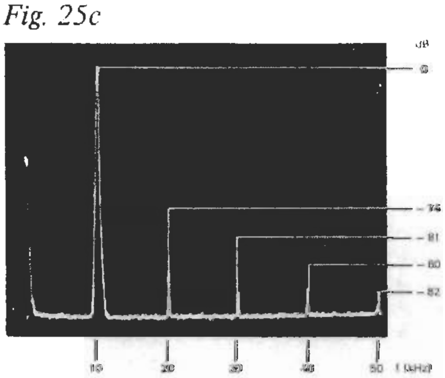


Fig. 25c



Spectrum analysis of harmonic distortion at rated power, signal frequency 10 kHz. Note progressively smaller amounts of even and odd higher order harmonics. 10 kHz is chosen to illustrate harmonic generation in the amplifier as this is generally more difficult to reproduce than lower frequencies.

Fig. 26a

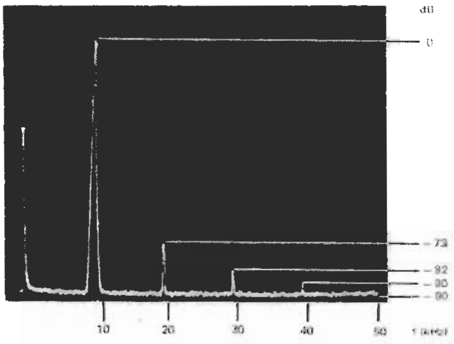


Fig. 26b

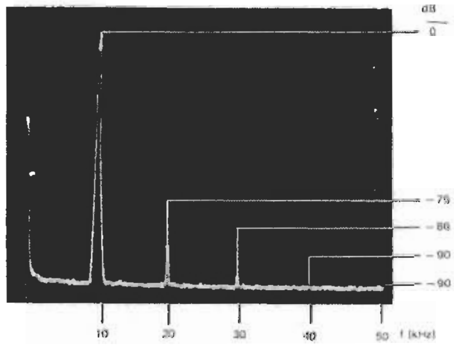
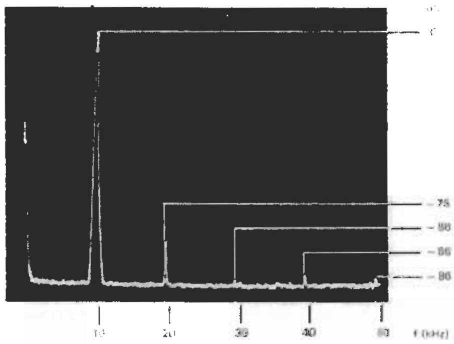


Fig. 26c



Spectral analysis, same as above at –3 dB rated power.

Fig. 27a

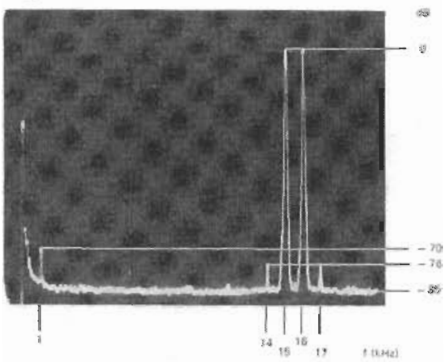


Fig. 27b

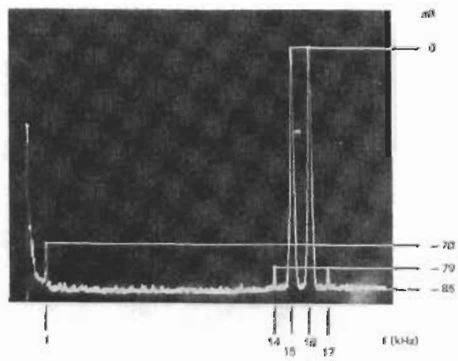
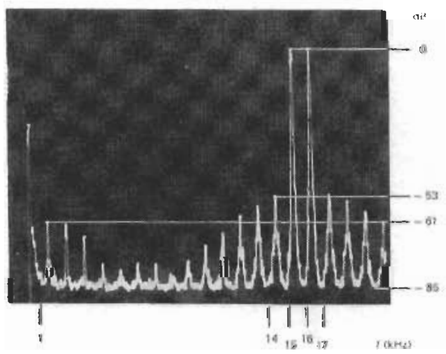


Fig. 27c



Spectrum analysis of intermodulation distortion at rated power, signal frequencies 15:16 kHz, amplitude ratio 1:1. Note that in Fig. c, the amplifier is just above its clipping limit, but there is no current limiting.

be delivered under all conditions, although it may deliver more than rated output under favourable conditions.

When pressed to deliver more than its capacity an amplifier distorts with unpleasant audible results. If however this distortion lasts a very short time, it may not be consciously noticed, or its cause identified. For this reason a visible overload indicator is placed on the front panel, which lights up at an output level, which gives about 1 % distortion,

Fig. 28a

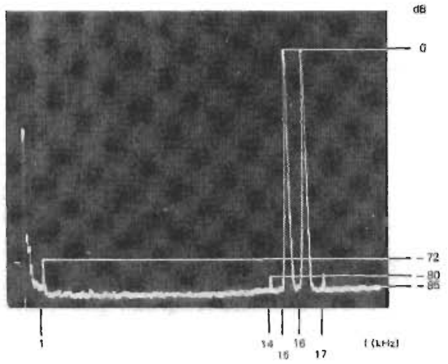


Fig. 28b

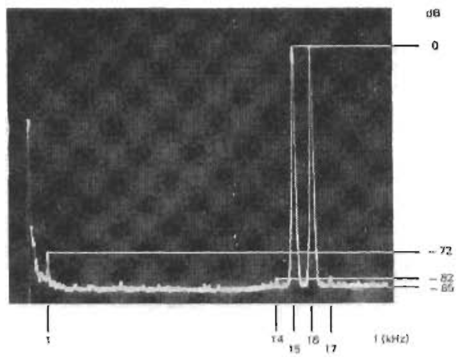
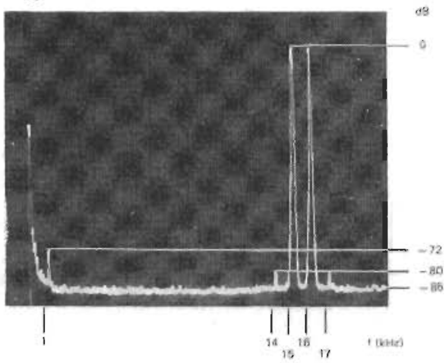


Fig. 28c



Spectrum analysis, same as above at –3 dB rated power.

Fig. 29a

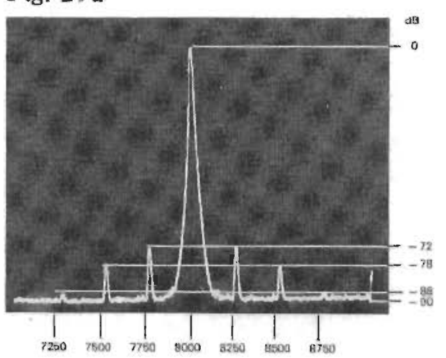


Fig. 29b

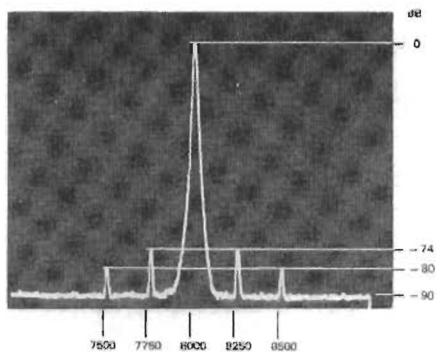
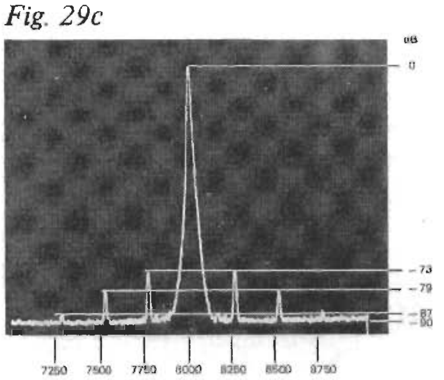


Fig. 29c



Spectrum analysis of intermodulation distortion at rated power, signal frequencies 250:8000 Hz, amplitude ratio 4:1.

even for a signal as short as one half of a sine wave. The light will then stay on long enough to be clearly visible. No damage will occur if the lamp lights, or even stays lit. It only indicates that the output stage is clipping, and therefore producing a higher than specified level of distortion.

The indicator is not a static device indicating a fixed power or voltage level. It operates

Fig. 30a

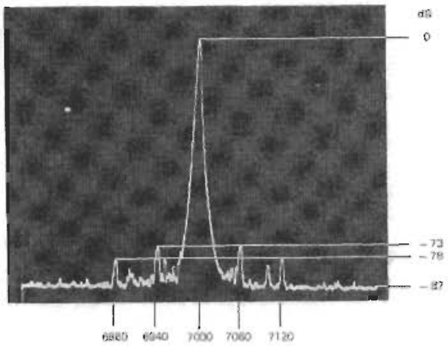


Fig. 30b

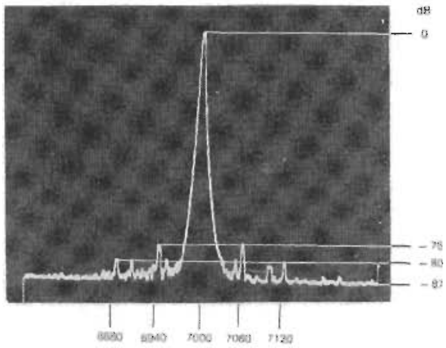
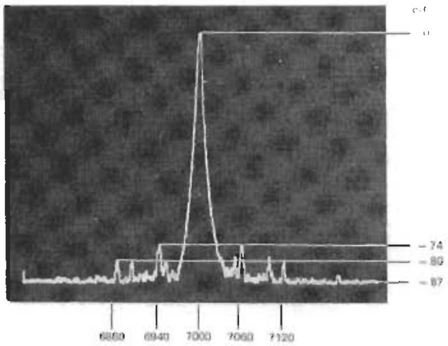


Fig. 30c



Spectrum analysis of intermodulation distortion at rated power, signal frequencies 60:7000 Hz, amplitude ratio 4:1.,

Fig. 31a

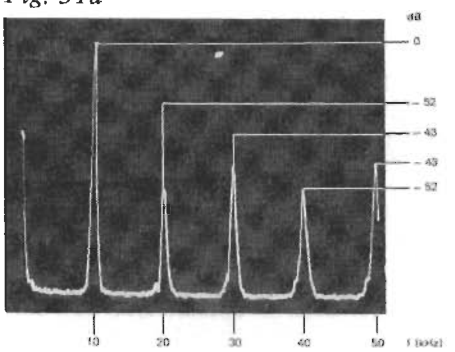


Fig. 31b

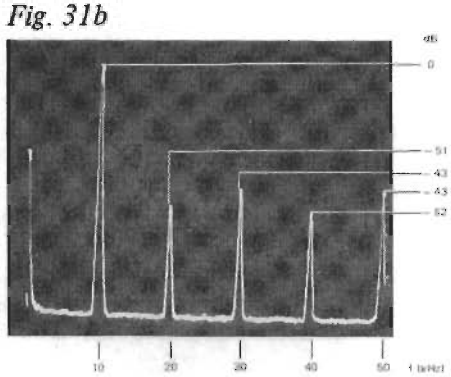
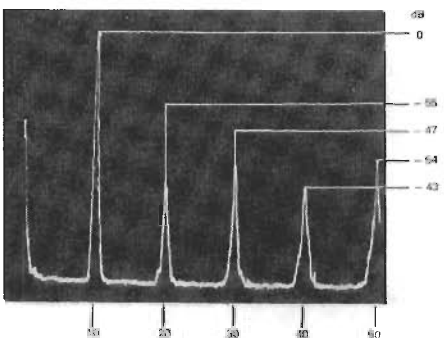


Fig 31c



Overload characteristics of the amplifier measured at 1 % THD. This is the level of overload indicated by the warning light on the front panel.

dynamically by measuring the instanteneous voltage output capability of the output stages, and comparing this to the output signal level. Thus, it indicates actual overload conditions, irrespective of the load, mains voltage, temperature or signal. This allows the listener to keep volume levels within the capacity of the amplifier under all combinations of operating conditions.

Fig. 32a

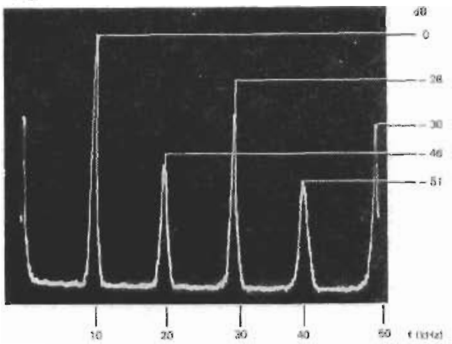


Fig. 32b

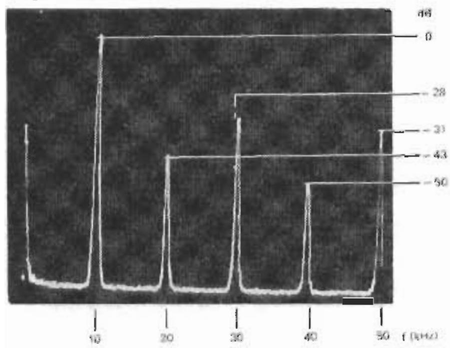
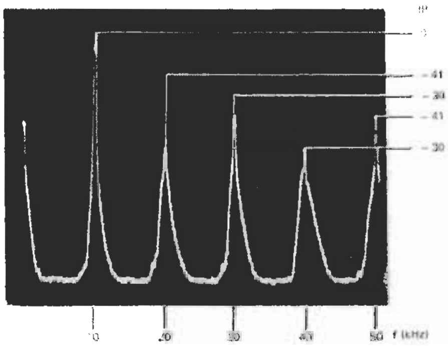


Fig. 32c



Overload characteristics of the amplifier measured at 5 % THD. Note that the spectrum analysis has characteristics similar to that at 1 % THD, showing the absence of current limiting in the amplifier protection circuit.

Protection Circuits.

Protection circuits are a part of every modern transistor amplifier, but their function, method of operation and subsequent effect on performance, as well as their ability to protect the amplifier and associated equipment, can vary widely.

The Beomaster 4400 features fully electronic protection for the output transistors in the event of short circuit of the output leads. But the power amplifier in the Beomaster 4400 is designed to deliver full power to even less than the minimum specified load, without activating the protection circuit, thus eliminating the unpleasant sound heard from less carefully designed amplifiers at high volume levels.

Even when short circuit conditions exist, the protection circuit will not necessarily operate, but will do so only if the current in the output transistors becomes so large, that protection is necessary. In such a case, overload will be seen on the indicator lamp on the front panel.

Should a short circuit condition persist, heating of the power transistors and heat sinks will result. In this case, before damage can occur, a thermal protection device will operate, and switch off a relay that supplies power to the amplifier circuit. As soon as the heat falls to a reasonable temperature, the realy will switch back to the operating position.

Finally, the receiver features protection for the loudspeakers in the event of output transistors failure. Since the Beomaster 4400 features fully complimentary output stages, there is no capacitor to stop DC voltage reaching the loudspeakers. Under normal operation this will not occur, but if one of the output devices fail, it may do so with catastrophic results for the loudspeakers. The DC voltage on the output leads it continuously measured, and in the event of abnormal operation the power supply relay will switch off so that the voltage does not reach the loudspeaker. The power supply will not be replaced, while the fault condition exists.

There are numerous other technical details, which contribute to the overall outstanding performance of the Beomaster 4400, but as they are mostly the result of conventional techniques, it will be pertinent to avoid lengthly discussion. Besides, in our opinion the success of the Beomaster 4400 is due, not so much to technical details but to a new emphasis on the formulation of our product's performance objectives combined with design ingenuity in achieving those objectives.

With the Beomaster 4400, Bang & Olufsen shows, that well-thought out and well-designed conventional technology can not only yield paper specifications, that are very close to the best obtainable but also secure an audible quality, which is equal to the performance traditionally associated with far more expensive receivers, amplifiers and tuners.

The Advantages of Integration.

An alternative to the integrated receiver is of course to design the amplifier and tuner electronics into two separate components. While matching problems are unusual for these two components, the optimum potential of individual components will not be realised unless the two sections are of equal quality.

From a consumer point of view the advantages of choosing separates instead of an integrated receiver are almost nonexistent today. This is so for several reasons. First and foremost, the receiver manufacturer does not need to duplicate a number of the components common to the tuner and amplifier, and is therefore always able to offer better overall performance for the same amount of money. For example, all electronics can be incorporated onto the same chassis. One common cabinet and front panel can be used. Only one power transformer and one power supply are necessary, etc.

Furthermore, the receiver manufacturer is able to guarantee a perfect match between the two sections not only in performance and specifications but also in terms of functional and aesthetic design.

The fact is, that the idea of designing separate tuners and amplifiers is a left-over from the days when the excess heat produced by very large vacuum tubes in the electronic section, tended to damage the operating characteristics of the more "delicate" tuner and preamplifier sections of a receiver. The use of "solid-state circuitry" has virtually eliminated this problem, so that there is no technical reason for choosing separates today. Even the usual claim, that separates provide additional connection facilities and features is no longer valid – especially when seen in relation to the actual use to which such extra facilities are put and the extra cost of having them there.

There seems to be one valid reason why a hi-fi consumer might prefer to buy separates instead of an integrated receiver. For example, the person who has a limited amount of money to spend, and whose main interest is listening to records, might be better advised to invest that money in a better-quality amplifier and a gramophone and put off purchasing FM or AM radio as programme sources until a later date, instead of settling for a "budget" system with perhaps "budget" quality.

Technical Data

The specification quoted below are the "worst case" values that may occur during the production run of the Beomaster 4400. Any sample taken at random will have appreciably better data. Should specification be directly compared, it is relevant only in cases where the alternative specification is specifically quoted to be minimum values to DIN standards.

Sound system and loudspeakers	Stereo and ambio or 2 set stereo
Features:	
Radio ranges	FM
FM pretuning	6
Tuning indicator	Dual light and pointer
Muting FM	Yes
Loudness	Yes
Tone filters	Yes
Headphone socket	Yes
Tape recorder sockets	2
Monitor	Yes
Power output at specified	
distortion 1000 Hz RMS	2 x 75 watts/4 ohms
	2 x 50 watts/8 ohms
Music power	2 x 110 watts/4 ohms
	2 x 60 watts/8 ohms
Speaker impedance	4 ohms
Harmonic distortion:	
1000 Hz 50 mW DIN 45 500	0,05 %
DIN 45 500	< 0.3 %
Intermodulation DIN 45 500	< 0.1 %
Frequency range ± 1.5 dB DIN 45 500	20 – 35.000 Hz
Power bandwidth 1 % distortion	10 – 75.000 Hz
Damping factor 1000 Hz DIN 45 500	65
Pickup low impedance	2.2 mV/47 k ohms
2 channel linear	200 mV/470 k ohms
Signal-to-noise ratio DIN 45 500	
50 mW pickup low impedance	> 60 dB
50 mW high impedance	> 60 dB
Channel separation 1000 Hz	
DIN 45 500	> 45 dB
250 – 10.000 Hz	> 35 dB
Tape	100 mV/100 k ohms
Headphones	Max. 17,3 V/200 ohms
Bass control at 40 Hz	± 12 dB
Treble control at 12.500 Hz	± 12 dB
FM tuner/Range	87,5 – 108 MHz
Sensitivity stereo 46 dB	< 20 μV/75 ohms
Frequency range ± 1.5 dB DIN 45 500	20 – 15.000 Hz
Harmonic distortion DIN 45 500	< 0.3 %
Stereo channel separation 1000 Hz	> 35 dB
Pilot suppression 19 kHz	> 65 dB
38 kHz	> 100 dB
Power supply	110 – 130 – 220 – 240 volts
Power consumption	30 – 310 watts
Dimensions W x H x D	57.5 x 9.5 x 28 cm
Weight	10 kg

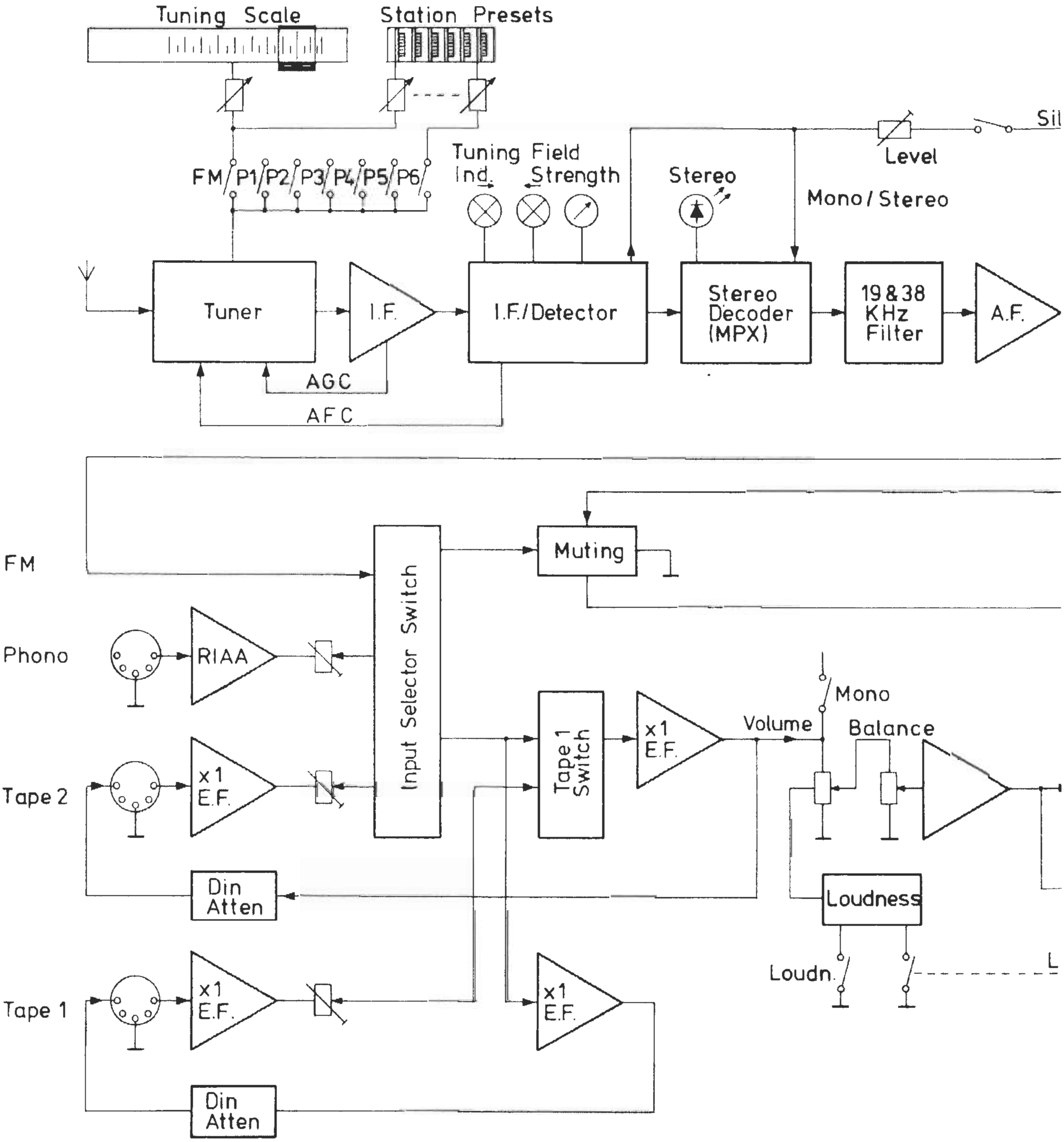
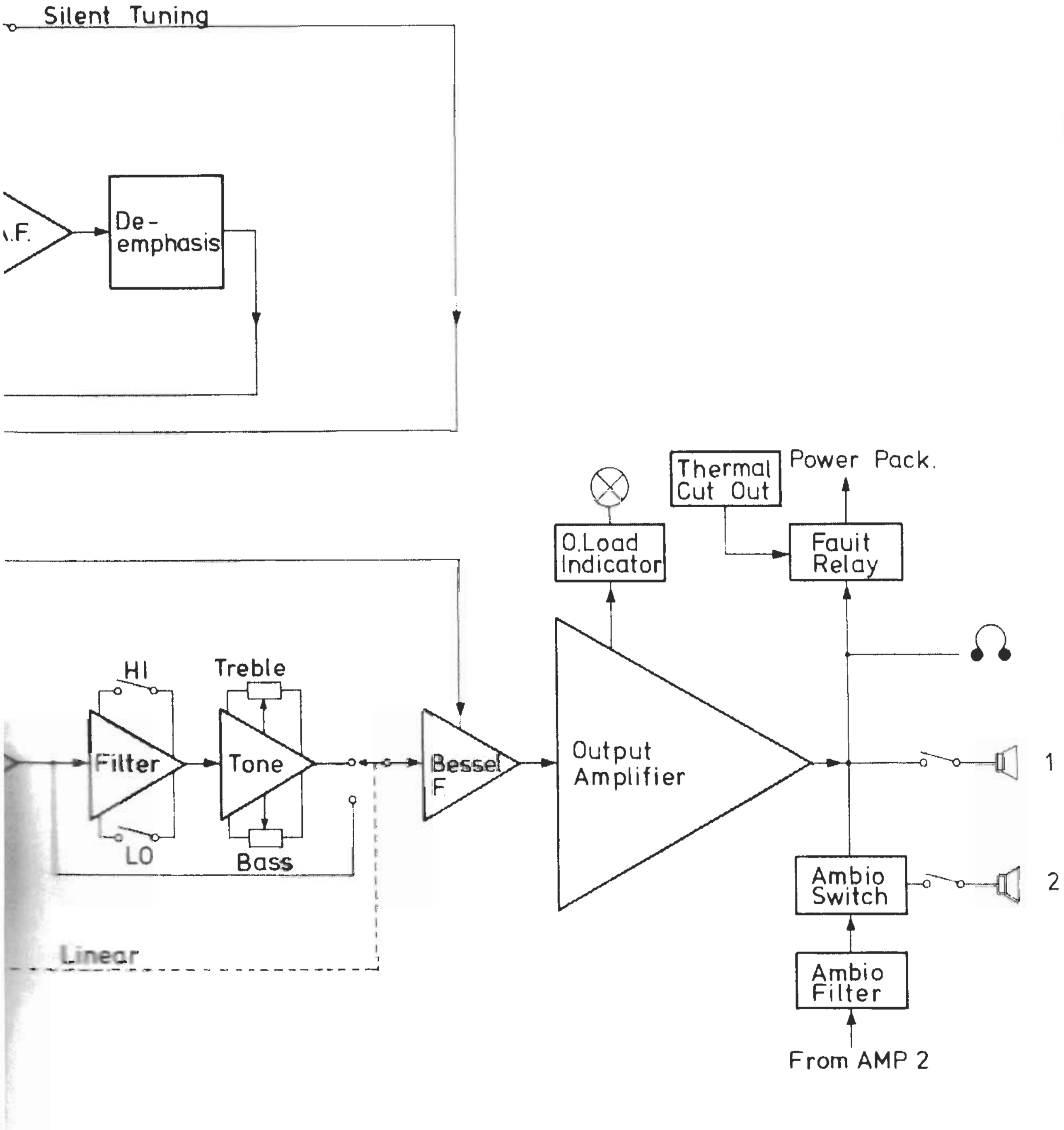
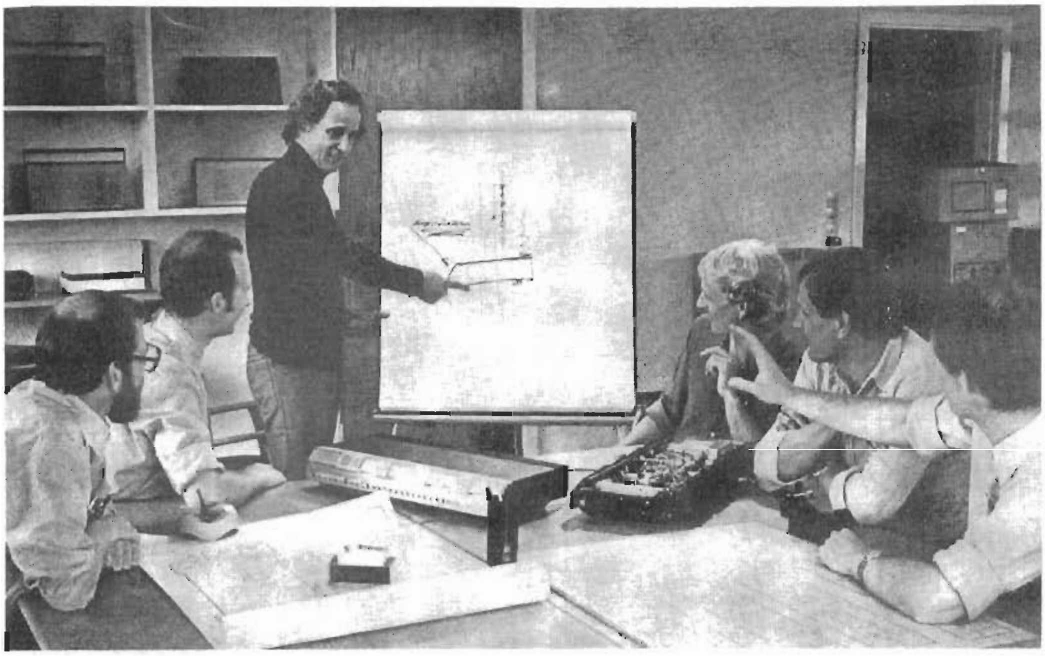


Fig. 33 – Block diagram of



ram of Beomaster 4400.

THE DESIGN TEAM



From left to right:

Niels Peter Jepsen – designed the power supply, switching and control circuits, and parts of the preamplifier circuits.

Egon Bjerre – did all experimental work on the effect of loudspeaker loads on amplifier performance, and also worked on the preamplifier and power amplifier circuits.

Jakob Jensen – was responsible for the aesthetic design for the Beomaster 4400, as for all other Bang & Olufsen audio products in recent years.

Eigil Thomsen – was in charge of mechanical design, cabinets and finish. He was also involved with production line layout, and the pilot run.

Tom Jelsing – did the theoretical word on TID, which enabled its elimination in a conventional amplifier. He also designed the power amplifier and parts of the FM tuner.

Claus Dissing – was mainly concerned with the FM tuner, particularly the intermediate frequency stages and the stereo decoder.